NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Computer Engineering

A Framework for Bandwidth Management in ATM Networks—Aggregate Equivalent Bandwidth Estimation Approach

GRADUATION PROJECT
COM – 400

Student : Orhan Gazi Kavak (990796)

Supervisor : Dr. Halil Adahan

Lefkoşa - 2003

ACKNOWLEDGMENT

I am glad to complete my project, which I had given with blessing of God (Thanks to God)

Next I would like to thank Dr.Halil Adahan for his endless and untiring support and help and his persistence, in the course of the preparation of this project.
Under his guidance, I have overcome many difficulties that I faced during the various stages of the preparation of this project.

I would like to thanks all of my friends who helped me to overcome my project especially Kadime Altungül,Harun Uslu,Mehmet Gögebakan,Celalettin Kunduraci,

Finally, I would like to thank my family, especially my parents for providing both moral and financial support Their love and guidance saw me through doubtful times. Their never-ending belief in me and their encouragement has been a crucial and a very strong pillar that has held me together.

They have made countless sacrifices for my betterment. I can't repay them, but I do hope that their endless efforts will bear fruit and that I may lead them, myself and all who surround me to a better future.

Also thanks all Teachers who behaved me in patient and understanding during my studying time

Specially to Assoc.Prof.Dr. Doğan Ibrahim for Everything he has done till now to help

ABSTRACT

A unified framework for traffic control and bandwidth management in ATM networks is proposed. It bridges algorithms for real-time and data services. The central concept of this framework is adaptive connection admission. It employs an estimation of the aggregate equivalent bandwidth required by connections carried in each output port of the ATM switches. The estimation process takes into account both the traffic source declarations and the connection superposition process measurements in the switch output ports. This is done in an optimization framework based on a linear Kalman filter. To provide a required

quality of service guarantee, bandwidth is reserved for possible estimation error. The algorithm is robust and copes very well with unpredicted changes in source parameters, thereby resulting in high bandwidth utilization while providing the required quality of service. The proposed approach can also take into account the influence of the source policing mechanism. The tradeoff between strict and relaxed source policing is discussed.

TABLE OF CONTENTS

CHAPTER 2

DYNAMIC BANDWIDTH MANAGEMENT IN ATM NETWORKS

CHAPTER 3

A FRAMEWORK FOR BANDWİDTH MANAGEMENT İN ATM NETWORKS AGGREGATE EQUİVALENT BANDWİDTH ESTİMATİON APPROACH

LIST OF FIGURES

LİST OF TABLES

CHAPTER 1

ASYNCHRONOUS TRANSFER MODE (ATM) NETWORKS

1. INTRODUCTION TO ATM

Asynchronous Transfer Mode, or ATM is a network transfer technique capable of supportinga wide variety of multimedia application with diverse service and performance requirements. It supports traffic bandwiths ranging from a few kilobites per second to several hundred megabits per second. And traffic types ranging from continuous, fixed-rate traffic to highly bursty traffic. ATM was designated by the telecommunication standardization sector of the International Telecommunication Union (ITU-T).

ATM is a form of packet-switching technology. That is, ATM networks transmit their information in small, fixed length packets called "cells", each of which contains 48 octets (or bytes) of data and 5 octets of header information. The small, fixed cell size was chosen to facilitate the rapid processing of packets in hardware And to minimize the amount of the time required to fill a single packet. This is particularly important for real-time applications such as voice and video that require short packetization delays.

ATM is also connection oriented. In other words, a virtual connection must be established before a "call" can take place, where a call is defined as the transfer of information between two or more end points.

Another important characteristic of ATM is that its network functions are typically implemented in hardware. With the introduction of high speed fiber optic transmission lines, the communication bottleneck has shifted from the communication links to the processing at switching nodes and at terminal equipment. Hardware implementation is necessary to overcome this bottleneck, because it minimizes the cell processing overhead, thereby allowing the network to match link rates on the order of Gbit/s.

Finally, as its name indicates, ATM is asynchronous. Time is slotted into cell-sized intervals, and slots are assigned to calls in an asynchronous, demand-based manner. Because slots are allocated to calls on demand ATM can easily accommodate traffic whose bit rate fluctuates over time . Moreover, in ATM also gains bandwidth efficiency by being able to statistically multiplex bursty traffic sources.

Since bursty traffic does not require continuous allocation of the bandwidth at its peak rate, statistical multiplexing allows a large number of bursty sources to share the network's bandwidth.

Since its birth in the mid-1980s, ATM has been fortified bt a number of robust standards and realized by a significant number of network equipment manufacturers.

International standards-making bodies such as the ITU and independent consort a like the ATM forum have developed a significant body of standards and implementation agrements for ATM.

1.1. Historical Background of ATM

Everyday the world seems to be moving at a faster and faster pace with new technological advances occurring constantly. In order to deliver new services such as video conferencing and video on demand, as well as provide more bandwidth for the increasing volume of traditional data, the communications industry introduced a technology that provided a common format for services with different bandwidth requirements. This technology is **Asynchronous Transfer Mode (ATM)**. As ATM developed, it became a crucial step in how companies deliver, manage and maintain their goods and services.

ATM was developed because of developing trends in the networking field. The most important parameter is the emergence of a large number of communication services with different, sometimes yet unknown requirements. In this information age, customers are requesting an ever increasing number of new services. The most famous communication services to appear in the future are HDTV(High Definition TV), video conferencing, high speed data transfer, videophony, video library, home education and video on demand.

This large span of requirements introduces the need for one universal network which is flexible enough to provide all of these services in the same way. Two other parameters are the fast evolution of the semi conductor and optical technology and the evolution in system concept ideas the shift of superfluous transport functions to the edge of the network. Both the need for a flexible network and the progress in technology and system concepts led to the definition of the Asynchronous Transfer Mode (ATM) principle.

Before there were computers that needed to be linked together to share resources and communicate, telephone companies built an international network to carry telephone calls. These wide area networks (WAN) were optimized to carry multiple telephone calls from one person to another, primarily using copper cable. As time passed, the bandwidth limitations of copper cable became apparent, and these WAN carriers began looking into upgrading their copper cable to fiber cable.

Because of its potential for almost unlimited bandwidth, carriers saw fibers as an essential part of their future. However, other limitations of the voice network still existed. Even though WAN carriers were upgrading to fiber, there were still no agreed upon standards that allowed equipment from different vendors' fiber-based equipment to be integrated together. The short-term solution to this problem was to upgrade to fiber; however, this was costly and time consuming. In addition, the lack of sophisticated network management in these WANs made them difficult to maintain.

Around the same time, computers were becoming more prevalent in the office. Networking these computers together was desirable and beneficial. When linking these computers over a long distance, the existing voice optimized WANs were used. Because computers send data instead of voice, and data has different characteristics, these WANs did not send computer data very efficiently. Therefore, separate WANs were sometimes built specifically to carry data traffic. Also, a network that could carry voice, data and video had been envisioned - something needed to be done.

To address these concerns, ITU-T (formerly CCITT) and other standards groups started work in the 1980s to establish a series of recommendations for the transmission, switching, signaling and control techniques required to implement an intelligent fiber-based network that could solve current limitations and would allow networks to be able to efficiently carry services of the future. This network was termed Broadband Integrated Services Digital Network (B-ISDN). By 1990, decisions had been made to base B-ISDN on SONET/SDH (Synchronous Optical Network/Synchronous Digital Hierarchy) and ATM.

THE HISTORY OF ATM TECHNOLOGY

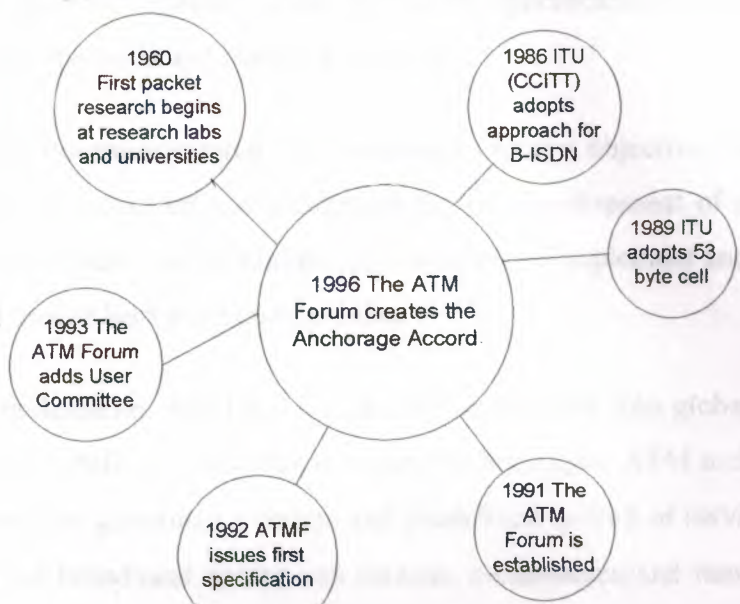


Figure 1.1 Historical Development of ATM.

SONET describes the optical standards for transmission of data. SONET/SDH standards specify how information can be packaged, multi-plexed and transmitted over the optical network. An essential element of SONET/SDH is to ensure that optical equipment and services from different vendors/service providers are interoperable and manageable. ITU-T now needed as switching standard to complement SONET in the B-ISDN model.

Because SONET only describes the transmission and multiplexing of information, without knowing what type of data or switching is being used, it can operate with nearly all emerging switching technologies. For B-ISDN, two types of switching were considered by the ITU-T: Synchronous and Asynchronous. An intelligent switching fabric with the ability to switch all forms of traffic at extremely high speeds, while maximizing the use of bandwidth, was needed to optimize the potential of B-ISDN. Ideally, maximum bandwidth should be accessible to all applications and users, and should be allocated on demand. ATM was chosen as the standard for B-ISDN that will ultimately satisfy these stringent requirements. Even though ATM was initially considered part of the solution for WANs, local area network (LAN) architects and equipment vendors saw ATM as a solution to many of their network limitations, and cable TV operators looked at ATM as a possible addition to their existing networks.

The ATM Forum was established in October, 1991 and issued its first specifications eight months later. The ATM Forum was formed to accelerate the user of ATM product and services through a rapid convergence of interoperability specifications. In addition, the Forum promotes industry cooperation and market awareness.

By 1996 The ATM Forum presented the Anchorage Accord objective. Fundamentally, the message is that the set of specifications needed for the development of multiservice ATM networks is available. These specifications were complete to implement and manage an ATM infrastructure, and ensure backward compatibility.

Entering the new millennium, ATM services are still in demand. The global market for ATM is in the billions of US dollars. Even with emerging technologies, ATM technology is still the only technology that can guarantee a certain and predefined quality of service. The growth of the Internet, need for broadband access and content, e-commerce and more are spurring the need for a reliable, efficient transport system - ATM Technology. For voice, video, data and images together, the next generation network depends on ATM.

1.1.1 ATM Technology

Asynchronous Transfer Mode (ATM) is the world's most widely deployed backbone technology. This standards-based transport medium is widely used within the core at the access and in the edge of telecommunications systems to send data, video and voice at ultra high, speed.

ATM is best known for its easy integration with other technologies and for its sophisticated management features that allow carriers to guarantee quality of service. These features are built into the different layers of ATM, giving the protocol an inherently robust set of controls.

Sometimes referred to as cell relay, ATM uses short, fixed-length packets called cells for transport. Information is divided among these cells, transmitted and then re-assembled at their final destination.

1.1.2. ATM in The Telecommunications Infrastructure

A telecommunications network is designed in a series of layers. A typical configuration may have utilized a mix of time division multiplexing, Frame Relay, ATM and/or IP. Within a network, carriers often extend the characteristic strengths of ATM by blending it other technologies, such as ATM over SONET/SDH or DSL over ATM. By doing so, they extend the management features of ATM to other platforms in a very cost-effective manner.

ATM itself consists of a series of layers. The first layer known as the adaptation layer holds the bulk of the transmission. This 48-byte payload divides the data into different types. The ATM layer contains five bytes of additional information, referred to as overhead. This section directs the transmission. Lastly, the physical layer attaches the electrical elements and network interfaces.

1.1.3. ATM as The Backbone for Other Networks

The vast majority (roughly 80 percent) of the world's carriers use ATM in the core of their networks. ATM has been widely adopted because of its unmatched flexibility in supporting the broadest array of technologies, including DSL, IP Ethernet, Frame Relay, SONET/SDH and wireless platforms. It also acts a unique bridge between legacy equipment and the new generation of operating systems and platforms. ATM freely and easily communicates with both, allowing carriers to maximize their infrastructure investment.

1.1.4. ATM in The LAN (Local Area Network)

The LAN environment of a campus or building appears sheltered from the headaches associated with high-volumes of traffic that deluge larger networks. But the changes of LAN interconnection and performance are no less critical.

The ATM/LAN relationship recently took a giant step forward when a prominent U.S. vendor announced a patent for its approach to extending ATM's quality of service to the LAN. The filing signals another birth in a long lineage of applications that prove the staying power and adaptability of ATM.

ATM is a proven technology that is now in its fourth generation of switches. Its maturity alone is not its greatest asset. Its strength is in its ability to anticipate the market and quickly respond, doing so with the full confidence of the industry behind it.

1.1.5. ATM in The WAN (Wide Area Network)

A blend of ATM, IP and Ethernet options abound in the wide area network. But no other technology can replicate ATM's mix of universal support and enviable management features. Carriers inevitably turn to ATM when they need high-speed transport in the core coupled with the security of a guaranteed level of quality of service. When those same carriers expand to the WAN, the vast majoriy does so with an ATM layer.

Distance can be a problem for some high-speed platforms. The integrity of the transport signal is maintained even when different kinds of traffic are traversing the same network. And because of its ability to scale up to OC-48, different services can be offered at varying speeds and at a range of performance levels.

1.1.6. ATM in The MAN (Metropolitan Area Network)

The MAN is one of the hottest growing areas in data and telecommunications. Traffic may not travel more than a few miles within a MAN, but it's generally doing so over leading edge. Technologies and at faster than lightening speeds.

The typical MAN configuration is a point of convergence for many different types of traffic that are generated by many different sources. The beauty of ATM in the MAN is that it easily accommodates these divergent transmissions, often times bridging legacy equipment with ultra high-speed networks. Today, ATM scales form T-1 to OC-48 at speeds that average 2.5 Gb/s in operation, 10 Gb/s in limited use and spanning up to 40 Gb/s in trials.

1.2. Bandwidth Distribution

The Bandwidth Distribution (BD) component is responsible for the management of the bandwidth allocated to working VPC's according to actual traffic conditions. That is, it adjusts the bandwidth allocated to the working VPC's to their actual usage to avoid situations where in the same links some VPC's tend to become over-utilized while other VPC's remain underutilized.

The dynamic management of the working VPC allocated bandwidth is achieved by distributing portions (as a common pool) of the link working bandwidth (link capacity minus restoration bandwidth) among the working VPC's. Specifically, the management of the VPC allocated bandwidth is done within specific (upper and lower) bounds on the VPC bandwidth as originally estimated by VPC_LD (VPC required bandwidth).

The activities of BD are required to compensate for inaccuracies in traffic predictions and in the VPC bandwidth as estimated by VPC_LD as well as to withstand (short to medium) actual traffic variations. This is so, since it cannot be taken for granted that the traffic predictions will be accurate and furthermore even if they are accurate, they are accurate within statistical range. In this respect, considering the random nature of the arriving traffic, the bandwidth that needs to be allocated to VPC, so that certain objectives (regarding connection admission) to be met, is a stochastic variable, depending on the connection arriving pattern. The VPC_LD component estimates originally the bandwidth that needs to be allocated to the VPC's (required bandwidth) so that satisfy traffic predictions. This is viewed as the mean value of the (stochastic in nature) bandwidth that needs to be allocated to the VPC's. It is the task then of the BD component, to manage the allocated bandwidth of VPC's, around the (mean) required bandwidth, according to actual traffic conditions.

By monitoring the usage on working VPC's, the BD component also emits warnings to VPC_LD indicating insufficient usage of the planned network resources. The warnings are issued in cases where some VPC's remain under-utilized (with respect to their required bandwidth as specified by VPC_LD) for a significant period of time. This implies that these resources cannot be utilized in the routes by the Load Balancing component and therefore such cases are interpreted as indicating overestimation of network resources.

The proposed algorithm for bandwidth redistribution assumes that there is a common pool of bandwidth per link to be redistributed to the VPC's when necessary. The algorithm assumes that this common pool of bandwidth per link is the links' unallocated bandwidth. This pool of bandwidth is not totally allocated to the VPC's at any instant; but it is there to be allocated to the VPC's that go highly utilized only when such conditions occur. Each VPC grabs or returns portions of its allocated bandwidth to the common pool of bandwidth according to its congestion level. We assume here that the modification of the bandwidth of a VPC does not impact on the traffic parameters (QoS) of the VCs using this VPC or other VPC's sharing the same links or nodes.

1.3. ATM Standards

The telecommunication standardization sector of the ITU, the international standards agency commissioned by the United Nations for the global standardization of telecommunication, has developed standards for ATM networks. Other standards bodies and consortia have also contributed to the development of ATM.

1.3.1. Protocol Reference Model

The purpose of the protocol reference model is to clarify the funtions that ATM networks perform by grouping them into a set of interrelated, function-specific layers and planes. The reference model consist of a user plane, a control plane and a management plane. within the user and control planes is a hierarchical set of layers.

The user plane defines a set of functions for the transfer of user information between communication end-points. The conrol plane defines the control functions such as call establishment, call maintanance, and call release; and the management plane defines the operations necessary to control information flow between planes and layers, and to maintain accurate and fault –tolerant network operation.

Within the user and control planes, there are three layers: the physical layer, the ATM layer, and the ATM adaptation layer (ALL).

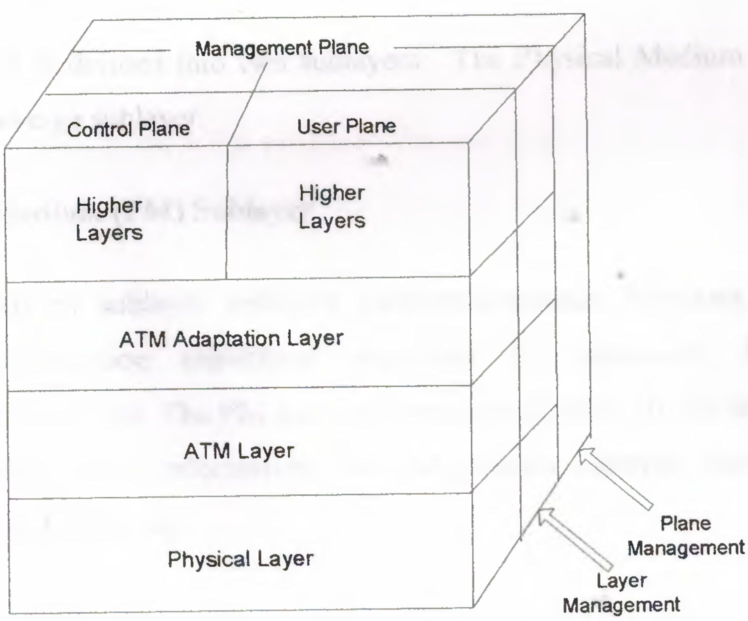


Figure 1.2 Protocol Reference Model for ATM.

Within the user and control planes, there are three layer, the physical layer, the ATM layer and the ATM adaptation layer (ALL). Table 1.1 summarizas the functions of each layer. The physical layer performs primarily bit level functions, the ATM layer is primarily responsible for the switching of ATM cells, and the ATM adaptation layer is responsible for the conversion of higher layer protocol frames into ATM cells. The functions that the physical, ATM, and adaptation layers perform are described in more detail in the following.

Layer Management	Higher Layer Function	Higher Layer	
	Convergance	CS	ALL
	Segmentation and reassambly	SAR	
	Generic Flow control Cell header generation / extraction Cell VPI / VCI translation Cell multiplex and demultiplex	ATM	
	Cell flow Control Header Error Control (HEC) Cell delineation Transmission frame adaptation Transmission frame generation / recovery	TC	Physical Layer
	Bit timing Physical Medium	PM	

Table 1.1 Functions of Each Layer in the Protocol Reference Model.

1.4. Physical Layer

The physical layer is divided into two sublayers: The Physical Medium sublayer and The Transmission Converge sublayer.

1.4.1. Physical Medium (PM) Sublayer

The physical medium sublayer performs medium-dependent functions. For example, it provides bit transmission capabilities including bit alignment, line coding and electrical/optical conversion. The PM sublayer is also responsible for bit timing, the insertion and extraction of bit timing information. The PM sublayer currently supports two types of interface, optical and electrical.

1.4.2. Transmission Convergence (TC) Sublayer

Above the physical medium sublayer is the transmission converge sublayer, which is primarily responsible for the framing of data transported over the physical medium. The ITU_T recomendation specifies two options for TC sublayer transmission frame structure cell-based and Synchronous Digital Hierarchy (SDH). In the cell-based case, cells are transported continuously without any regular frame structure. Under SDH, cells are carried in a special frame structure based on the north American SONET (Synchronous Optical Network) protocol.

Regardless of which transmission frame structure is used, the TC sublayer is responsible for the following four funtions. Cell rate decoupling, header error control, cell delineation, and transmission frame adaptation. Cell rate decoupling is the insertion of idle cells at the sending side to adapt the ATM cell stream's rate to the rate of the transmission path .

Header error control is the insertion of an 8-bit CRC polynomial in the ATM cell header to protect the contents of the ATM cell header. Cell delineation is the detection of cell boundaries. Transmission frame adaptation is the encapsulation of departing cells into an appropriate framing structure.

1.5. ATM Layer

The ATM layer lies a top the physical layer and specifies the functions required for the switching and flow control of ATM cells.

There are two interfaces in an ATM network: The user network interface (UNI) between the ATM end point and the ATM switch, and the network-network interface (NNI) between two ATM switches.

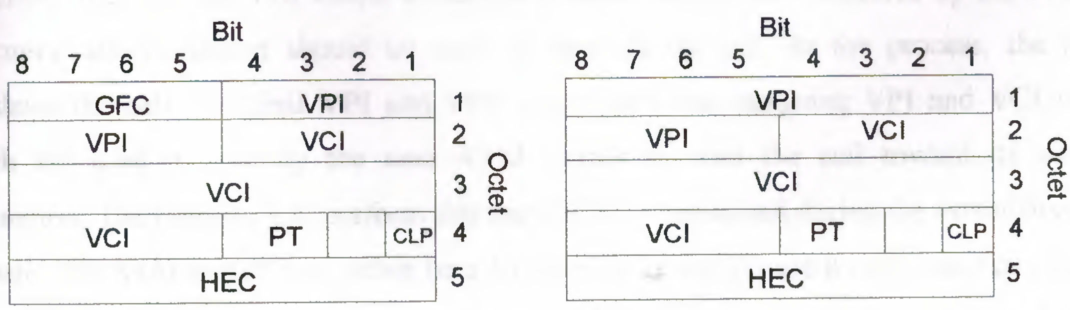


Figure 1.3 ATM Cell Header Structure.

Although a 48-octet cell payload is used at both interfaces, the 5 octet cell header differs slightly at these interfaces. Figure 1.3. shows the cell header structures used at the UNI and NNI .

At the UNI , the header contains a 4- bit generic flow control (GFC) field, a 24-bit label field containing Virtual Path Identifier (VPI) and Virtual Channel Identifier (VCI) subfields (8 bits for the VPI and 16 bits for the VCI), a 2-bit payload type (PT) field, a 1-bit priority (PR) field, and an 8-bit header error check (HEC) field. The cell Header for an NNI cell is identical to that for the UNI cell, except that it lacks the GFC field; these four bits are used for an additional 4 VPI bits in the NNI cell header.

The VCI and VPI fields are identifier values for virtual channel (VC) and virtual path (VP), respectively. A virtual channel connects two ATM communication end-points. A virtual path connects two ATM devices, which can be switches or end-points, and several virtual channels may be multiplexed onto the same virtual path.

The 2-bit PT field identifies whether the cell payload contains data or control information. The CLP bit is used by the user for explicit indication of cell loss priority. If the value of the CLP is 1 the the cell is subjected to discarding in case of congestion. The HEC field is an 8 bit CRC polynomial that protects the contents of the cell header.

The GFC field, which appears only at the UNI, is used to assist the customer premises network in controlling the traffic flow for different qualities of service . At the time of writing the exact procedures for use of this field have not been agreed upon .

1.6. ATM Layer Functions

The primary function of the ATM layer is VPI/VCI translation. As ATM cells arrive at ATM switches, the VPI and VCI values contained in their headers are examined by the switch to determine which outport should be used to forward the cell. In the process, the switch translates the cell's original VPI and VCI values into new outgoing VPI and VCI values, which are used in turn by the next ATM switch to send the cell toward its intended destination. The table used to perform this translation is initialized during the establishment of the call. An ATM switch may either be a VP switch, in which case it only translates the VPI values contained in cell headers, or it may be a VP/VC switch, in which case it translates the incoming VCI value into an outgoing VPI/VCI pair.

Since VPI and VCI values do not represent a unique end-to-end virtual connection. They can be reused at different switches through the network. This is important, because the VPI and VCI fields are limited in lenght and would be quickly exhausted if they were used simply as destination addresses.

The ATM layer supports two types of virtual connections : switched virtual connection (SVC) and permanent, or semipermanent, virtual connections(PVC). Switched virtual connections are established and torn down dynamically by an ATM signaling procedure. That is they only exist for the duration of a single call.

Permanent virtual connections, on the other hand, are established by network administrators and continue to exist as long as the administrator leavesthem up, even if they are not used to transmit data. Other important functions of the ATM layer include cell multiplexing and demultiplexing, cell header creation and extraction, and genericflow control.

Cell multiplexing is the merging of cells from several calls onto a single transmission path, cell header creation is the attachment of a 5- octet cell header to each 48 octet block of user payload, and generic flow control is used at the UNI to orevent short-term overload conditions from occurring within the network.

1.7. ATM Layer Service Categories

The ATM Forum and ITU-T have defined several distinct service categories at ATM layer. The categories defined by the ATM forum include constant bit rate (CBR), real-time variable bit rate (VBR-rt), non real-time variable bit rate (VBR-nrt), available bit rate (ABR), and unsipecific bit rate (UBR). ITU-T defines four service categories, namely, deterministic bit rate(DBR), statistical bit rate (SBR), avalable bit rate (ABR) and ATM block transfer(ABT). The first of the three ITU-T service categories correspond roughly to the ATM Forum's CBR,VBR and ABR classifications, respectively.

The fourth service category, ABT, is solely defined by ITU-T ans is intended for bursty data application.The UBR category defined bt the ATM Forum is for calls that request no quality of service guarantees at all. The constant bit rate CBR (or deterministic bit rate DBR) service category provides a very strict QoS guararentee. It is targeted at real-time applications, such as voice and raw video, which mandate severe restrictions on delay, delay variance(jitter) and cell loss rate.

The only traffic description required by the CBR service are the peak cell rate and the cell delay variation tolerance. A fixed amount of bandwidth, determined primarily by the call's peak cell rate, is reservedfor each CBR connection. The real-time variable bit rate VBR-rt (or statistical bit rate SBR) service category is intended for real time bursty application, which also require strict QoS guarantees.

The primary difference CBR and VBR-rt is in the traffic descriptions they use. The VBR-rt service requires the specification of the sustained cell rate and bursty tolerance in addition to the peak cell rate and the cell delay variation tolerance. The ATM Forum also defines a non-real-time VBR-nrt service category, in which cell delay variance is not guaranteed.The available bit rate(ABR) service category is defined to exploit the network's unutilized bandwidth. It is intended for non-real time data application in which the source is amenableto enforced adjustment of its transmission rate.

A minimum cell rate is reserved for the ABR connection and therefore guaranteed by the network. When the network has unutilized bandwidth, ABR sources are allowed to increase their cell rates up to an allowed cell rate(ACR), a value which is periodically updated by the ABR flow control mechanism. The value of ACR always falls between the minimum and the peak cell rate for the connection and is determined by the network.

The ATM forum defines another service category for non-real-time application called the unspecified bit rate (UBR) service caregory. UBR service is entirelybest efford, the call is provided with no QoS guarantees. The ITU-T also defines an additional sewrvice category for non-real-time data applications. The ATM block transfer (ABT) service category is intended for the transmission option (ABT/IT), the block of data is sent at the same time as the reservation request.

If bandwidth is not available for transporting block, then it is simply discarded; and the source must retransmit it. In the ABT service with delayed transmission (ABT/DT); the source waits for a confirmation from the network that enough bandwitdh is available before transmitting the block of tha data. In both cases, the network temprorarily resource reserves bandwitdh according to the peak. Cell rate for each block.Imediately after transporting the block, the network releases the reserved bandwidh.

ITU-T Service Categories	DBR	SBR	ABT	ABR	
ATM Forum Services Categories	CBR	VBR-rt	VBR-ntr	ABR	UBR
Cell Loss Rate	specified				unspecified
Cell Transfer Delay	specified			unspecified	
Cell Delay Variation	specified		unspecified		
Traffic Descriptors (Contract)	PCR/CDVT	PCR/CDVT SCR/BT		PCR/CDVT MCR/ACR	PCR/CDVT

PCR = Peak Cell Rate
CDVT = Cell Delay Variation Tolerance
MRC = Minimum Cell Rate
SCR = Sustained Cell Rate
BT = Burst Tolerance
ACR = Allowed Cell Rate

Table 1.2 ATM Layer Service Categories.

1.8. ATM Adaptation Layer

The ATM adaptation layer (AAL), which resides a top ATM layer, is responsible for mapping the requirements of higher layer protocols onto the ATM network. It operates in ATM devices at the edge of the ATM network and is totally up sent in ATM switches. The adaptation layer is divided into two sublayers: The convergence sublayer (CS), which performs error detection and handling, timing and clock recovery and the segmentation and reassembly (SAR) sublayer, which performs segmentation of convergence sublayer protocol data units (PDUs) into ATM cell-sized SAR sublayer service data units data units (SDUs) and vice versa In order to support different service requirements, the ITU-T proposed for AAL-specific services classes.

Note that while these ALL service classes are similar in many ways to the ATM layer service catagories defined in the perivious section, they are not the same; each exists at a different layer of the protocol reference model, and each requires a different set of functions. ALL service class A corresponds to constant bit rate (CBR), services with a timing the relation required between source and destination. The connection mode is connection – oriented . CBR audio and video blong to this class. Class B corresponds to variable bit rate (VBR) services. This class also requires timin between sources and destination, and its mode is connection-oriented. VBR audio and video are examples of class B services. Class C also corresponds to VBR connection oriented services but the timing between source and destination needs not be related. Class C includes connection-oriented data transfer such as X.25, signaling and future high speed data services. Class D corresponds to connectionless services. Connectionless data services such as those supported by LANs and WANs are examples of class D services.

Four AAL types, each with a unique SAR supplier and CS sublayer, are defined to support the four service classes. ALL type 1 supports constant bit rate services (Class A), and AAL type 2 supports available bit rate services with a timing relation between source and destination (Class B). ALL type 3 /4 was orginally specified as two different AAL type (Type 3 and Type 4), but due to their inherient similarities, they were eventually merget to support both Class C and Class D services. AAL Type 5 also supports class C and Class D services.

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

Table 1.3 Service Classification for AAL.

1.8.1. ALL Type5

Currently the most widely used adaptation layer is AAL type 5. AAL type 5 supports connection-oriented and connectionless services in which there is no timing relation between source and destination(class C and class D). Its functionality was intentionaly made simple in order to support high speed data transfer. AAL type 5 assumes that the layers above the ATM adaptation layer can perform error recovery. Retransmition and sequence numbering when require and those it does not provide this functions. Therefore, only none assured operation is provided, lost or corrupted AAL type 5 packet will not be corrected by retransmition.

Figure 4 depicts the SAR-SDU format for AAL type 5. the SAR supplier of AAL type 5 performs segmentation of CS-PDU into a size suitable for the SAR-SDU pay load. Unlike other AAL types, Type 5 devotes the entire 48-octet payload of the ATM cell to the SAR-SDU ; there is no overhead. An AAL specific flag in the ATM Payload Type (PT).

Figure 1.4 SAR - SDU Format for AAL Type 5.

Field of the cell header is set when the last cell of a CS-PDU is sent . the assembly of the CS-PDU frames at the destionation is controlled by using this flag.

Figure 5 depicts the CS-PDU format for AAL type 5. it conteins the user data payload, along with any recessary padding bits (PAD) and a CS-PDU trailer, which are added by the CS supplier when it recveives the uder information from the higher layer. The CS–PDU is padded using 0 + 47 bytes of PAD field to make the land of the CS-PDU and integral muliple of 48 bytes (the size of the SAR -SDU) at the receiving end, reassembled PDU is passed to the CS sublayer from the SAR sublayer, CRC values are then calculared and compared.

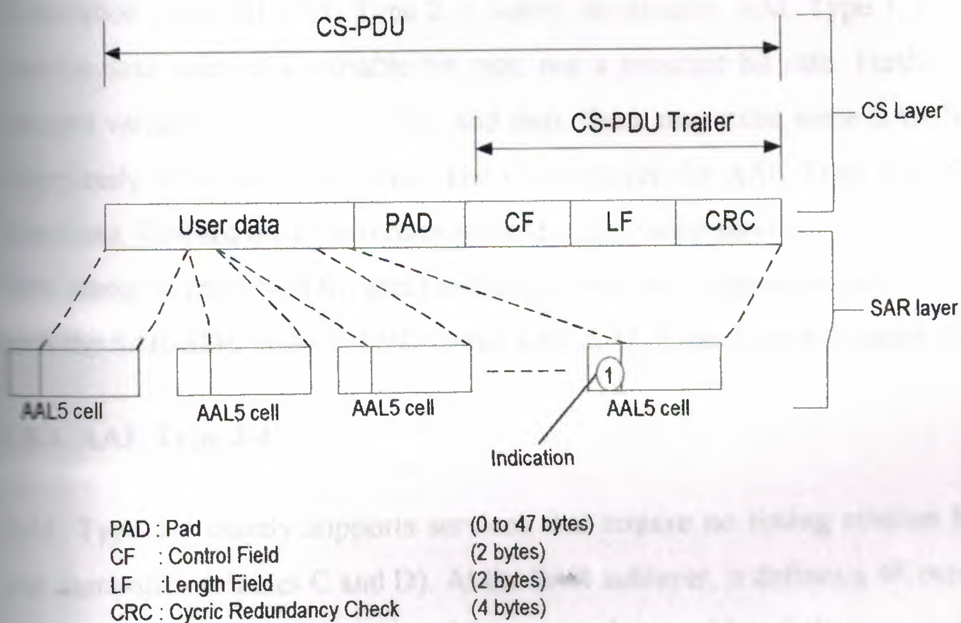


Figure 1.5 CS-PDU Format Segmentation and Reassembly of AAL Type 5.

If there is no error, the PAD field is removed by using the value of lenght field(LF) in the CS-PDU trailer, and user data is passed to the higher layer. If an error is detected, the erroneous information is either deliverded to the use or discardedaccording to user 's choice . the use of the CF field is for further study.

1.8.2. AAL Type1

AAL Type supports constant bit rate services with a fixed timing relation between source and destination users (class A) At the SAR sublayer, it defines a 48-octet service data unit (SDU), which contains octets of user payload, bits for a sequence number, and a 4-bit CRCvalue to detect errors in the sequence number field. AAL Type1 performs the following services at the CS sublayer forward error correction to ensure high quality of audio and video applications, clock recovery by monitoring the buffer filling, explicit time indication by inserting a time stamp in the CS-PDU, and handling of lost and misinserted cells which are recognized by the SAR. At the time of writing, the CS-PDU format has not been decided.

1.8.3. AAL Type2

AAL Type 2 supports variable bit rate services with a timing relation between source and destination (class B) AAL Type 2 is nearly identical to AAL Type 1, except that it transfers service data units at a variable bit rate, not a constant bit rate. Furthermore, AAL Type 2 accepts variable length CS-PDUs, and thus, there may exist some SAR-SDUs which are not completely filled with user data. The CS sublayer for AAL Type 2 performs the following functions, forward error correction for audio and video services, clock recovery by inserting a time stamp in the CS-PDU, and handling of lost and misinserted cells. At the time of writing, both the SAR-SDU and CS-PDU formats for AAL Type 2 are still under discussion.

1.8.4. AAL Type 3/4

AAL Type 3/4 mainly supports services that require no timing relation between the source and destination (classes C and D). At the SAR sublayer, it defines a 48-octet service data unit, with 44 octets of user payload, a 2-bit payload type eld to indicate whether the SDU is the beginning, middle, or end of a CS-PDU, a 4-bit cell sequence number, a 10-bit multiplexing identifier that allows several CS-PDUs to be multiplexed over a single VC, a 6-bit cell payload length indicator, and a 10-bit CRC code that covers the payload. The CS-PDU format allows for up to 65535 octets of user payload and contains a header and trailer to delineate the PDU.

The functions that AAL Type 3/4 performs include segmentation and reassembly of variable-length user data and error handling. It supports message mode (for framed data transfer) as well as streaming mode (for streamed data transfer). Since Type 3/4 is mainly intended for data services, it provides a retransmission mechanism if necessary.

1.9. ATM Signalling

ATM follows the principle of out-of-band signaling that was establihed for N-ISDN . In other words, signaling and data channels are separate. The main purposes of signaling are:

1) To establish, maintain and release ATM virtual connections.
2) To negotiate the traffic parameters of new connections

The ATM signaling standards support the creation of point to point as well as multicast connections. Typically certain VCI and VPI values are reserved by ATM networks for signaling messages. If additional signaling VCs are required, they may be establish through the process of meta-signaling.

2. ATM TRAFFIC CONTROL

The control of ATM traffic is complicated due to ATM's high link speed and small cell size, the diverse service requirements of ATM applications, and the diverse characteristics of ATM traffic. Furthermore, the configuration and size of the ATM environment, either local or wide area, has a significant impact on the choice of traffic control mechanisms.

The factor which most complicates traffic control in ATM is its high link speed. Typical ATM link speeds are155.52 Mbit/s and 622.08 Mbit/s At these high link speeds, 53 byte ATM cells must be switched at rates greater than one cell per 2.726 μs or 0.682μs, respectively. It is apparent that the cell processing required by traffic control must perform at speeds comparable to these cell switching rates. Thus, traffic control should be simple and efficient, without excessive software processing.

Such high speeds render many traditional traffic control mechanisms inadequate for use in ATM due to their reactive nature. Traditional reactive traffic control mechanisms attempt to control network congestion by responding to it after it occurs and usually involves sending feedback to the source in the form of a choke packet. However; a large propagation bandwidth product (i.e.; the amount of traffic that can be sentin a single propagation delay time) renders many reactivecontrol schemes ineffective in high speed networks. When a node receives feedback, it mayhave already transmitted a large amount of data. Consider a cross continental 622 Mbit/s connection with a propagation delay of 20 ms (propagation bandwidth product of 12.4 Mbits). If a node at one end of the connection experiences congestion and attempts to throttle the source at the other end by sending it a feedback packet, the source will already have transmitted over twelve megabits of information before feedback arrives. This example illustrates the ineffectiveness of traditional reactive traffic control mechanisms in high speed networks and argues for novel mechanisms that takeinto account high propagation bandwidth products.

Not only is traffic control complicated by high speeds, but it is made more di cult by the diverse quality of service (QoS) requirements of ATM applications. For example, many applications have strict delay requirements and must be delivered within a speci ed amount of time. Other applications have strict loss requirements and must be delivered reliably without an inordinate amount of loss. Traffic controls must address the diverse requirements of such applications.

Another factor complicating traffic control in ATM networks is the diversity of ATM traffic characteristics. In ATM networks continuous bit rate traffic is accompanied by bursty traffic. Bursty traffic generates cells at a peak rate for a very short period of time and then immediately becomes less active, generating fewer cells. To improve the e ciency of ATM network utilization, bursty calls should be allocated an amount of bandwidth that is less than their peak rate. This allows the network to multiplex more calls by taking advantage of the small probability that a large number of bursty calls will be simultaneously active. This type of multiplexing is referred to as statistical multiplexing. The problem then becomes one of determining how best to statistically multiplex bursty calls such that the number of cells dropped due to excessive burstiness is balanced with the number of bursty traffic streams allowed. Addressing the unique demands of bursty traffic is an important function of ATM traffic control.

For the reasons mentioned above, many traffic control mechanisms developed for existing networks may not be applicable to ATM networks, and therefore novel forms of traffic control are required. One such class of novel mechanisms that work well in high speed networks falls under the heading of preventive control mechanisms. Preventive control attempts to manage congestion by preventing it before it occurs. Preventive traffic control is targeted primarily at real time traffic. Another class of traffic control mechanisms has been targeted toward non real time data traffic and relies on novel reactive feedback mechanisms.

2.1. Preventive Traffic Control

Preventive control for ATM has two major components; call admission control and usage parameter control. Admission control determines whether to accept or reject a new call at the time of call set up. This decision is based on the traffic characteristics of the new call and the currentnetwork load. Usage parameter control enforces the traffic parameters of the call once it has been accepted into the network. This enforcement is necessary to insure that the call's actual traffic flow conforms with that reported during call admission.

Before describing call admission and usage parameter control in more detail, it is important to first discuss the nature of multimedia traffic. Most ATM traffic belongs to one of two general classes of traffic; continuous traffic and bursty traffic. Sources of continuous traffic are easily handled, because their resource utilization is predictable and they can be deter ministically multiplexed. However, bursty traffic (e.g. voice with silence detection; variable bit rate video) is characterized by its unpredictability, and it is this kind of traffic which complicates preventive traffic control.

Burstiness is a parameter describing how densely or sparsely cell arrivals occur. There are a number of ways to express traffic burstiness, the most typical of which are the ratio of peak bit rate to average bit rate, and the average burst length. Several other measures of burstiness have also been proposed. It is well known that burstiness plays a critical role in determining network performance, and thus, it is critical for traffic control mechanisms to reduce the negative impact of bursty traffic.

2.1.1. Call Admission Control

Call admission control is the process by which the network decides whether to accept or reject a new call. When a new call requests access to the network, it provides a set of traffic descriptors (e.g.; peak rate, average rate, average burst length) and a set of quality of service requirements (e.g.; acceptable cell loss rate, acceptable cell delay variance, acceptable delay). The network then determines, through signaling, if it has enough resources (e.g.; bandwidth, buffer space) to support the new call's requirements. If it does, the call is immediately accepted and allowed to transmit data into the network. Otherwise it is rejected. Call admission control prevents network congestion by limiting the number of active connections in the network to a level where the network resources are adequate to maintain quality of service guarantees.

One of the most common ways for an ATM network to make a call admission decision is to use the call's traffic descriptors and quality of service requirements to predict the "equivalent bandwidth" required by the call. The equivalent bandwidth determines howmany resources need to be reserved by the network to support the new call at its requested quality of service. For continuous, constant bit rate calls, determining the equivalent bandwidth is simple. It is merely equal to the peak bit rate of the call. For bursty connections, however, the process of determining the equivalent bandwidth should takeinto accountsuch factors as a call's burstiness ratio (the ratio of peak bit rate to average bit rate), burst length, and burst interarrival time. The equivalent bandwidth for bursty connections must be chosen carefully to ameliorate congestion and cell loss while maximizing the number of connections that can be statistically multiplexed.

2.1.2. Usage Parameter Control

Call Admission Control is responsible for admitting or rejecting new calls. However, call admission by itself is ineffective if the call does not transmit data according to the traffic parameters it provided. Users may intentionally or accidentally exceed the traffic parameters declared during call admission, there by overloading the network. In order to prevent the network users from violating their traffic contracts and causing the network to enter a congested state, each call's traffic flow is monitored and, if necessary, restricted. This is the purpose of usage parameter control. (Usage parameter control is also commonly referred to as policing, bandwidth enforcement, or flow enforcement.)

To effciently monitor a call's traffic, the usage parameter control function must be located as close as possible to the actual source of the traffic. An ideal usage parameter control mechanism should have the ability to detect parameter violating cells, appear transparent to connections respecting their admission parameters, and rapidly respond to parameter violations. It should also be simple, fast, and cost effective to implement in hardware. To meet these requirements, several mechanisms have been proposed and implemented.

The leaky bucket mechanism originally proposed in is a typical usage parameter control mechanism used for ATM networks. It can simultaneously enforce the average bandwidth and the burst factor of a traffic source. One possible implementation of the leaky bucket mechanism is to control the traffic flow by means of tokens. A conceptual model for the leaky bucket mechanism is illustrated in figure 1.6.

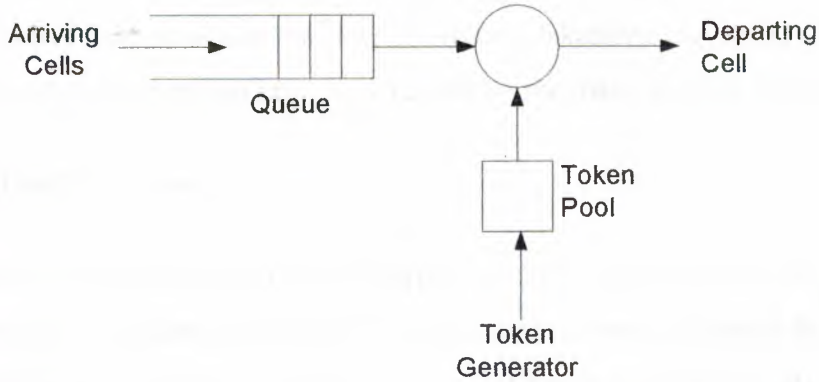


Figure 1.6 Leaky Bucket Mechanism.

In Figure 1.6, an arriving cell first enters a queue, if the queue is full. cells are simply discarded, to enter the network, a cell must first obtain a token from the token pool ,if there is no token, a cell must wait in the queue until a new token is generated. Tokens are generated at a fixed rate corresponding to the average bit rate declared during call admission. If the number of tokens in the token pool exceeds some predefined threshold value, token generation stops. This threshold value corresponds to the burstiness of the transmission declared at call admission time; for larger threshold values, a greater degree of burstiness is allowed. This method enforces the average input rate while allowing for a certain degree of burstiness.

One disadvantage of the leaky bucket mechanism is that the bandwidth enforcement introduced by the token pool is in effect even when the networkloadislight and there is no need for enforcement. Another disadvantage of the leaky bucket mechanism is that it may mistake non violating cells for violating cells. When traffic is bursty, a large number of cells may be generated in a short period of time, nevertheless conforming to the traffic parameters claimed at the time of call admission. In such situations, none of these cells should be considered violating cells. Yet in actual practice, leaky bucket may erroneously identify such cells as violations of admission parameters. To overcome these disadvantages, a virtual leaky bucket mechanism (also referred to as a marking method) has been proposed. In this mechanism, violating cells, rather than being discarded or buffered, are permitted to enter the networkat alower priority (CLP=1). These violating cells are discarded only when they arrive at a congested node. If there are no congested nodes along the routes to their destinations, the violating cells are transmitted without being discarded. The virtual leaky bucket mechanism can easily be implemented using the leaky bucket method described earlier. When the queue length exceeds a threshold, cells are marked as "droppable" instead of being discarded. The virtual leaky bucket method not only allows the user to takeadvantage of a light network load, but also allows a larger margin of error in determining the token pool parameters.

2.2. ReactiveTraffic Control

Preventive control is appropriate for most types of ATM traffic. However, there are cases where reactive control is beneficial. For instance, reactivecontrol is useful for service classes like ABR, which allow sources to use bandwidth not being utilized by calls in other service classes. Such a service would be impossible with preventive control, because the amount of unutilized bandwidth in the network changes dynamically, and the sources can only be made aware of the amount through reactive feedback.

There are two major classes of reactive traffic control mechanisms: Rate Based and Credit Based. Most rate based traffic control mechanisms establish a closed feedback loop in which the source periodically transmits special control cells, called resource management cells, to the destination (or destinations). The destination closes the feedback loopby returning the resource management cells to the source. As the feedback cells traverse the network, the intermediate switches examine their current congestion state and mark the feedback cells accordingly. When the source receives a returning feedback cell, it adjusts its rate, either by decreasing it in the case of network congestion, or increasing it the case of network underutilization. An example of a rate based ABR algorithm is the Enhanced Proportional Rate Control Algorithm (EPRCA) which was proposed, developed, and tested through the course of ATM Forum activities.

Credit-based mechanisms use link-by-link traffic control to reduce loss and optimize utilization. Inter mediate switches exchange resource management cells that contain "credits," which reflect the amountof buffer space available at the next downstream switch A source cannot transmit a new data cell unless it has received at least one credit from its downstream neighbor An example of a credit based mechanism is the Quantum Flow Control QFC algorithm developed by a consortium of reseachers and ATM equipment manufacturers.

3. HARDWARE SWITCH ARCHITECTURES FOR ATM NETWORKS

In ATM networks, information is segmented into fixed length cells, and cells are asynchronously transmitted through the network. To match the transmission speed of the network links, and to minimize the protocol processing overhead, ATM performs the switching of cells in hardware switching fabrics, unlike traditional packet switching networks, where switching is largely performed in software.

A large number of designs have been proposed and implemented for ATM switches. While many di erences exist, ATM switch architectures can be broadly classi ed into two categories, Asynchronous Time Division (ATD) and Space Switched Architectures.

3.1. Asynchronous Time Division Switches

ATD, or single path, architectures provide a single, multiplexed path through the ATM switch for all cells. Typically a bus or ring is used. Figure 1.7. shows the basic structure of the ATM switch proposed in. In this figure, four input ports are connected to four output ports by a time division multiplexing TDM bus. Each input port is allocated a fixed time slot on the TDM bus, and the bus is designated to operate at a speed equal to the sum of the incoming bit rates at all input ports. The TDM slot sizes are fixed and equal in length to the time it takes to transmit one ATM cell. Thus, during one TDM cycle the four input ports can transfer four ATM cells to four output ports.

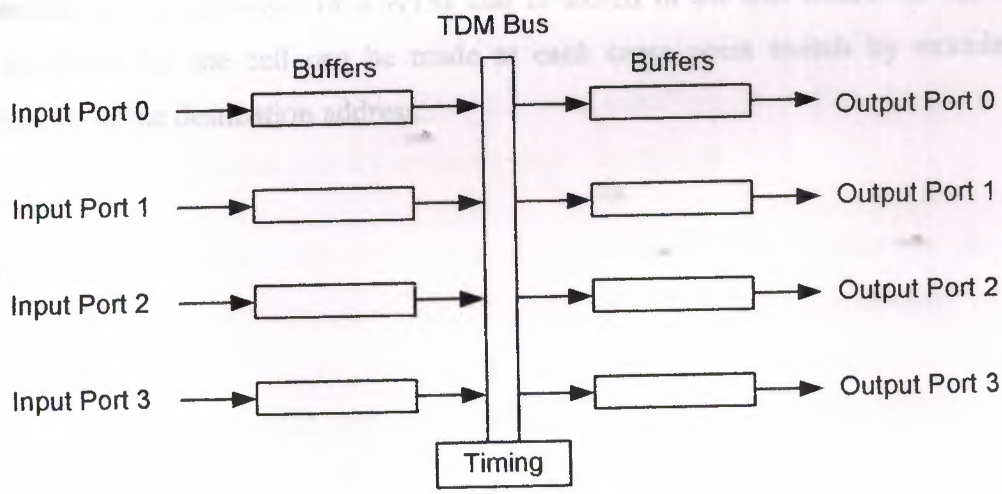


Figure 1.7 A 4 x 4 Asynchronous Time Division Switch.

In ATD switches, the maximum throughput is determined by a single, multiplexed path. Switches with N input ports and N output ports must run at a rate N times faster than the transmission links. Therefore, the total throughput of ATD ATM switches is bounded by the current capabilities of device logic technology. Commercial examples of ATD switches are the Fore Systems ASX switch and Digital's VNswitch.

1.2. Space Division Switches

To eliminate the single path limitation and increase total throughput, space division ATM switches implementmultiple paths through switching fabrics. Most space division switches are based on multi stage interconnection networks, where small switching elements (usually 2 x 2 cross point switches) are organized into stages and provide multiple paths through a switching fabric. Rather than being multiplexed onto a single path, ATM cells are space switched through the fabric. Three typical types of space division switches are described below.

Banyan Switches: Banyan switches are examples of space division switches. An N x N Banyan switchis constructed by arranging a number of binary switching elements into several stages (log $_2$ N stages). Figure 1.8 depicts an 8 x 8 self routing Banyan switch. The switch fabric is composed of twelve 2 x2 switching elements assembled into three stages. From any of the eight input ports, it is possible to reach all of the eight output ports. One desirable characteristic of the Banyan switch is that it is self routing. Since each cross point switch has only two output lines, only one bit is required to specify the correct output path. Very simply, if the desired output addresses of a ATM cell is stored in the cell header in binary code, routing decisions for the cell can be made at each cross point switch by examining the appropriate bit of the destination address.

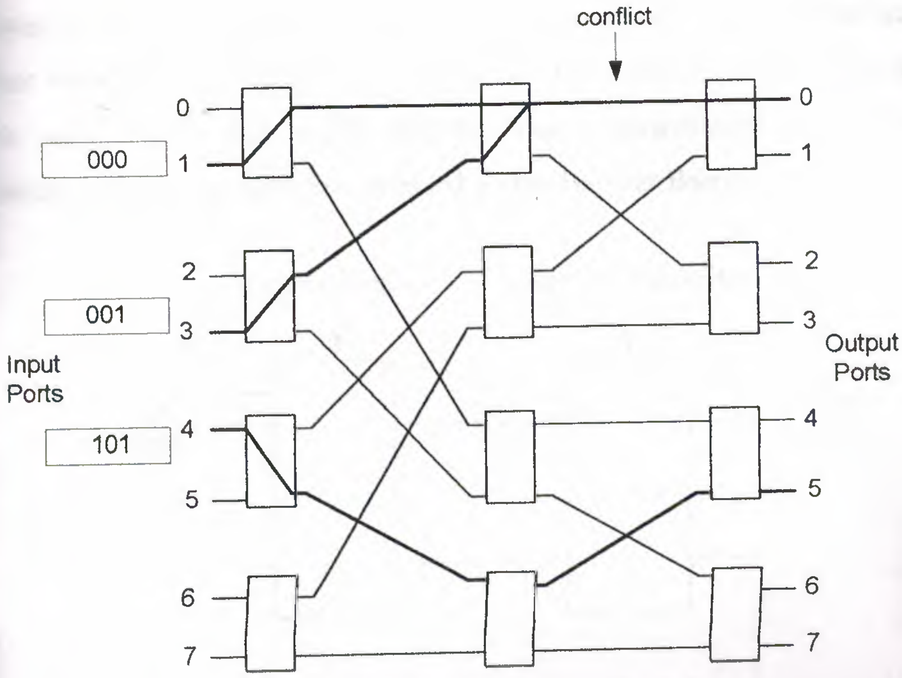


Figure 1.8 An 8 x 8 Banyan Switch With Binary Switching Elements.

Although the Banyan switch is simple and possesses attractive features such as modularity which makes it suitable for VLSI implementation, it also has some disadvantages. One of its disadvantages is that it is internally blocking. In other words, cells destined for di erent output ports may contend for a common link within the switch. This results in blocking of all cells that wish to use that link, except for one. Hence, the Banyan switch is referred to as a blocking switch. In Figure 1.8, three cells are shown arriving on input ports 1, 3 and 4 with destination port addresses of 0,1 and 5, respectively. The cell destined for output port 0 and the cell destined for output port 1 end up contending for the link between the second and third stages. As a result, only one of them (the cell from input port 1 in this example) actually reaches its destination (output port 0), while the other is blocked.

Batcher Banyan Switches: Another example of space division switches is the Batcher Banyan switch. It consists of twomulti stage interconnection networks: a Banyan self routing network, and a Batcher sorting network. In the Batcher Banyan switch the incoming cells first enter the sorting network, which takes the cells and sorts them into ascending order according to their output addresses. Cells then enter the Banyan network, which routes the cells to their correct output ports.

As shown earlier, the Banyan switch is internally blocking. However, the Banyan switch possesses an interesting feature. Namely, internal blocking can be avoided if the cells arriving at the Banyan switch's input ports are sorted in ascending order by their destination addresses. The Batcher Banyan switch takes advantage of this fact and uses the Batcher soring network to sort the cells, thereby making the Batcher Banyan switchinternally non blocking. The Starlite switch, designed by Bellcore, is based on the Batcher Banyan architecture.

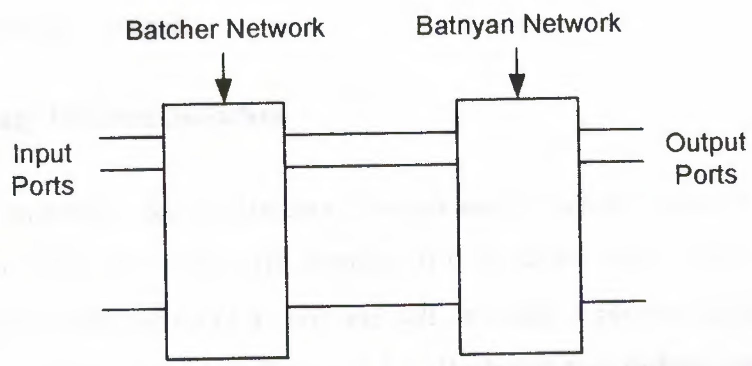


Figure 1.9 Batcher Banyan Switch.

Crossbar Switches: The crossbar switchinterconnects N inputs and N outputs into a fully meshed topology; that is, there are N^2 cross points within the switch. Since it is always possible to establish a connection between any arbitrary input and output pair, internal blocking is impossible in a crossbar switch.

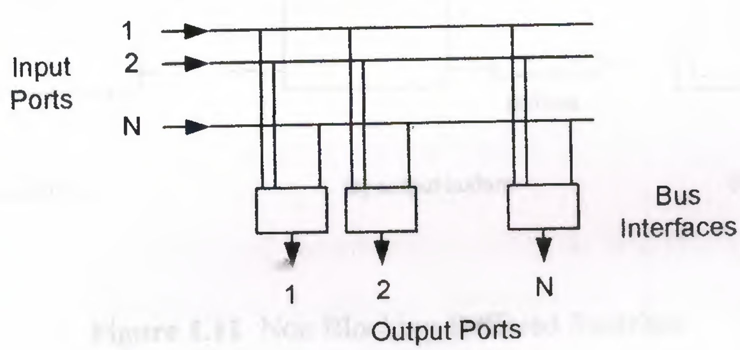


Figure 1.10 A Knockout Crossbar Switch.

The architecture of the crossbar switch has some advantages. First, it uses a simple two state cross point switch (open and connected state) which is easy to implement. Second, the modularity of the switch design allows simple expansion. One can build a larger switch by simply adding more cross point switches. Lastly, compared to Banyan based switches, the crossbar switch design results in low transfer latency, because it has the smallest number of connecting points between input and output ports .One disadvantage to this design, however, is the fact that it uses the maximum number of crosspoints (cross point switches) needed to implementan N x N switch.

The Knockout Switch by ATT Bell Labs is a non blocking switch based on the crossbar design. It has N inputs and N outputs and consists of a crossbar based switch with a bus interface module at each output.

3.3 Non Blocking Buffered Switches

Although some switches such as Batcher Banyan and crossbar switches are internally non blocking, two or more cells may still contend for the same output port in a non blocking switch, resulting in the dropping of all but one cell. In order to prevent such loss, the buffering of cells by the switch is necessary. Figure 1.11. illustrates that buffers may be placed (1) in the inputs to the switch, (2) in the outputs to the switch, or (3) within the switching fabric itself, as a shared buffer. Some switches put buffers in both the input and output ports of a switch.

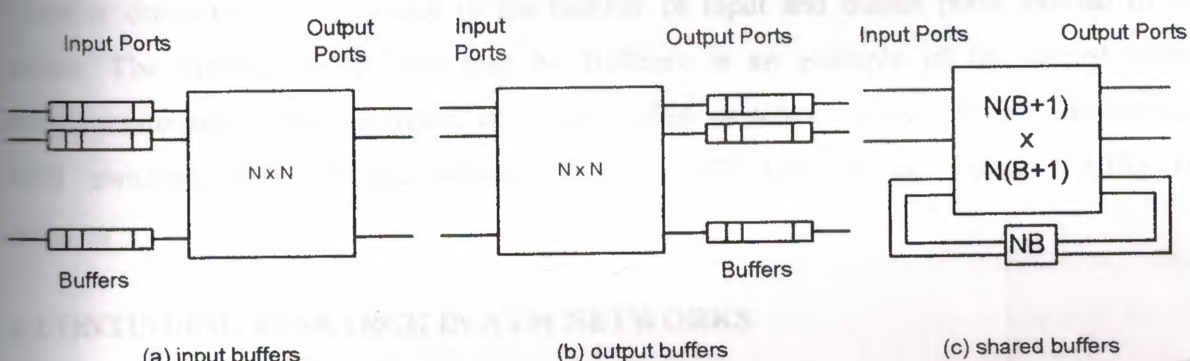


Figure 1.11 Non Blocking Buffered Switches.

The first approach to eliminating output contention is to place buffers in the output ports of the switch. In the worst case, cells arriving simultaneously at all input ports can be destined for a single output port. To ensure that no cells are lost in this case, the cell transfer must be performed at N times the speed of the input links, and the switchmust be able to write N cells into the output buffer during one cell transmission time. Examples of output buffered switches include the Knockout switchbyAT & T Bell Labs, the Siemens & Newbridge MainStreetXpress switches the ATML's VIRATA switch, and Bay Networks' Lattis switch.

The second approach to buffering in ATM switches is to place the buffers in the input ports of the switch. Each input has a dedicated buffer, and cells whichwould otherwise be blocked at the output ports of the switch are stored in input buffers. Commercial examples of switches with input buffers as well as output buffers are IBM's Nways switches, and Cisco's Lightstream 2020 switches.

A third approach is to use a shared buffer within the switch fabric. In a shared buffer switch there is no buffer at the input or output ports. Arriving cells are immediately injected into the switch and when output contention happens, the winning cell goes through the switch, while the the losing cells are stored for later transmission in a shared buffer common to all of the input ports. Cells just arriving at the switchjoin buffered cells in competition for available outputs. Since more cells are available to select from, it is possible that fewer output ports will be idle when using the shared buffer scheme. Thus, the shared buffer switch can achieve high throughput. However, one drawback is that cells may be delivered out of sequence, because cells that arrived more recently may win over buffered cells during contention.

Another drawbackis the increase in the number of input and output ports internal to the switch. The Starlite switch with trap by Bellcore is an example of the shared buffer switcharchitecture. Other examples of shared buffer switches include Cisco's Lightstream 1010 switches, IBM's Prizma switches, Hitachi's 5001 switches, and Lucent's ATM cell switches.

4. CONTINUING RESEARCH IN ATM NETWORKS

ATM is continuously evolving, and its attractive ability to support broadband integrated services with strict quality of service guarantees has motivated the integration of ATM and existing widely deployed networks. Recent additions to ATM research and technology include, but are not limited to, seamless integration with existing LANs emulation, effcient support for traditional Internet IP networking, and further developmentof flow and congestion control algorithms to support existing data services. Research on topics related to ATM networks is currently proceeding and will undoubtedly continue to proceed as the technology matures.

CHAPTER 2

DYNAMIC BANDWIDTH MANAGEMENT IN ATM NETWORKS

1. INTRODUCTION

The aim of the REFORM project is to specify, implement and test a reliable system that offers ATM, multi-class, switched services. Generally speaking, network reliability entails, network survivability and network availability. Network survivability refers to the necessary functions to guarantee a continuous service for established connections in cases of failures occurring within the network. Network availability refers to the optimal configuration and operation of the network at all times, to accept successfully the highest potential amount of new service requests. Within the REFORM system, network survivability is implemented by means of an ATM layer protection switching mechanism. This mechanism targets at the reconfiguration of the VP layer infrastructure by switching the failed VPCs to standby (predetermined) alternative VPCs. The full methodology, including restoration resource control protocols and network reconfiguration algorithms that go along with this mechanism are documented in but this aspect of the REFORM system is not the subject of this paper. Network availability is concerned with the cost-effective planning and maintenance of network resources so that to maximise user connection admissions. The planning aspect is not covered by this paper but the project has described suitable VP layer design algorithms needed to originally configure the ATM layer (given a physical network) so as to meet the traffic predictions. These algorithms include the configuration of the necessary protection resources required by the protection switching mechanism.

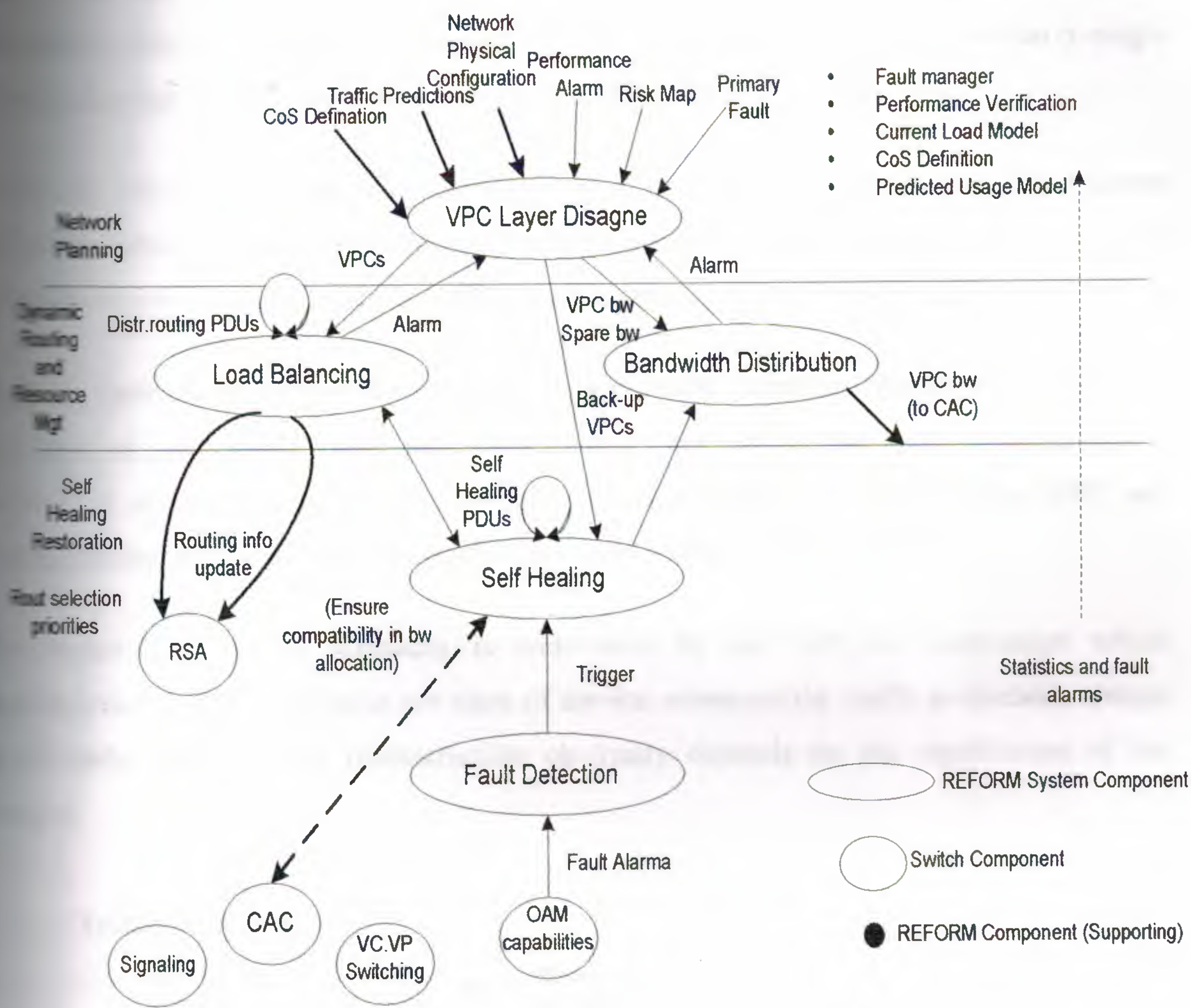


Figure 2.1 REFORM Functional Model.

For the purposes of this report we can consider that network operation is generally decomposed into two distinct operational phases. During the initialisation phase, the network is prepared for service provisioning at a certain service level. During the normal phase, the network delivers services, and sees to the active management of its resources so as to guarantee its service levels under deviations of offered traffic at its edges. This paper is concerned with the dynamic management of bandwidth during the normal phase according to network designs created during the initialisation phase. The initialisation phase results in the definition of a suitable network of working VPCs (for carrying user traffic) and admissible routes based on them per source-destination and Class of Service 1 (CoS) so that to preserve the performance characteristics of each CoS. Furthermore, for the network to cope gracefully (without affecting the integrity of existing services and its availability to future services) with fault conditions, protection VPCs need also to be planned and the appropriate restoration bandwidth need to be allocated. The initialisation activities are undertaken within a single functional component, called VPC Layer Design (VPC_LD).

Since user behaviour changes dynamically there is a chance that the network may become unbalanced when the bandwidth allocated to VPCs on the existing admissible routes are not in accordance with the quantity of the traffic that it is actually offered to be routed over them.

There are basically two levels at which adaptivity to traffic variations should be provided, one at a level of (structural) traffic prediction changes and one at the level of actual traffic fluctuations around the predictions. Therefore, it is reasonable to consider that VPC and routing management is achieved through a two level hierarchy (see Figure 2.2).

The higher level of the hierarchy is undertaken by the VPC_LD component which reconfigures the VPC and routes per class of service whenever the traffic predictions change significantly. The level of reconstruction obviously depends on the significance of the changes.

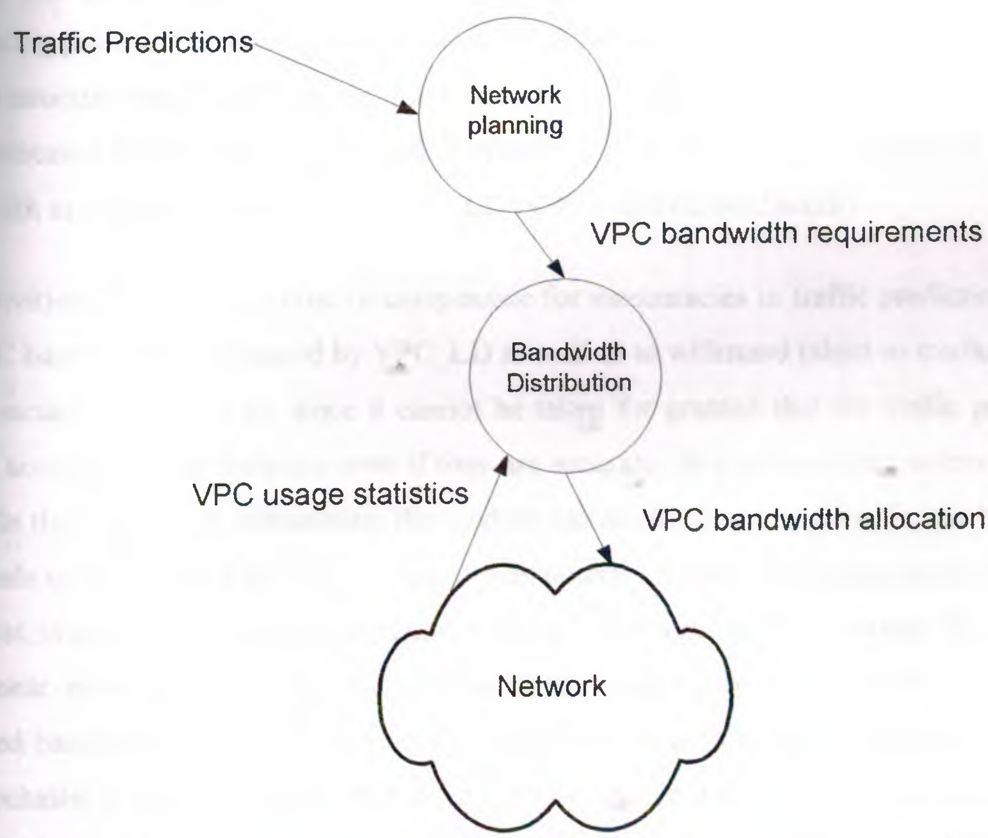


Figure 2.2 Hierarchical Approach to Bandwidth Management.

The lower level in the hierarchy is introduced to compensate for inaccuracies in the traffic predictions and short term fluctuations in actual load around the predictions. The lower level functionality operates on the set of existing routes and redefines the working VPC bandwidth and route selection parameters.

The above discussion indicates that two further functional components need to play part in the resource and routing management hierarchy: Bandwidth Distribution (for updating working VPC bandwidth) and Load Balancing (for updating route selection parameters). This report discusses the objectives, design constraints and proposes an algorithm for the first of these two components, Bandwidth Distribution.

2. BANDWIDTH DISTRIBUTION

The Bandwidth Distribution (BD) component is responsible for the management of the bandwidth allocated to working VPCs according to actual traffic conditions. That is, it adjusts the bandwidth allocated to the working VPCs to their actual usage to avoid situations where in the same links some VPCs tend to become over-utilised while other VPCs remain under utilised. The dynamic management of the working VPC allocated bandwidth is achieved by distributing portions (viewed as a common pool) of the link working bandwidth (link capacity minus restoration bandwidth) among the working VPCs. Specifically, the management of the VPC allocated bandwidth is done within specific (upper and lower) bounds on the VPC bandwidth as originally estimated by VPC_LD (VPC required bandwidth).

The activities of BD are required to compensate for inaccuracies in traffic predictions and in the VPC bandwidth as estimated by VPC_LD as well as to withstand (short to medium) actual traffic variations. This is so, since it cannot be taken for granted that the traffic predictions will be accurate and furthermore even if they are accurate, they are accurate within statistical range. In this respect i.e. considering the random nature of the arriving traffic, the bandwidth that needs to be allocated to VPC, so that certain objectives (regarding connection admission) to be met, is a stochastic variable, depending on the connection arriving pattern. The VPC_LD component estimates originally the bandwidth that needs to be allocated to the VPCs (required bandwidth) so that satisfy traffic predictions. This is viewed as the mean value of the (stochastic in nature) bandwidth that needs to be allocated to the VPCs. It is the task then of the BD component, to manage the allocated bandwidth of VPCs, around the (mean) required bandwidth, according to actual traffic conditions.

By monitoring the usage on working VPCs, the BD component also emits warnings to VPC_LD indicating insufficient usage of the planned network resources. The warning are issued in cases where some VPCs remain under-utilised (with respect to their required bandwidth as specified by VPC_LD) for a significant period of time. This implies that these resources cannot be utilised in the routes by the Load Balancing component and therefore such cases are interpreted as indicating overestimation of network resources.

The proposed algorithm for bandwidth redistribution assumes that there is a common pool of bandwidth per link to be redistributed to the VPCs when necessary. The algorithm assumes that this common pool of bandwidth per link is the links' unallocated bandwidth. Note, that by its definition, this pool of bandwidth is not totally allocated to the VPCs at any instant, but it is there to be allocated to the VPCs that go highly utilised only when such conditions occur. Each VPC grabs or returns portions of its allocated bandwidth to the common pool of bandwidth according to its congestion level. We assume here that the modification of the bandwidth of a VPC does not impact on the traffic parameters (QoS) of the VCs using this VPC or other VPCs sharing the same links or nodes.

2.1. Possible Approaches

In distributing resources we have three main approaches.

2.1.1. Generous

The generous approach in the BD context would consist in allocating as much bandwidth possible to every VPC in order that for example all VPCs have an equal share of the spare bandwidth. In fact this scheme would be very reasonable except for one thing: It would make creation of new VPC, and thus reconfiguration impossible or very difficult.

2.1.2. Greedy

In this strategy we allocate "just enough" bandwidth to each VPC, so to maintain spare link bandwidth for new VPCs. If you say that "just enough" is the used VPC bandwidth + 20% then a small VPC might not be able to accept any new VC. And a big VPC will have a large spare bandwidth which will imply a high acceptance potential and might lead to an unbalanced network. If you say that "just enough " is the used VPC bandwidth + Constant, then you have to be careful that this constant is large enough to allow creation of new VC and not too large to avoid wasting resources.

2.1.3. Fair

Fairness here means taking into account information and the dynamics of the other components of the system, here mainly from VPC_LD, Reconfiguration and LB.

1. VPC_LD provides BD with the required bandwidth for each VPC: Vreq.. Either:

- Hypothesis 1: Vreq is only an initial value.
- Hypothesis 2: The traffic predictions shows that Vreq (or an interval around Vreq) is the probable bandwidth needed at some point by this VPC. In that case even if the used bandwidth is low, BD shouldn't allocate the VPC less bandwidth than Vreq.

2. Reconfiguration needs spare link bandwidth.

3. LB is most concerned with the overall distribution of spare bandwidth within the VPCs, since it determines how the VC will be routed. Either;

- It is important that LB should try to fulfil the prediction made by VPC_LD. In that case even if a VPC is empty you should allocate him Vreq so that LB will have the incentive to give him traffic.
- VPC_LD only provides a topology. In that case BD should shape the spare bandwidth resources so that the spare bandwidth is evenly distributed between the different VPCs.

2.2. Chosen Solution

The approach consists in defining for each VPC an Admissible Zone AZ in which the VPC bandwidth is allocated. This Admissible Zone is defined around the Vreq value given by the VPC_LD component. BD is not allowed to allocate bandwidth outside these boundaries. We also define the operating point of the VPC as the last measured Vused send to BD by NRM, this is the freshest actually measured used bandwidth that BD knows. Around the operating point of the VPC we define a working zone WZ which defines an upper and lower threshold. Above this working zone we define a buffer zone BZ.

The general rule of operation goes as follows: Each time the bandwidth used crosses one of the working zone boundaries an alarm is raised. BD then centres the working zone around the new operating point, and allocates the VPC a new bandwidth equal to the upper boundary of the working zone augmented by the buffer zone. Special cases occurs when this bandwidth reaches the upper limit of the admissible zone or when this bandwidth is not available because of link capacity restrictions along the VPC path or its protection path. Other special cases occurs when reaching the lower bound of the admissible zone.

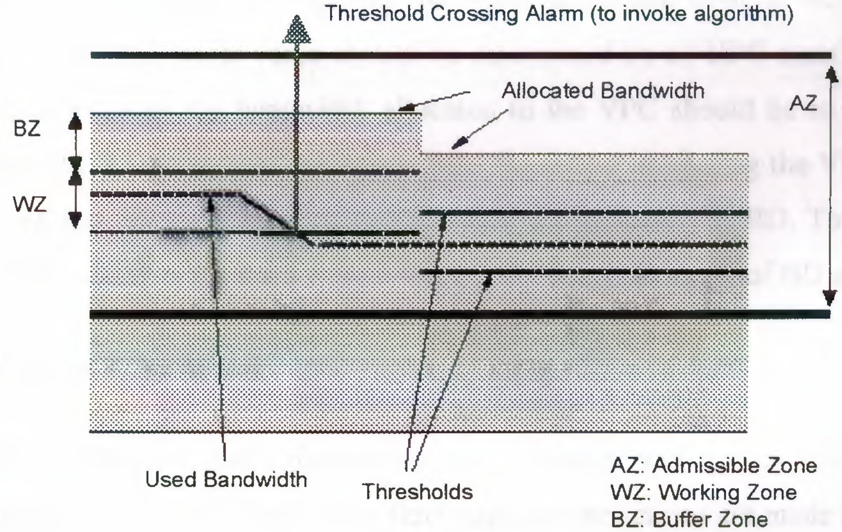


Figure 2.3 Tracking Used Bandwidth on VPCs.

Advantages seen to the moving working zone and buffer zone approach:

- A VPC wants more bandwidth but we can't allocate any to him: The WZ being centred around Vused, BD will not repeatedly receive alarms since the upper boundary of the WZ will be above Valloc. What should be done is to store the fact that this VPC needs more bandwidth as soon as possible, in a list of pending issues, until the full WZ and BZ can be allocated.

- If the VPC is empty or has very low traffic, again BD will not receive repeated alarms since the lower boundary of the WZ will be negative (or zero).
- We have a fairly good idea of how much unutilised bandwidth there is on each link: spare link bandwidth + protec bandwidth + sum(BZ) + sum(1/2WZ).
- The unutilised bandwidth in each VPC is approximately constant: BZ+1/2WZ and bounded: max=BZ+WZ, min=BZ.

2.2.1. The Admissible Zone

The Admissible Zone of one VPC defines the upper and lower limit of the VPC allocated bandwidth. Its upper and lower value should be determined by a "VPC class". The width of AZ determines how close the bandwidth allocated to the VPC should be to the Vreq value determined by VPC_LD, a narrow AZ means little flexibility in altering the VPC's bandwidth while a wide AZ means that Vreq shouldn't be taken too seriously by BD. The exact use and range of the "VPC class" is for further study during the detailed design of BD and VPC_LD.

2.2.2. The Working Zone Width

The width of the Working Zone determines the frequency with which threshold crossing alarms will be generated. The Vused value fluctuates as connections are made and dropped on the VPC in question. The magnitude and frequency of the fluctuations depends on the bandwidth of the VCs and on the connection arrival and drop patterns (which is influenced by, the arrival and drop rates to the network in terms of CoS and source-destination pairs; the admissible routes defined for VCs of a particular CoS, and the behaviour of load balancing in selecting routes for VCs among the set of admissible ones).

The width of WZ is therefore dependent on two factors: the nature of the fluctuations in the Vused signal; and the required sensitivity of BD to used bandwidth changes. Regarding the Vused fluctuations, exact traffic models are not known at the time of writing. Further work on traffic simulation is required under varying conditions of sets of admissible routes and LB behaviour.

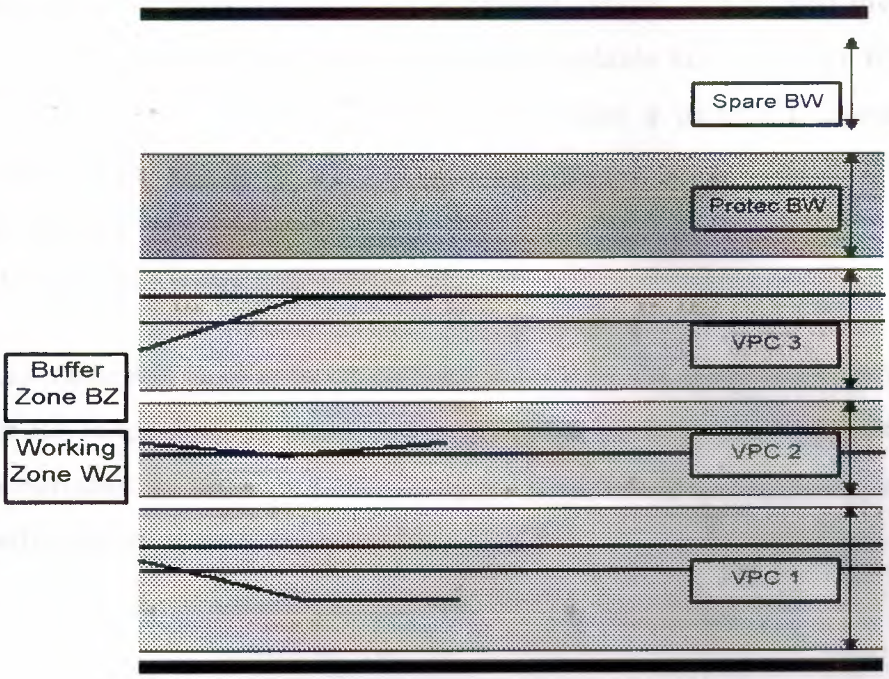


Figure 2.4 Distributing Link Capacity to VPCs.

Given a specific model of Vused variations over time (to be established experimentally) we can consider the second factor, the required sensitivity of BD to variations in Vused. If WZ is made small then a large number of threshold crossings will take place, requiring BD to be invoked frequently. On the one hand this is positive in the sense that it will cause the bandwidth allocated to VPCs to closely follow the Vused value thereby ensuring that link capacity isn't needlessly allocated to VPCs. On the other hand this is negative in the sense that it will increase the quantity of management traffic (alarms) and the processing overhead of BD. If WZ is made large the number of alarms and hence the processing load on BD will be reduced but at the cost of a less accurate reflection of Valloc to Vused. There is clearly a trade off between these two approaches and this will be experimented with during the testing phase of BD both via simulations and testbed runs.

2.2.3. The Buffer Zone Width

The width of the Buffer Zone is related to the widest bandwidth of the VC class supported by the VPC and will have to be given by a VPC class parameter. In fact the Buffer Zone has to be at least the widest width of the VC supported by the VPC to ensure that the VPC which is has a Vused of slightly below the upper WZ threshold can accept (at least) one more VC.

This VC will cause the upper threshold to be crossed, which, in turn, will invoke the BD algorithm to increase the VPC bandwidth (if there is available link capacity). If the interval between the time the VC is accepted and the time when a successful increase in VPC bandwidth has taken place is longer than the inter-connection arrival time then VCs may be blocked if the BZ isn't wide enough to accommodate the expected VC arrivals in that period. In this case BZ should be wider than a single VC.

The exact value will have to be determined experimentally when traffic arrival patterns are known. If BZ is too small it could lead to unacceptable Cell Blocking Rates (although these may be mitigated by alternate routes using different VPCs). If BZ is too large it would lead to an inefficient use of the link bandwidth.

CHAPTER 3

A FRAMEWORK FOR BANDWIDTH MANAGEMENT IN ATM NETWORKS AGGREGATE EQUIVALENT BANDWIDTH ESTIMATION APPROACH

1. INTRODUCTION

Traffic control and bandwidth management in ATM networks has attracted a lot of attention due to the potential for improved utilization through statistical multiplexing of traffic sources. In spite of considerable research effort, no universal solution to this problem has been found. The proposed approaches usually are suited to a particular type of traffic and in general can be divided into two categories. In the first one, the focus is on restricting the traffic entering the network based solely on the source type and declarations, regardless of the current level of traffic generated by the sources. A typical example is the equivalent bandwidth allocation based on source declarations and policing mechanisms. The second category covers algorithms which include some kind of dependence on the current state of the network. Congestion notification are examples of the second type. The limitations of the first category are source restrictions and design for the worst-case (potential for low utilization). The second category is more flexible (potential for high utilization) but does not provide guarantees for throughput nor for quality-of-service (QoS).

These two approaches constitute almost two disjoint worlds. They are focused on different traffic sources (real-time versus controllable data) and the key elements of the algorithms have no common protocol or data base structure. We propose a framework for traffic control and bandwidth management which bridges these two worlds. We start from a simple observation that since the congestion occurs at the switch output ports, any efficient and flexible traffic control mechanism should have access to some information about the current cell process in the switch output ports. Note that this is the case in most of the congestion control mechanisms for controllable data traffic where the switch output buffer states are measured directly (congestion notification mechanisms) or indirectly (window mechanisms). We argue that direct measurement of the cell process in the switch output port can constitute a common base for traffic control and bandwidth management of both real-time and controllable data services.

For the controllable data services we adopt the rate-based approach recommended by the ATM Forum. This adaptive scheme explicitly regulates the source rates by means of the switch output buffer state measurements and resource management cells. For the real-time services we propose an approach based on estimation of the aggregate equivalent bandwidth required by all connections served by each of the switch output ports. In order to use all available relevant information, the estimation procedure employs the source parameter declarations and direct measurement of the connection superposition cell process parameters at the switch output ports. In this scheme, the connection admission procedure can verify the available bandwidth on all considered links by means of a resource management cell passing through the transit nodes. One of the key advantages of the proposed framework is that the traffic control and bandwidth management algorithms for real-time and controllable data services are based on the same data base structure and signaling protocol. Besides reduction in algorithms complexity and cost, this feature can also increase bandwidth utilization by enabling coordination of both algorithms. Another important feature of the proposed scheme is that it is not restricted to any particular link or source model. In other words, any reasonable model for analytical evaluation of the connection equivalent bandwidth can be used in this framework.

In the literature, one encounters other connection admission algorithms based on measurements. They differ in many aspects from our framework. For example, the mechanism described in measures individual source process parameters which are then used to adapt source policing mechanism parameters and the equivalent bandwidth required by the connection. On the other hand, the algorithm proposed in uses measurements of the superposition cell process to estimate the cell arrival distribution but does not operate in the equivalent bandwidth domain. In and, Bernoulli link models and Bayesian decision theory are used for connection admission based on the measured link load. It is observed that measurements of the cell process in the low frequency band can constitute a basis for adaptive connection admission control.

In the second part of the paper, we design and analyze a particular model for connection admission based on aggregate equivalent bandwidth estimation in the switch output ports. The estimation process is decomposed into two parts. First the algorithm estimates certain parameters of the link connection superposition process. Then, based on the estimated and declared parameters, the aggregated equivalent bandwidth is estimated. The estimation of the chosen parameters (mean and variance of the instant cell rate process) employs a two-state near Kalman filter. The important feature of the Kalman filter model is that it provides information about the estimation error in terms of the declaration and measurement errors. This information is used to evaluate bandwidth reserved for estimation errors in order to provide statistical guarantees for the QoS. The reserved bandwidth enables the source parameter declarations to be more relaxed and the source policing less stringent compared to equivalent bandwidth allocation based solely on the source parameter declarations. Concerning the ATM Forum and ITU recommendations for source parameter declaration, the proposed approach can be seen as complementary. In particular, the predicted connection traffic parameters, if not declared, can be estimated by the network operator based on the declared policing parameters, source type and long term statistics concerning this type of source.

The accuracy of the estimation process is analyzed in three stages. First, we concentrate on the evaluation of the measurement error. Then we analyze the errors of estimates provided by the Kalman filter. Finally, we verify the accuracy of the model used for estimation of the aggregate bandwidth and the bandwidth reserved for the estimation error. The results show that, for the Gaussian declaration model, the estimation process is very accurate. The accuracy of the whole connection admission algorithm is verified under nonstationary conditions and large, non-Gaussian, declaration errors. The results demonstrate that the algorithm copes very well with unpredicted changes in source parameters by providing high bandwidth utilization and the required QoS. We can extend the estimation model to take into account the in uence of source policing mechanisms. The numerical study of this option illustrates the trade-off between strict and relaxed source policing.

2. FRAMEWORK FOR UNIFIED TRAFFIC CONTROL AND BANDWIDTH MANAGEMENT

The issue of traffic control and bandwidth management in ATM based networks is complex due to a mixture of different connection traffic types, QoS requirements and time scales. In this section, we describe a framework which provides a coherent and effective structure for traffic control and bandwidth management of all services.

We start from a classification of the service categories. According to the ATM-Forum recommendations, real-time variable bit rate (VBR-RT), non-real-time variable bit rate (VBR-NRT), available bit rate (ABR), and unspecified bit rate (UBR). From the bandwidth management and traffic control viewpoint, these categories can be aggregated into two macro-categories: controllable traffic parameters (CTP = ABR) and noncontrollable traffic parameters (NCTP = CBR + VBR-RT + VBR-NRT). Notethat the UBR category is not included in the macro-categories. This follows from the fact that UBR services have smallest priority on the cell layer and do not require any additional traffic admission control or resource allocation.

In the following, we present a framework which, to alarge extent, can unify data base structure, signaling protocols and algorithms for traffic control and bandwidth management of NCTP and CTP services. There are four principles from which the framework is derived. The algorithms should operate explicitly in terms of the bandwidth so the cooperation between bandwidth management algorithms on the connection and higher layers is straightforward. All available information should be used in an optimal way (in particular, it means that the algorithms should use explicit measurements of the relevant network state and source traffic parameters declarations). The algorithms should not be restrictive in the sense that even an unpredicted traffic increase should not be rejected if there is available bandwidth. Finally, the algorithms for different service categories (CTP and NCTP) should be based on the same data base structure and similar signaling protocols in order to reduce complexity and cost of traffic control and bandwidth management.

We adopt the ATM Forum recommendation for traffic control of the CTP (ABR) service class. The basic idea of this rate-based recommendation is illustrated in Fig. 3.1(a). During a connection, at least every 100 ms, the source sends to the destination a resource management (RM) cell with information about the current cell rate, CCR. At each node, the bandwidth management algorithm evaluates the desired rates (DR) for each virtual channel (VC) based on the declared CCR's (received from RM cells) and measurements of the cell process parameters, Z, in the switch output ports used by VC's. The desired rates are inserted into the passing RM cells which are returned to the source by the destination node. If the RM cell is lost, the source automatically reduces the rate. Note that one can establish a simple relation between the connection rate and the equivalent bandwidth required by the connection. Thus, the rate-based algorithm is in accordance with the first three adopted principles since it can operate in.

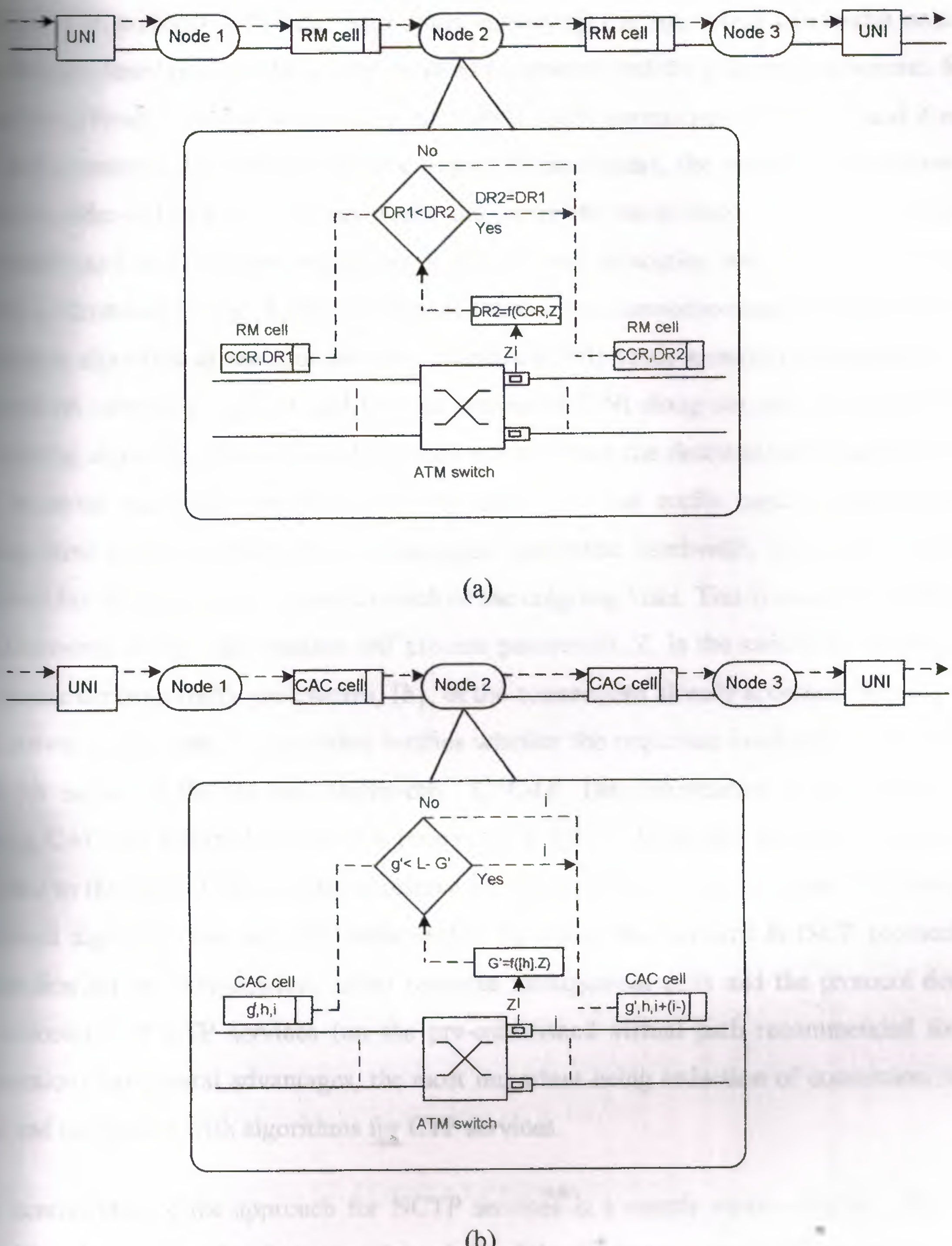


Figure 3.1 TR&BM Mechanism for (a) CTP Services and (b) NCTP Services.

the bandwidth domain, uses source declarations and explicit measurements of the relevant network state, and allows a rate increase whenever there is available bandwidth.

In the literature, one finds several propositions for logical bandwidth allocation to virtual channels for the NCTP category of services. The most attractive are the ones employing equivalent bandwidth allocation since in this case the gain from statistical multiplexing can be realized. The problem with most algorithms employing the equivalent bandwidth notion is that they are based only on the source declared parameters and the policing mechanism. Since it can be difficult for some sources to predict their traffic parameters in advance and it might be hard to enforce declarations of some statistical parameters, the scheme is restrictive and prone to under-utilization of the bandwidth (design for the worst-case). In order to avoid these drawbacks and fulfill all the objectives of the adopted principles, we propose an approach which is illustrated in Fig. 3.1(b). In response to the new connection request, the connection admission algorithm at the user-network interface (UNI) sends a resource management cell (henceforth referred to as CAC cell) to the destination UNI along the path recommended by the routing algorithm. The cell contains information about the declared traffic parameters, h, and required equivalent bandwidth, g'. In each node, the traffic control and bandwidth management algorithm estimates the aggregate equivalent bandwidth, G', which should be reserved for all connections carried on each of the outgoing links. This is carried out using the measurements of the superposition cell process parameters, Z, in the switch output ports and the source declared traffic parameters, {h}, of the connections already accepted. When a CAC cell arrives at the node, thealgorithm verifies whether the requested bandwidth, g', is smaller than (or equal to) the residual bandwidth, Ç=L-G'. This information is inserted into the passing CAC cell and the bandwidth is reserved if $g' \leq C$. At the destination UNI, the cell is returned to the origin UNI in order to reserve the bandwidth on the return path. Obviously the proposed algorithm can also be implemented by using the standard B-ISUP protocol for connection set-up. Nevertheless, using resource management cells and the protocol derived for owcontrol of CTP services (on the pre-established virtual path recommended for the connection) has several advantages, the most important being reduction of connection set-up time and integration with algorithms for CTP services.

The central idea of the approach for NCTP services is a simple observation that the CAC algorithm does not need to know precise values of the equivalent bandwidth required by each individual connection. On the contrary, it is the cell superposition process which defines the quality of service and that is why the aggregate equivalent bandwidth is employed as the main control variable. Another advantage of this approach is that the scheme can be much less restrictive compared to the schemes based on policing mechanisms. This is caused by two factors, namely, the statistical multiplexing of the bandwidth allocation errors and fast adaptability. The first factor concerns the problem of the bandwidth reservation for the worst-case when a policing mechanism enforces bandwidth allocation to the source. This reservation has to take into account possible errors in declared traffic parameters. Although one can claim that the same problem exists with the aggregated bandwidth, it can be easily shown that the standard deviation of the aggregated process parameter error is in general significantly smaller than the sum of the standard deviation of individual connection traffic parameter errors. The second factor relates to the fact that measurement of the cell superposition process is much more reliable than the sum of each connection process measurements. This feature ensures that even if there is a large declaration error it will be quickly reduced by adjusting the bandwidth allocation. Thanks to both effects, the source policing can be significantly relaxed and the interpretation of its function is changed from bandwidth enforcement to controlling the magnitude of the aggregate bandwidth estimation error.

Comparison of the traffic control and bandwidth management structures for CTP and NCTP services (Fig. 3.1) shows that both approaches employ the same data base structure and signaling protocol. In addition to significant reduction of the bandwidth management algorithm complexity and cost, this feature can also increase bandwidth and buffer utilization by close cooperation of both algorithms. For example, the explicit rate allocation algorithms for CTP connections would be more efficient and stable if parameters of the aggregated cell process of NCTP services could be predicted in advance. Since this prediction is an inherent feature of the proposed connection admission for NCTP services, both algorithms may use a common prediction algorithm where some parts would be parameterized according to the needs of each of the two applications. In the remainder of this paper, we consider implementation of the proposed framework forconnection admission for NCTP services only. First, we focus on an aggregate equivalent bandwidth estimation model and its error analysis. Then, the connection admission control mechanism is studied.

3. MODEL FOR AGGREGATE EQUIVALENT BANDWIDTH ESTIMATION

In the proposed framework, an estimate of the aggregate equivalent bandwidth, $\hat{G}$, is a function of the declared pa-rameters of accepted connections, H={ h_j }, and measured parameters, Z, of the connection superposition process at the switch output port. It is obvious that the estimate $\hat{G}$ may be different from the real value, $\hat{G}$, due to the declaration and estimation errors. If we assume that the bandwidth reserved for accepted connections, G', equals $\hat{G}$, the QoS constraints can be violated with high probability. In order to keep this probability at an acceptable level, a bandwidth, R, reserved for the estimation error is introduced so the connection acceptance rule is based on the bandwidth reserved for aggregate traffic defined as;

$G' = \hat{G} + R$.........(1)

Thus the objective of the estimation process is to find $\hat{G}$ and R such that

$P\{ G > \hat{G} + R \} \leq €_1$.........(2)

where $€_1$ is the estimation error constraint.

Note that *from the connection admission point of view, we are interested in discrete points of* time. Thus, the system is modeled in the discrete time domain where t_k, k =1,2... denotes the instant of the system state change, X_k-1→ X_k, caused either by a new connection, or by a connection release. In general, the system state can be defined by some parameters being a function of the cell rate superposition process, S(t). Based on estimation theory, it can be shown that, by applying a recursive discrete filter, the state estimate of our system, $\hat{X}_k$, and the covariance matrix of its error, P_k, can be evaluated as a function of the following parameters: h_k, Z_k, X_{k-1} (where denotes the declared parameters of the connection added or released in the transition $X_{k-1} \rightarrow X_k \rightarrow$). In addition, the parameters, q_k, Y_k defining the declaration and measurement error distributions (assumed to be Gaussian) are required. The issue of $\hat{G}_k$ and R_k estimation fits very well into the framework of recursive discrete filters which suggests that the natural choice for the state description would be the aggregate equivalent bandwidth, G_k, so the required estimates, $\hat{G}_k$, R_k, would be achieved directly. While this approach is possible, in the following we propose another state description which simplifies the estimation process and ensures that it is not limited to a particular model for equivalent bandwidth evaluation.

There are two difficulties with direct estimation of $\hat{G}_k$ The relationship between the equivalent bandwidth and the parameters which can be directly measured, $\hat{G}_k = f_{\hat{G}}(Z_k)$, is in general nonlinear. This suggests that a nonlinear filter should be applied which is typically more complex than a linear filter.

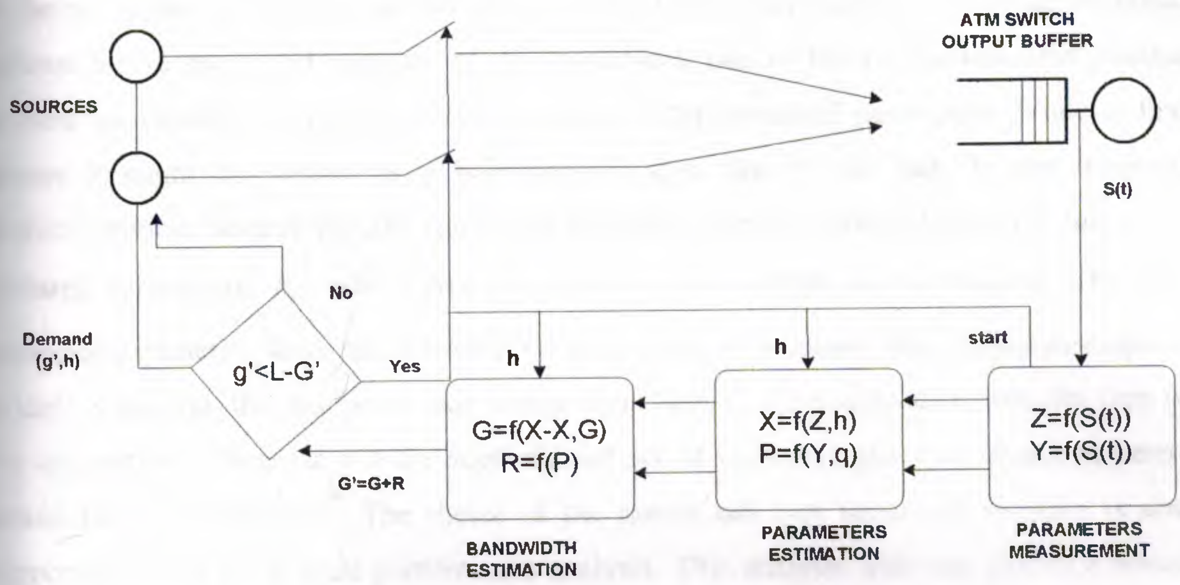


Figure 3.2 Structure of The Control System.

Although in the literature one can find several models for evaluation of the equivalent bandwidth, these algorithms are relatively complex and do not provide a universal closed form solution. These characteristics significantly increase the complexity of the problem, since the function, $f_{\hat{G}}$ must be inverted in the estimation algorithm.

To avoid these problems, we chose a vector of the super-position cell rate process parameters as the state description which can be directly measured at the switch output port. Thus a linear filter can be applied and the inverse function problem becomes trivial. Having the estimate of the system state, X_k, the estimate of aggregate equivalent bandwidth can then be evaluated from a differential approach

$$G_k = \hat{f}_{\Delta\hat{G}}(\hat{X}_k - X_k^d, G_k^d) \quad \text{...........(3)}$$

where X^d, G^d denote the values evaluated from the declared parameters. The differential approach provides that the proposed CAC algorithm can work with any reasonable model for evaluation of the connection equivalent bandwidth since this model is not directly used in the estimation procedure. The structure of the control system based on the differential approach is illustrated in Figure 3.2.

In the remainder of the paper, we describe and investigate a particular version of the proposed scheme where mean and variance of the instant cell rate of the connection superposition process are chosen as the system state description and measured parameters. While at first glance it might be viewed as a bold simplification due to the lack of any temporal characterization, observe that the equivalent bandwidth can be evaluated from the full set of declared parameters, h_k, which take into account all important source features while the estimated parameters serve only to correct this bandwidth allocation. The implied assumption is that, in general, the sources do not change their basic features associated with the type of the connection (otherwise a more sophisticated set of estimated and measured parameters would have to be chosen). The choice of the instant cell rate mean and variance is also supported by the burst scale performance analysis. This analysis indicates that in a robust CAC algorithm for real-time applications the arrival link cell rate should only exceed link capacity with a very small probability while the buffer dimensioning should take care of the cell scale congestion (small time scale variability factor). In this case the temporal characterization of the cell rate process is not critical. Obviously, in the (unlikely) case of very large buffers the chosen parameters should be complemented by a parameter(s) characterizing the cell rate autocorrelation function.

Concerning the source declaration parameters h_k we assume that they include the predicted connection average cell rate, m_k^k, cell rate variance, v_k^d, and variances of these prediction errors, $q_k = \left[v_{e,k}^m, v_{e,k}^v\right]^T$, respectively. These values can be de-clared directly by the source or can be estimated based on the source type, declared policing parameters (peak rate, sustained rate, and burst tolerance), and long term statistics concerning this source type behavior. It should be underlined that the study presented in this paper focuses on the proposed estimation procedure and adaptive connection admission control. Since these algorithms are independent from the model for equiv-alent bandwidth evaluation based on the connection declared parameters, we do not discuss such models and throughout the paper we assume that the function for equivalent bandwidth evaluation is given, $g_i^d = f_g(h_i)$. The reader interested in models for equivalent bandwidth evaluation is referred to a substantial literature on this subject.

3.1. Estimation of The Cell Rate Mean and Variance

The objective of the estimation process is to provide the best estimate of the mean, M_k, and variance, V_k, of the instant cell rate of the connection superposition process. In this paper, we treat these two variables as independent although the proposed approach can also take into account the correlated case. The state of our system is defined as $X_k = [M_K, V_k]^T$ The dynamics of the system model is illustrated in Fig. 3.3 and is described by;

$X_k = X_{k-1} + x_k + e_k$ ....(4)

where, e_k denotes the model error and $\chi_k = [\alpha_m m_k, \alpha_v v_k]^T$ denotes either the declared mean and variance of the accepted connection ($\alpha_m = \alpha_v = 1$) or the normalized declared mean and variance of the released connection ($\alpha_m = -\hat{M}_k \;/\; M_k^d, \alpha_v = -\hat{V}_k \;/\; V_k^d$, where index d denotes parameters evaluated from declarations). For the time being we assume that the model error, $e_k = [\delta_k^m, \delta_k^v]^T$, is a Gaussian random variable with zero mean and covariance matrix, Q_k.

The system model in Figure 3.3 is complemented by the measurement model which provides the measurement of the cell rate mean and variance in state $k, Z_k = [\overline{M}_k, \overline{V}_k]^T$. The delay corresponds to the fact that the result of measurement of the parameters is required at the time of the next state change. The measurement model is defined as follows;

$Z_k = X_k + u_k$ ....(5)

where $u_k = [\overline{\delta}_k^m, \overline{\delta}_k^v]^T$ is the measurement error. The measurement error is assumed to be a Gaussian random variable with zero mean and known covariance matrix, Y_k. The system and measurement model fit very well into the framework of linear recursive filters. In the following;

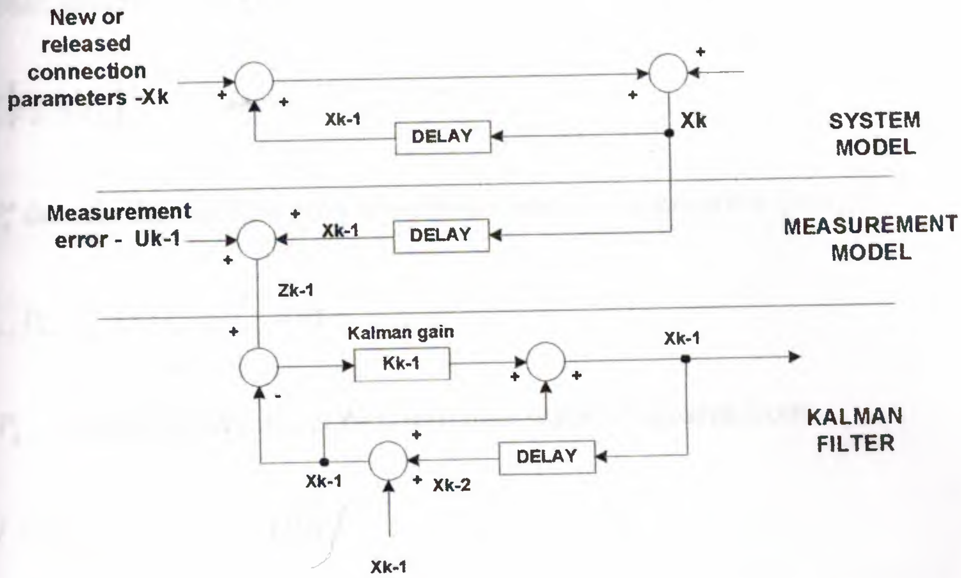


Figure 3.3 System Model and Kalman Filter.

we demonstrate application of the linear discrete Kalman filter to the considered system. In general, the Kalman filter provides an optimal least-square estimate of the system state for a linear system and the model and measurement errors are Gaussian random variables. The block diagram of the applied filter is shown in Figure 3.3. The system state estimate update is given by;

$$\hat{X}_k = \hat{X}_k^e + K_k\left[Z_k - \hat{X}_k^e \right] \text{.............(6)}$$

where $\hat{X}_k^e = \hat{X}_{k-1} + \aleph_k$ denotes the state estimate extrapolation and Kk is the Kalman filter gain. It is clear that the gain is a weight which decides how much confidence should be given to the measurement versus the declaration.

To simplify notation let us introduce the transition matrix, Fk, with the following elements:

$$a_{22} = \frac{V_{k-1} + a_v v_k}{V_{k-1}}, a_{11} = \frac{M_{k-1} + \alpha_m m_k}{M_{k-1}}$$

and $a_{12} = a_{21} = 0$. Thus, the system model can be described by;

$$X_k = P_k X_{k-1} + e_k \text{............(7).}$$

The Kalman gain is defined by;

$$K_k = P_k^e \left[P_k^e + Y_k\right]^{-1} \text{..........(8)}$$

where P_k^e denotes the estimate error covariance matrix extrapolation given by;

$$P_k^e = F_{k-1} P_{k-1} F_{k-1}^T + Q_k \text{............(9)}$$

wheren P_{k-1} is updated estimate error covariance matrix evaluated from

$$P_{k-1} = \left[I - K_{k-1}\right] P_{k-1}^e \text{.............(10).}$$

Note that under the Gaussian assumption of the model and measurement errors, the estimation error distributions are also Gaussian with zero mean and variance defined by diagonal elements of the error covariance matrix, $\hat{v}_{e,k}^m, \hat{v}_{e,k}^v$, for cell rate mean and variance, respectively. These variances can be used to evaluate the bandwidth R_k reserved for the error in evaluation of $\hat{G}_k$. The details of the procedure are given in the next section. Concerning the initial values we assume that the system is empty at the time t = 0 so X_0 and $P_o = 0$.

3.2. Estimation of Aggregate Equivalent Bandwidth

As indicated, we assume that the function for equivalent bandwidth evaluation from the declared parameters is given $g_i^d = f_g(h_i), G_k^d = \sum_i g_i^d$. In the following, we focus on estimation of the aggregate equivalent bandwidth from the differential approach. In this case one needs to evaluate the sensitivity of G with respect to the cell rate mean and variance around the declaration point G^d (to simplify presentation the time index is omitted in this section). Note that the sensitivity function should not be complex since it is used in on-line evaluations. To fulfill this requirement, we derive this function from a simple expression suggested for approximate equivalent bandwidth evaluation.

$$g^d = \gamma . m^d + \theta . v^d \text{...................(11).}$$

Based on (11), for a particular state of accepted connections, we have;

$G^d = \gamma M^d + \theta V^d$ ....(12).

Since the values, G^d, M^d, and V^d and are known for each link state, one can evaluate parameters γ, θ from two recent link states. Alternatively, one can assume that the coefficient γ is independent from the current state and can be evaluated off-line using the link speed, QoS constraint, and average traffic mixture. Then the coefficient, θ for a given state, is given by;

$\theta = \frac{G^d - \gamma M^d}{V^d}$ ....(13).

Finally, the estimated equivalent bandwidth of the connection superposition process is evaluated from;

$\hat{G} = \gamma.\hat{M} + \theta.\hat{V}$ ....(14).

It should be emphasized that, contrary to its appearance, (14) serves only for evaluation of the correction to the declared equivalent bandwidth G^d implied by deviation of the estimated parameters, $\hat{M}, \hat{V}$, from the declarations, $M^d, V^d, (G, M^d, V^d$ and the function for equivalent bandwidth allocation are hidden in γ and θ).

Concerning the bandwidth reserved for the estimation error, under the Gaussian assumptions of the estimated mean and variance errors, the estimated bandwidth error, $\delta_G = \hat{G} - G$, is also Gaussian. Thus, under the assumption of mean and variance independence we have;

$R = V(\varepsilon_1)\sqrt{\gamma^2 \hat{v}_e^m + \theta^2 \hat{v}_e^m}$ ....(15)

where $\hat{v}_e^m, \hat{v}_e^v$ denote diagonal elements of the error covariance matrix, P, and $U(\varepsilon_1)$ denotes a coefficient derived from the normalized Gaussian distribution which ensures that .

$P\{G > \hat{G} + R\} \le \varepsilon_1$ ....(16).

4. ERROR ANALYSIS

4.1. Measurement Error

An optimal cell process measurement is complex and should involve analysis of the autocorrelation function. This issue is large enough to be treated in a separate publication. In this research, we adopt an approach which is simple to implement but still enables the analysis of important features of the proposed traffic admission control model.

In general, the measurement process can be divided into two stages. The first one provides information concerning the instant cell rate. In the paper we assume that this estimate is given. In the second stage, mean M_k, variance V_k of the instant rate and the measurement errors are estimated. In the following, we analyze a standard approach based on instant cell rate samples, $\{ d_j \}$, taken in regular intervals in the period $\left[t_{k,} t_{k-1}\right]$ and assumed to be independent. (A more general approach based on the autocorrelation function of the instant cell rate process would require estimation of the autocorrelation function, possibly using declarations and measurements).

The standard estimates of the measured cell rate mean and variance are given by;

$$\overline{M}_k = \frac{\sum_i d_i}{N_k} \qquad \text{................(17)}$$

$$\overline{V} = \frac{\sum_i (d_i - \overline{M}_k)^2}{N_k - 1} \qquad \text{................(18)}$$

where N_k denotes the number of the samples. Let us define the measurement errors for mean; and variance of the instant rate as;

$$\delta_k^{\overline{m}} = \overline{M}_k - M_k \qquad \text{................(19)}$$

$$\overline{\delta_k^{m}} = \overline{V}_k - V_k \qquad \text{..................(20)}$$

respectively. The theoretical values of these error variances are given by;

$$v^{m}_{e,k} = \frac{V_k}{N_K} \text{ ....(21)}$$

$$v^{v}_{e,k} = \frac{S_k - (V_k)^2}{N_k} \text{ ....(22)}$$

where Vk, Sk denote the variance and forth central moment of an instant rate, respectively. Since in our framework only measured values are available we apply the following estimates:

$$\overline{v}^{m}_{e,k} = \frac{\overline{V}_k}{N_k} \text{ ....(23)}$$

$$v^{v}_{e,k} = \frac{\overline{S_k} - (\overline{V_k})^2}{N_k} \text{ ....(24)}$$

where

$$\overline{S_k} = \frac{\sum_i (d_i - \overline{M_k})^4}{N_k - 1} \text{ ....(25).}$$

In the following, we analyze the accuracy of the approach by means of the connection admission process simulation at one switch output port (the details of the simulation model are given in Section V). The selected example is defined by the following parameters: link capacity L=25, connection requests intensity $\lambda = 200^{-1}$, mean connection holding time, $\mu^{-1} = 5.10^4$, source parameters peak rate PR=1, fixed burst length B = 50, exponentially distributed silence length with average S = 70, simulation run-time—4.10^6. The measurement quality is assessed for the case when the sources conform to the declared parameters. In this example, it is assumed that the sampling period, equal to the average on–off source period (120), provides sufficient independence of samples from a practical viewpoint.

The state duration is a critical factor for the quality of the measurement process, therefore the information concerning measured mean, variance and their errors are gathered as a function of the number of samples, N. The analyzed variables related to the mean cell rate measurement are defined as follows;

1) measured average error of the measured mean instant rate:

$$(26) \quad \overline{m_e^m}(N_k) = \frac{\sum \overline{\delta_k^m}(N_n)}{L_N}$$

where L_N expresses the size of the state population with N_K samples;

2) measured variance of the error of the measured mean instant rate:

$$\overline{V_e^m}(N_k) = \frac{\sum (\overline{\delta_k^m}(N_k))^2}{L_N} - (\overline{m_e^m})^2 \text{................(27).}$$

3) estimated variance of the error of the measured mean instant rate:

$$\hat{V}_e^m(N_k) = \frac{\sum \overline{v}^{m}_{e,k}(N_k)}{L_N} \text{.....................(28).}$$

Fig.3.4(a) depicts results obtained for the mean instant rate measurement. We observe that an underestimation appears for he short state periods. This spurious fault finds an explanation in the nature of the connection admission algorithm. When the state duration is short, the measured value is in general more likely to be significantly different from the declared one. In the figure 3.4(a), this fact is expressed by the error variance which indeed increases for small state durations. As a consequence, the over-estimation of the mean cell rate results in an overestimation of the equivalent bandwidth, thus a new connection is more likely to be rejected (no state change), while in the opposite case (underestimation of the mean cell rate), a new connection is more likely to be accepted. This effect results in the fact that there are more short duration states with a corresponding underestimate of the mean cell rate. The correctness of the above reasoning is illustrated in Figure 3.4 (b). In this case, the admission decisions were made using the declared aggregated bandwidth instead of the estimated one. The bias caused by the feedback is significant only when the number of rejected connections is large as is the case in the chosen example. It can also be removed by an appropriate modification of the state duration definition (introduction of state changes when the connections are rejected). Note that in both instances the variance of the error, estimated from (24), is close to the measured value.

1) measured average error of the measured mean instant rate:

$$(26)\quad \overline{m_e^m}(N_k) = \frac{\sum \overline{\delta_k^m}(N_n)}{L_N}$$

where L_N expresses the size of the state population with N_K samples;

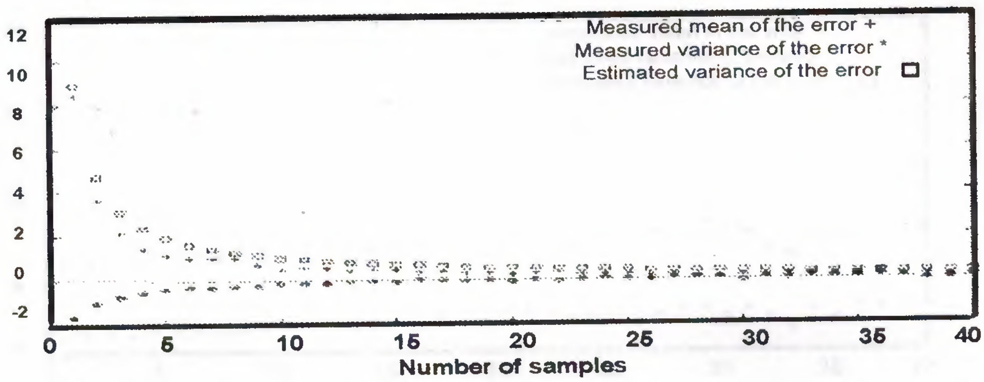
2) measured variance of the error of the measured mean instant rate:

$$\overline{V_e^m}(N_k) = \frac{\sum(\overline{\delta_k^m}(N_k))^2}{L_N} - (\overline{m_e^m})^2 \quad \text{................(27).}$$

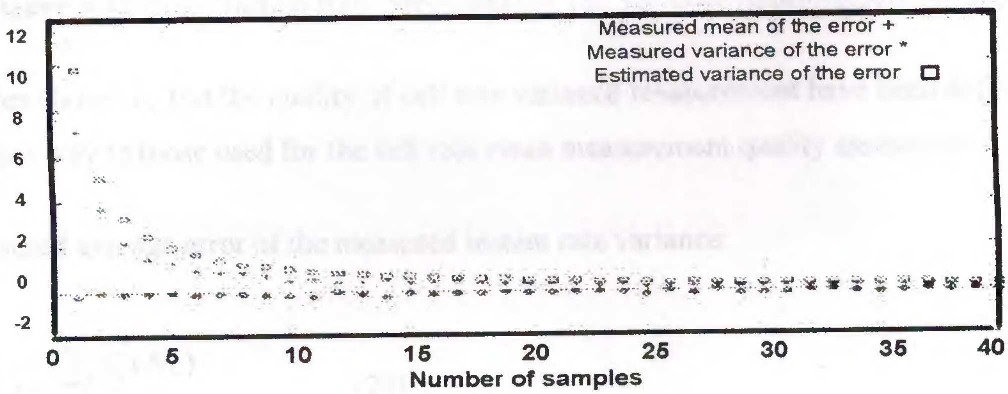
3) estimated variance of the error of the measured mean instant rate:

$$\hat{V}_e^m(N_k) = \frac{\sum \overline{v}^{\,m}_{e,k}(N_k)}{L_N} \quad \text{....................(28).}$$

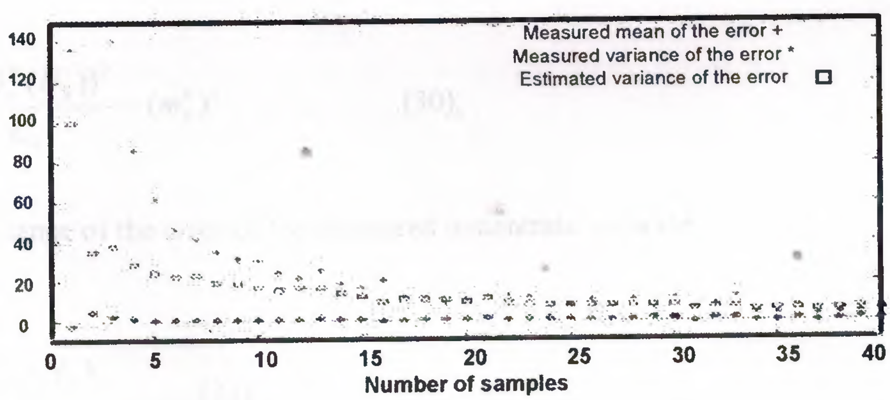
Fig.3.4(a) depicts results obtained for the mean instant rate measurement. We observe that an underestimation appears for he short state periods. This spurious fault finds an explanation in the nature of the connection admission algorithm. When the state duration is short, the measured value is in general more likely to be significantly different from the declared one. In the figure 3.4(a), this fact is expressed by the error variance which indeed increases for small state durations. As a consequence, the over-estimation of the mean cell rate results in an overestimation of the equivalent bandwidth, thus a new connection is more likely to be rejected (no state change), while in the opposite case (underestimation of the mean cell rate), a new connection is more likely to be accepted. This effect results in the fact that there are more short duration states with a corresponding underestimate of the mean cell rate. The correctness of the above reasoning is illustrated in Figure 3.4 (b). In this case, the admission decisions were made using the declared aggregated bandwidth instead of the estimated one. The bias caused by the feedback is significant only when the number of rejected connections is large as is the case in the chosen example. It can also be removed by an appropriate modification of the state duration definition (introduction of state changes when the connections are rejected). Note that in both instances the variance of the error, estimated from (24), is close to the measured value.


Measured mean of the error +
Measured variance of the error *
Estimated variance of the error
12
10
8
6
4
2
0
-2
0
5
10
15
20
25
30
35
40
Number of samples

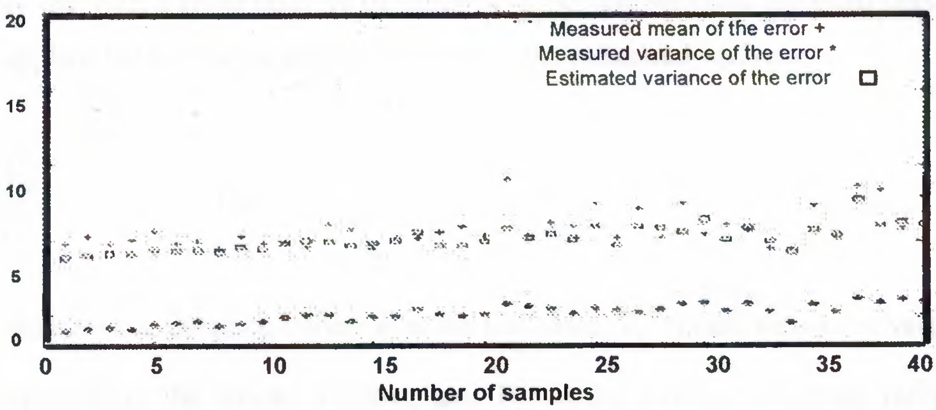
(a)


Measured mean of the error +
Measured variance of the error *
Estimated variance of the error
12
10
8
6
4
2
0
-2
0
5
10
15
20
25
30
35
40
Number of samples

(b)


Measured mean of the error +
Measured variance of the error *
Estimated variance of the error
140
120
100
80
60
40
20
0
0
5
10
15
20
25
30
35
40
Number of samples

(c)



(d)

Figure 3.4 (a), (b) Instant Rate Mean and (c), (d) Variance Measurement Analysis.

Variables chosen to test the quality of cell rate variance measurement have been defined in an analogous way to those used for the cell rate mean measurement quality assessment:

1) measured average error of the measured instant rate variance:

$$\overline{m}_e^m(N_k) = \frac{\sum \overline{\delta}_k^v(N_k)}{L_N} \quad \text{..................(29)}$$

2) measured variance of the error of the measured instant rate variance:

$$V_e^v(N_k) = \frac{\sum (\overline{\delta_v^k}(N_k))^2}{L_N} - (m_e^v)^2 \quad \text{.....................(30);}$$

3) estimated variance of the error of the measured instantrate variance:

$$\hat{V}_e^v(N_k) = \frac{\sum \overline{v}_{e,k}^v(N_k)}{L_N} \quad \text{...........(31)}$$

Results are presented in the Fig. 3.4(c). Discrepancy in the measured and estimated variance of the error for the small number of samples shows that, as could be expected, (18) cannot guarantee reliable results. To cope with the problem, we use an approximation based on a moving window with a fixed number of samples. Since the window span can contain several states, we applied the following approximation for the measured variance:

$$\overline{V}_k = \frac{V_k^d . \overline{V_k^w}}{V_k^a} \quad \text{.....................(32)}$$

where V_k^d denotes the declared variance in the k th state, $\overline{V}_k^w$ is the measured variance based on the samples from the whole window, and V_k^a is the average declared variance in the window. The efficiency of the applied mechanism may be verified through the comparison of Fig. 3.4(c) (standard procedure) and Fig. 3.4(d) (procedure enhanced by the moving window with 40 samples), both simulations run under the same conditions. Improvement of the accuracy, particularly for the short state durations is significant.

4.2. Mean and Variance Estimation Error

Let us define the estimation errors for mean and variance of instant rate as;

$$\delta_m^k = \hat{M}_k - M_k \quad \text{..................(33)}$$

$$\delta_k^v = \hat{V}_k - V_k \quad \text{......................(34)}$$

respectively. Under the Gaussian assumption of the model and measurement errors, the estimation error distributions are also Gaussian with zero mean and variance defined by the diagonal elements of the error covariance matrix, $\hat{v}_{e,k}^m, \hat{v}_{e,k}^v$. The error covariance matrix is a function of the measurement and model errors. The assessment of the measurement error was discussed in the previous section. In the following, we focus on the model error.

In general, the model error is a function of the declaration errors which are defined as a difference between the real and declared connection parameters.

$$c_j(t) = h_j^T(t) - h_j \quad \text{.................(35)}$$

where time t indicates that in general the error can be nonstationary. The transformation of the error into the discrete time domain can be done in several ways. One possibility is to assume

$c_{k+i} = c(t_{k+i}); i = 0,1,...,$ where t_k is the time of connection acceptance. Another possibility is to use an average error over the discretization interval.

$$c_{j,k+1} = \frac{\int_{k+i}^{k+i+1} c_j(t)}{t_{k+i+1} - t_{k+i}} \quad \text{.......................(36)}$$

In the context of our problem, the second option seems to be more appropriate since we are interested in QoS in the whole period between the state transitions. Then the model error is defined by;

$$e_k = c_{j,k} + \sum_i \left[c_{i,k} - c_{i,k-1}\right] \quad \text{....................(37)}$$

where index j corresponds to the new or departing connection and index i corresponds to an existing connection. Observe that, in general, the periods between the state transitions are significantly shorter than the connection duration. This feature indicates that although the connection parameters might be different from declarations, one can expect that there will be very large autocorrelation between the values in the subse-quent system states so the terms under summation in will be small. Moreover, under the assumption of statistical independence of the connection instant rate processes, these terms will be positive and negative. These premises lead to the conclusion that the second term in can be neglected so the model error can be approximated by the declaration error of the new or departing connection;

$$e_k = c_{j,k} \quad \text{..........................(38)}$$

This approximation is exact when each connection process is stationary. Based on (38), the covariance matrix of the model error, Q_k, is defined by the predicted variances of declaration errors, $\hat{v}^m_{e,k}, \hat{v}^v_{e,k}$ for cell rate mean and variance, respectively. These values can be evaluated from statistics. In our model it is assumed that the declaration errors have Gaussian distribution with zero mean. Nevertheless, the algorithm can cope very well with distributions quite different from the Gaussian, as is shown in Section V.

To verify the accuracy of the instant rate mean and variance estimation we use the example from the previous section, with introduced declaration error. In this case, the source declared parameters, m^d, v^d are still the same (m^d=0.5, v^d=0.25), but the actual ones are generated with the Gaussian error with zero mean and variances: $\hat{v}^{m}_{e,k} = 0.05$ (for practical reasons the distribution is limited to the range $m\varepsilon[0.2,0.8]$) $v^{v}_{e} = 0.01$ (in this case, the distribution is limited to the range $v\varepsilon[0.1,0.4]$). Note that due to the relation between the mean and variance, $v_k = m_k.(PR_k - m_k)$, the peak rates, PR_k, have different values for each connection. A set of the following variables will be used in the sequel:

1) The measured average error of the measured instant rate mean and variance, respectively,

$$\overline{m}^{m}_{e} = E\left[\overline{M}_k - M_k\right]; \overline{m}^{v}_{e} = E\left[\overline{V}_k - V_k\right] \ldots\ldots\ldots\ldots\ldots\ldots\ldots (39).$$

2) The measured variance of the error of the measured instant rate mean and variance, respectively,

$$\begin{aligned} \overline{V}^{m}_{e} &= E\left[(\overline{M}_k - M_k)^2\right] - (\overline{m}^{m}_{e})^2 \\ \overline{V}^{v}_{e} &= E\left[(\overline{V} - V_k)^2\right] - (\overline{m}^{v}_{e})^2 \end{aligned} \ldots\ldots\ldots\ldots\ldots\ldots\ldots (40).$$

3) The estimated variance of the error of the measured instant rate mean and variance, respectively,

$$\hat{V}^{m}_{e} = E\left[\overline{v}^{m}_{e,k}\right]; \hat{V}^{v}_{e} = E\left[\overline{v}^{v}_{e,k}\right] \ldots\ldots\ldots\ldots\ldots\ldots\ldots (41).$$

4) The measured average error of the estimated instant rate mean and variance, respectively,

$$\hat{m}^{m}_{e} = E\left[\hat{M}_k - M_k\right]; \hat{m}^{v}_{e} = E\left[\hat{V} - V_k\right] \ldots\ldots\ldots\ldots\ldots\ldots\ldots (42).$$

Model	$\overline{m}^{m}_{c}$	$\overline{M}$	$\overline{V}^{m}_{c}$	$\hat{V}^{m}_{c}$	$\hat{m}^{m}_{c}$
1	-0.090	18.5	4.99	4.78	-0.068
2	-0.091	18.5	4.69	4.79	-0.064

Table 3.1 Instant Rate Mean Estimation Results.

Model	$\overline{m}_c^v$	$\overline{V}$	$\overline{V}_c^v$	$\hat{V}_c^v$	$\hat{m}_c^v$
1	-0.053	9.27	4.39	4.09	-0.733
2	-0.134	9.11	4.23	4.20	-0.018

Table3.2 Instant Rate Mean Estimation Results.

Model	$\hat{m}_c^m$	$\hat{m}_c^v$	$\overline{E}[\hat{G}-G]$	$\overline{E}[\hat{G}]$
2	-0.020	-0.003	-0.243	22.1
3	-0.054	-0.019	-0.010	22.3

Table3.3 Instant Rate Mean Estimation Results.

The results corresponding to the instant rate mean and variance estimation are presented in Tables I and II, respectively. Model 1 corresponds to the algorithms and formulae presented in the previous section. The mean instant rate is estimated with a satisfactory accuracy, whereby the error does not exceed 0.5% of the estimated value. However, the results of the instant rate variance estimation are less satisfactory. We observe that the measurement part works appropriately, but the estimation produced a significant error. This can be explained by a strong correlation between the measured values of instant rate variance, $\overline{V}_k$, and variance of the instant rate variance error, $\overline{v}_{e,k}^v$, since the same set of samples is used to evaluate both values. As a result, whenever, for statistical reasons, the value of the measured variance is underestimated it has a larger weight in the Kalman filter due to the underestimated measurement error. This effect gives, on average, underestimation of the estimated variance of instant rate. This bias can be reduced by extending the window span used to evaluate the forth moment. The results for the extended window with Next.win =100 samples are presented in Tables I and II (Model 2). The quality of variance measurement is noticeably improved.

4.3. Equivalent Bandwidth Estimation Error

The evaluation of the equivalent bandwidth estimation ac-curacy is based on the example from the previous section. The results concerning the error in equivalent bandwidth estimation are presented in the Table III (Model 2), where $\overline{E}[.]$ denotes the measured average value. Note that although the mean and variance of the instant rate are estimated correctly, the equivalent bandwidth is slightly underestimated. This is due to the fact that the exact relation between the equivalent bandwidth and the instant rate mean and variance is slightly nonlinear and nonsymmetrical with respect to the central point of source parameter distribution while the applied approximation (14) is linear. To illustrate this effect, we evaluated "optimal" coefficients, θ, γ by integrating the exact equivalent bandwidth function. The results are given in the Table III (Model 3). In this case, the error is negligible.

Imp. €1	$\hat{P}$\{Gk>$\hat{G}$k+Rk\}
10^{-1}	$3.2.\ 10^{-1}$
10^{-2}	$3.5.\ 10^{-2}$
10^{-3}	$4.7.\ 10^{-3}$
10^{-4}	$1.1.\ 10^{-4}$
10^{-5}	$1.3.\ 10^{-5}$

Table 3.4 Quality of The Equivalent Bandwidth Estimation.

To verify the accuracy of the bandwidth reserved for the estimation error, R_k, the probability $P\{G_k > \hat{G}_k + R_k\}$ was estimated in a simulation experiment. In order to reduce the declaration error distribution deformations caused by the phys-ical constraints, the declaration error for variance was reduced: $v^{v}_{e,k}$=0.001. The estimated probabilities $P\{G_k > \hat{G}_k + R_k\}$ are in Table IV for different values of the constraint ε_1. Note that in all cases the estimated probabilities are close to the constraint defined by (2).

5. CONNECTION ADMISSION ANALYSIS

5.1. Connection Admission Procedure

As stated in Section II, in the proposed framework a new connection is accepted if

(43) $g_k^r \leq L - G_{k-1}^t$.

Note that in this condition the estimate from state $k-1$ is treated as a prediction for state k. The implication is that the aggregate connection process can be treated as stationary over the period of the two states. While estimation of $G_{k-1}^r = \hat{G}_{k-1} + R_{k-1}$ was already described, interpretation and eval-uation of the bandwidth reserved for a new connection, g_k^r, requires additional clarification. In particular, it is important to define more precisely what is the design objective of the connection admission procedure and how the quality of this procedure should be judged. The answers to these questions are not straightforward. Note that the main criterion of the connection admission is to ensure that the QoS constraint is met (on the cell layer). In our model, this requirement is fulfilled when the real bandwidth required by the admitted connections does not exceed the link capacity, $G_k \leq$ L . Obviously strict execution of this condition might cause the bandwidth utilization to be compromised. Thus, to improve bandwidth utilization, we allow that $G_k >$ Lwith a certain small probability;

$P\{G_k > L\} \leq \varepsilon_2$.............................(44)

There is one drawback with this formulation. Namely, this probability depends strongly on the connection arrival process and connection bandwidth requirements. In particular, the smaller traffic level the smaller P{ G_k > L}. This feature indicates that the condition (44) is not convenient for the connection admission algorithm design.

To deal with this issue, we propose another definition of the connection admission procedure quality which is independent from the connection arrival process and connection band-width requirements. It is based on the following conditional probability;

$P\{G_k > L \big/ g_k^r = L - G_{k-1}^r\} \leq \varepsilon_2$.................... (45)

In this case, the quality is defined for the critical case where the residual capacity is equal to the one required by a new connection. Observe that condition (45) is also fulfilled when;

$P\{G_k > g_k^r + G_{k-1}^r\} \le \varepsilon_2$.................(46).

The latter condition constitutes the basis for design and evaluation of the proposed connection admission procedure.

In analogy with G'_{k-1}, the bandwidth reserved for a new connection, g'_k, can be decomposed into two parts, the equivalent bandwidth, g_k^d, evaluated from the declared parameters and the bandwidth, γ_k, reserved for the declaration error. Note that the sum $\gamma_k IR_{k-1}$ could be evaluated from the superposition of distributions of gk and G_{k-1}. Nevertheless, from the connection admission viewpoint it is more convenient to separate evaluation of γ_k from evaluation of R_{k-1}. The main reason behind this approach is that in this case the processing of the CAC control cell in a transit ATM node is limited to a simple comparison of two numbers.

Based on the Gaussian assumption, the bandwidth reserved for the declaration error can be evaluated from;

$r_k = U(\varepsilon)\sqrt{\gamma^2 v_e^m + \theta^2 v_e^v}$ ....(47)

where the value of the constraint ε_3 can be chosen between $\varepsilon_3 = \varepsilon_2 - \varepsilon_1$ (conservative) and $\varepsilon_3 = \varepsilon_2 / \varepsilon_1$ (optimistic). An-other possibility is to apply an approach where the parameter γ_k is defined by the connection peak rate, PR_k;

$r_k = PR_k - g_k^d$ ....(48)

This approach is simpler and safer since the peak rate defines the upper boundary for the error. In the following, we use the second approach.

5.2. Numerical Examples

We start by describing a simulation model used for as-sessment of the proposed algorithms. To avoid excessive complexity we simplified the simulation model as much as possible to concentrate on the main issues. In particular, only the instant rate layer is modeled and the link buffer has zero length. It should be stressed that the buffer-less case was chosen only to simplify evaluation of the exact equivalent bandwidth allocation. This choice does not restrict the analyzed CAC model applications nor limit generality of the results. This follows from the fact that the CAC algorithm operates on the notion of the equivalent bandwidth in a way which separates the issue of CAC adaptiveness from buffer dimensioning and performance measure on the cell layer. Moreover, the buffer-less case exemplifies the burst scale layer model whose performance is critical for a robust CAC algorithms for real-time services.

The requests for connections of particular class are gen-erated with intensity λ (Poissonian distribution) and mean holding time μ^{-1} (exponential distribution). The connections are of on–off type and are described by the peak rate, PR, average burst length, B, (with programmable distribution) and the average silence length, S (exponential distribution). The QoS constraint (cell loss probability—B^c) is set to a relatively high value B^c=10^{-2} in order to achieve a reliable estimate of the cell loss probability distribution under nonstationary traffic conditions. The estimation error constraint is set to the same value, $\varepsilon_1 = 10^{-2}$.

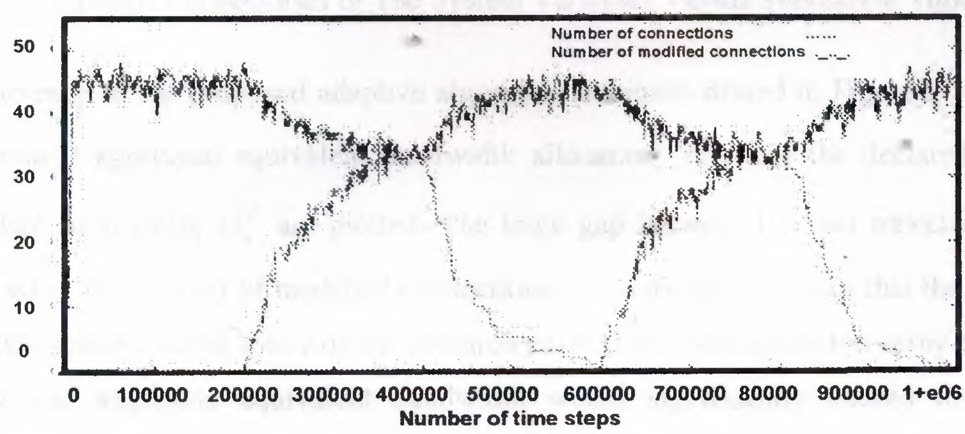
The declaration error can be generated in many ways. The results presented in the previous section are based on the Gaussian model for this error distribution, which conforms with the Kalman filter assumptions. In this section, we introduce another error model which is significantly different from the Kalman filter assumption. Namely, the error generator has two cyclic states ("on" and "off") with the same period T. When the generator is in the state "on" all connections accepted in this state are generated with the burst length larger than the declared one $(B' = B + \Delta B)$ In the "off" state, all new connections are generated with the declared burst length (for entire duration of the connection). Note that in this case the error has no zero mean. The reason for this model is to evaluate the adaptation scheme under more stressing conditions.

The binomial distribution is used to evaluate both the equivalent bandwidth from the source declarations, $g_k^d = f_g(h_k)$, and the exact aggregate equivalent bandwidth based on the real connection parameters parameters, G_k. The performance of the CAC mechanism under the deter-ministic and nonstationary error is illustrated on two exam-ples. The first one (Ex.2) is defined by: link and connection parameter $L = 25, \lambda = 200^{-1}, \mu^{-1} = 5.10^4$ source parameters

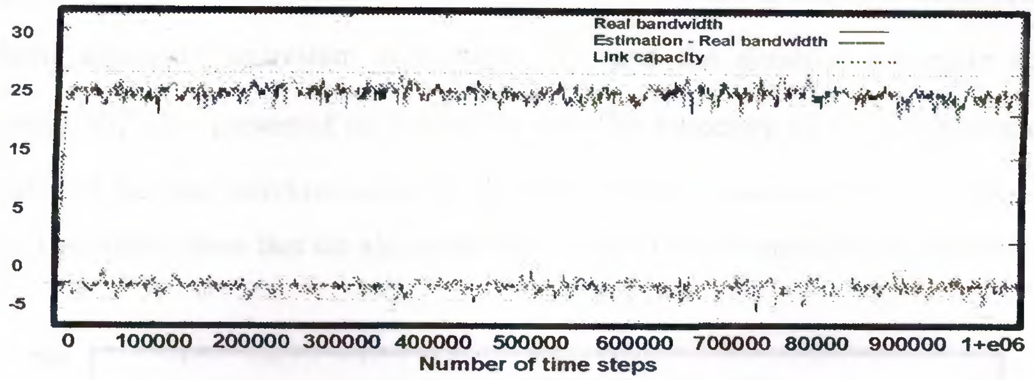
$PR = 1, B = 50, S = 70, v_e^m = 0.05, v_e^v = 0.0002$; error generator parameters $T = 2.10^5, B' = 100$

A sample of the equivalent bandwidth allocation dynamics, during a part of the simulation run, is presented in Fig. 3.5. In Fig. 3.5(a), the total number of connections and the number of connections with modified burst length are given. During the "on" period of the error generator the parameters of almost all connections are modified while at the end of the "off" period almost all connections have the declared parameters.

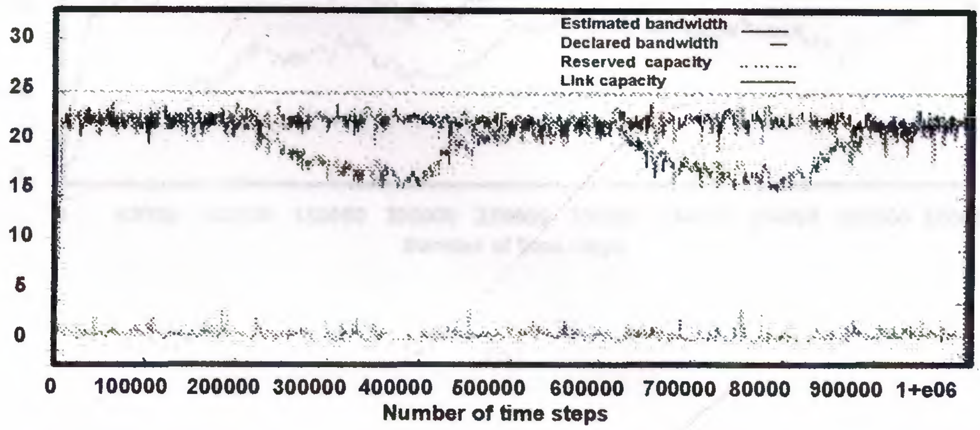
The trajectory of the real aggregate equivalent bandwidth, Gk, and the difference between the estimated and real ag-gregate equivalent bandwidth, $\hat{G}_k - G_k$, are depicted in Fig. 3.5(b). The estimated aggregate equivalent bandwidth accu-rately tracks the actual value. It can be noticed that the largest underestimation of the equivalent bandwidth occurs during the "on" period of the error generator. Nevertheless the bandwidth reserved for estimation error ensures that the real aggregate equivalent bandwidth does not exceed the link capacity. This result shows that the proposed scheme is very robust since during the error generation period the real declaration error is not only on average two times larger than that declared but also is always positive. This robustness results from two factors. First, at the moment of a new connection admission there is provision for the equivalent bandwidth equal to its peak rate so no matter how malicious the source, the QoS is kept under the constraint during the new state. Although in subsequent states this provision disappears (the connection is included in the aggregate equivalent bandwidth), the system has time to correct the bandwidth allocation by means of the measurement process incorporated in the estimation of the aggregate equivalent bandwidth.



(a)



(b)

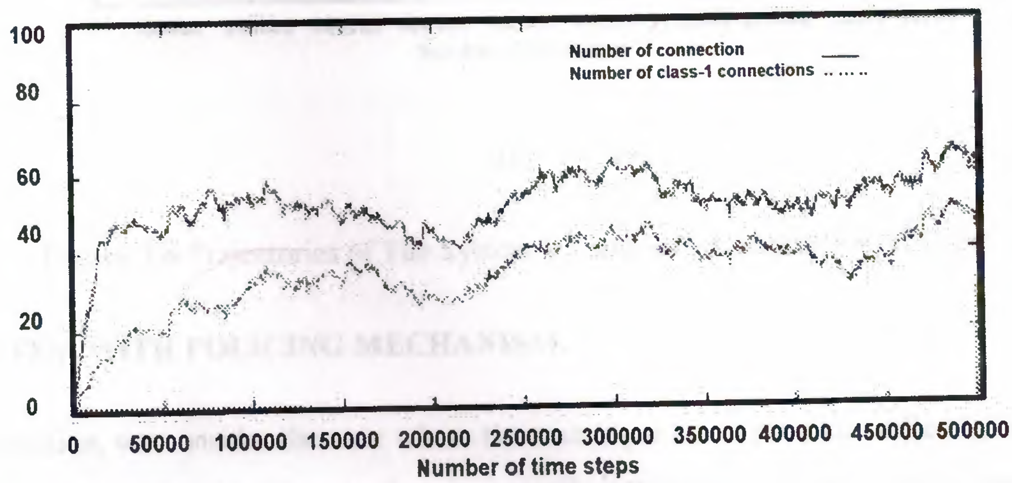


(c)

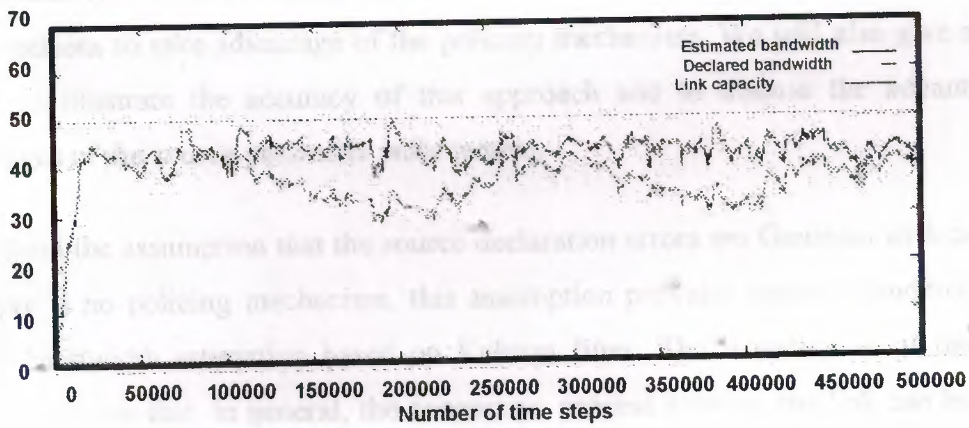
Figure 3.5 Trajectories of The System Variables Versus Simulation Time.

The efficiency of the proposed adaptive algorithm is demon-strated in Figure 3.5 (c), where the estimated aggregate equivalent bandwidth allocation, $\hat{G}_k$, and the declared aggregate equiv-alent bandwidth, G_k^d are plotted. The large gap between the two trajectories, in the periods when the number of modified connections is significant, indicates that the connection admis-sion scheme based solely on the declared parameter would allow too many connections and the real aggregate equivalent bandwidth would significantly exceed the allocated capacity. In addition, the bandwidth reserved for the estimation error R_k is depicted in Figure 3.5(c).

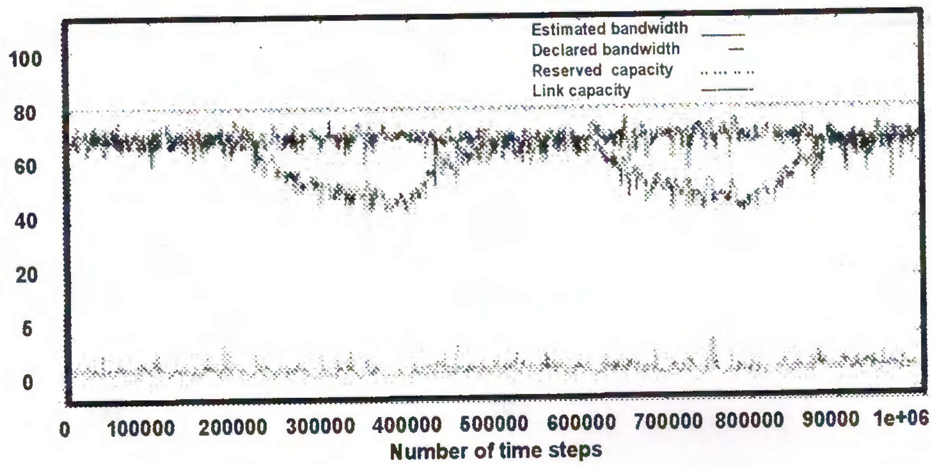
A sample of the system dynamics, during a part of the simulation run, is presented in Fig. 3.6. In Figure 3.6 (a), the tra-jectories of connection numbers are given. The trajectories of the estimated aggregate equivalent bandwidth, $\hat{G}_k$, and the declared aggregate equivalent bandwidth, G_k^d, are presented in Figure 3.6 (b). The trajectory of the cell loss probability (estimated in the time windows used for the cell variance measurements) is given in Figure 3.6 (c). The results show that the algorithm is also robust in the multiclass environment.



(a)



(b)



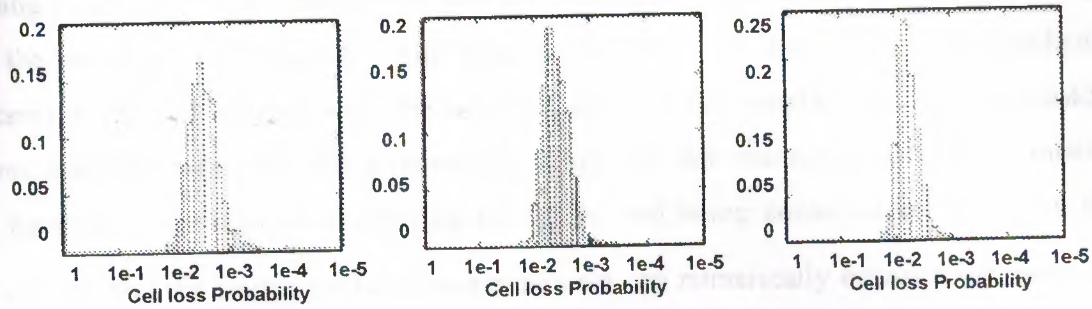
(c)

Figure 3.6 Trajectories of The System Variables Versus Simulation Time.

6. SYSTEM WITH POLICING MECHANISM

In this section, we consider the case where the source parameters are controlled by means of the policing mecha-nism. Observe that, in general, introduction of the policing mechanism reduces the magnitude of the source declaration error seen at the switch output ports. Thus, there is a potential to increase further the link bandwidth utilization without compromising the QoS performance. In the following we will extend the adaptive algorithm presented in the previous sections to take advantage of the policing mechanism. We will also give numerical examples to illustrate the accuracy of this approach and to discuss the advantages and disadvantages of the source parameter enforcement.

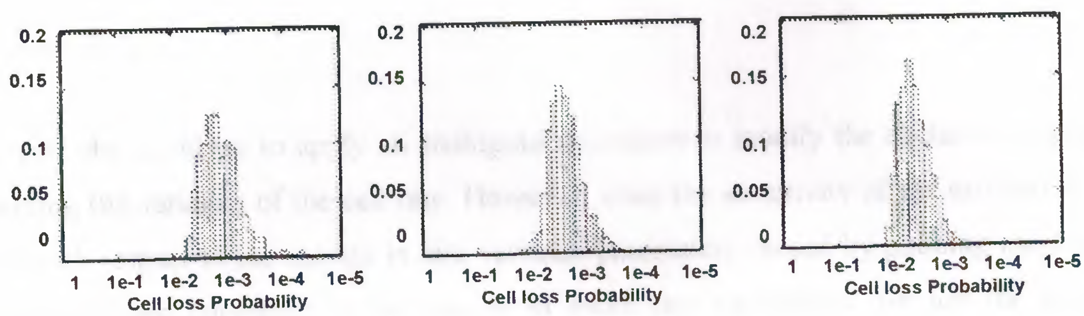
We start from the assumption that the source declaration errors are Gaussian with zero mean. When there is no policing mechanism, this assumption provides optimal conditions for the aggregate bandwidth estimation based on Kalman filter. The introduction of the policing mechanism implies that, in general, the connection process entering the link can be different from the source process. In particular, it means that the declaration error distributions are no longer Gaussian.



(a) (b) (c)

Figure 3.7 Cell Loss Probability Distributions, Ex.1: (a) LR = 1:0,

(b) LR=0:62, and (c) LR=0:50.



(a) (b) (c)

Figure 3.8 Cell Loss Probability Distributions, Ex.2: (a) LR = 1:0 ,

(b) LR = 0:62 and (c) LR =0:50.

In addition, they are not symmetrical (nonzero mean). To cope with this issue, in the following, we describe an approximation which enables the basic connection admission algorithm to be preserved with only a few minor modifications. This is achieved by a modification of the declared parameters con-cerning the mean cell rate with the result that the declaration error has zero mean. Then these parameters are used in the original algorithm which assumes Gaussian error distributions.

The influence of the policing mechanism on the mean rate declaration error distribution can be modeled as follows. First note that the upper Gaussian distribution tail of the original distribution is truncated at the threshold value which is defined by the leak rate, LR, of the policing mechanism. Then we assume that all connections with source mean cell rate higher than the threshold have mean cell rate equal to the threshold after the policing mechanism. Concerning the connections with the source mean cell rate smaller than the threshold we assume that their mean cell rate is preserved. While the two assumptions are approximations, they have the advantage of simplifying the model and being conservative. Based on these assumptions and the source declared parameters we can numerically evaluate the mean $m_{e,k}^{m'}$ and variance $v_{e,k}^{m'}$ of the modified error distribution. Then the modified declared connection mean cell rate, used by the original connection admission procedure, is defined as;

$$m_k^{\iota} = m_k + m_{e,k}^{m^{\iota}} \qquad (49)$$

and the mean rate declaration error is assumed to have Gaussian distribution with zero mean and variance $v_{e,k}^{m'}$.

Note, that one could try to apply an analogous procedure to modify the declared parameters concerning the variance of the cell rate. However, since the sensitivity of the estimation procedure with respect to the change in rate variance parameters caused by policing mechanism is less significant compared to the change in mean rate parameters, we use the source's declared rate variance and its error variance in the studied CAC procedure.

The modifications of the declared mean rate and the declaration error variance in uence the evaluation of the aggregate equivalent bandwidth and the bandwidth reserved for the estimation error. The policing mechanism also in uences the bandwidth reserved for the declaration error, γ_k In the original approach, this bandwidth was evaluated as the difference between the connection peak rate and equivalent bandwidth (48). Since the policing mechanism limits the worst-case equivalent bandwidth used by the connection, the bandwidth reserved for the declaration error can be reduced to;

$$r_k^{\iota} = g_k^{\max} - g_k^{d} \qquad (50)$$

where $g_k^{\max}$ denotes the equivalent bandwidth required by the worst-case source.

CONCLUSION

Most of the existing traffic control and bandwidth manage-ment algorithms for ATM based networks focus on particular types of services. In the paper, we have proposed a unified framework which bridges the gap between the algorithms for real-time services and controllable data services. The framework is based on the rate-based scheme for controllable data services, recommended by the ATM-Forum, and the adaptive connection admission algorithm proposed in the paper. Both algorithms react to measurements of the cell rate process at the switch output ports and use the same data base structure and signaling protocol. Besides reduction of the algorithms complexity and cost this feature can also increase bandwidth utilization by close coordination of both algorithms.

The central concept of the proposed connection admission algorithm is estimation of the aggregate equivalent bandwidth required by connections carried in each output port of the ATM switches. The estimation process takes into account both the traffic source declarations and the connection superposition process measurements in the switch output ports. This is done in an optimization framework using a linear Kalman filter. The Kalman filter approach fits naturally the estimation problem under consideration. It takes into account the connection level dynamics of the system in an optimal way and it gives information for evaluation of bandwidth reserved for possible estimation error. This bandwidth ensures a statistical guarantee for quality of service.

The numerical analysis of the estimation process showed that the bandwidth reserved for the estimation error is very accurate when the declaration error has Gaussian distribution. The study of the connection admission algorithm under nonsta-tionary conditions and large, non-Gaussian, declaration errors demonstrated that the approach is very robust and copes very well with undeclared changes in traffic parameters.

By applying an efficient approximation, the estimation model was extended for cases with source policing algorithms. The numerical study illustrated the tradeoff between relaxed and strict source policing. The results indicate that application of relaxed policing can be advantageous from both user and network viewpoints. The inclusion of a policing mechanism also shows that the proposed approach can coexist with the ATM Forum and ITU recommendations. There are several further model extensions which could be investigated. For example, the measurement process consti-tutes an important area for study where declared and measured autocorrelation functions can play an important role. More complex filters can be tried to check whether it is possible to estimate some characteristics corresponding more directly to the QoS metrics. Also the problem of correlated sources can be investigated since the model can handle such cases. Finally an integrated model for bandwidth management for CTP and NCTP services based on a partly common estimation algorithm can be studied.

SUMMARY

Asynchronous Transfer Mode(ATM) is an extremely high speed, low delay, multiplexing and switching technology that can support any type of user traffic including voice, data, and video applications. ATM is ideally suited to applications that cannot tolerate time delay, as well as for transforming frame delay and IP traffic that are chracterized as busty.

As we enter the 21st century a competitive environment meets us where high-speed virtual networking is the upcoming field of interest. The question, why is that we choose ATM networks . To answer that we can only say that, ATM networks are not only the highest speed networking in the 1990s. The fact that before ATM, separate networks were required to carry voice, data and video information is alone enough to support its importance. The unique profiles of these traffic types make significantly different demands on network speeds and resources.

Data traffic can tolerate delay, but voice and video cannot. With ATM, however, all of these traffic types can be transmitted or transported across one network (from megabit to gigabit speeds), because ATM can adapt the transmission of cells to the information generated.

ATM works by breaking information into fixed lenght 53-byte data cells. The cells are transported over traditional wire or fiber optic networks at extremely high speeds.

Because information can be moved in small, standard-sized cells, switching can take place in the hardware of a network, which is much faster thansoftware switching. As a result, cells are routed through the network in a predictable manner with very low delay and with determined time intervals between cells. For network users, this predictability means extremely fast, real-time transmission of information on the same network infrastructure that supports non-real-traffic support.

ATM is a connection-oriented protocol, meaning that ATM must establish a logical connection to a defined endpoint before this connection can transport data. Calls on each port are assigned a path and a channel identifier that indicates the path or channel over which the cell is to be routed. The connections are called virtual paths or virtual channels.

ese connections can be permanently established or they can be set up and released amicallly depending upon the requirements of the user. A path can be an end-to-end nection in itself, or it can be a logical association or bundle of virtual channels.

M creates a common way of transmitting any type of digital information from any other elligent device over a system of networks. Many types of networks can be consolidated ng one technology. This gives ATM a substantial advantage over other transport hnologies.

M was designed for user and network provides who require guaranteed real-time nsmission of voice, data, and images while also requiring efficient, high performance nsport of busty packet data. Hospitals are using ATM to share real-time video and images long distance consultation during diagnosis and operations. Schools are using ATM to ng students and instructors together, regardless of their locations.

rporations whose employees are in different location benefits by using ATM to effortlessly are the largest data files. The internet runs on high-speed ATM backbones.

TM is also proving to be an important technology for managing the demands placed on erburdened networks by the surge of internet traffic. By adding a rich layer of traffic gineering to the internet protocol (IP), ATM supports the transportation of traffic according priority level and class of service, neither of which are supported by IP alone. The result is more effective means of ensuring the transmission of mission-critical traffic, a growing oncern of services providers and businesses a like.

ABBREVIATION

AAL–1	ATM adaption Layer 1
AAL–2	ATM adaption Layer 2
AAL–3	ATM adaption Layer 3
AAL–4	ATM adaption Layer 4
AAL–5	ATM adaption Layer 5
ABR	available bit rate
ATM	asynchronous transfer mode
ATM UNI	ATM user network interface
BT	burst tolerance
CAC	connection admission control
CBR	constant bit rate
CCITT	Comite Consultif Internationale de Telegraphique et Telephonique
CDV	cell delay variation
CLR	cell loss ratio
CTD	cell transfer delay
IEEE	Institute of Electrical and Electronic Engineers
ITU–T	International Telecommunications Union–Telecommunications Standards Sector
LAN	local-area network
LANE	ATM LAN emulation
LES	LAN emulation server
LUNI	LAN emulation user-network interface
MPOA	multiple protocol over ATM
P-NNI	public network-to-network interface
PCR	peak cell rate
PVC	permanent virtual circuit
RM	resource management

D	speech activity detection
R	sustained cell rate
H/SONET	synchronous digital hierarchy/synchronous optical network
C	switched virtual circuit
P/IP	transmission control protocol/Internet protocol
BR	unspecified bit rate
BR–NRT	variable bit rate–nonreal time
CC	virtual-channel connections
PC	virtual-path connections

REFERENCES

[1] J. S. Turner, "The challenge of multi-point communication," in 5th ITC Seminar, Lake Como, Italy, May 1987.

[2] G. Woodruff, R. Kositpaiboon, G. Fitzpatrick, and P. Richards, "Control of ATM statistical multiplexing performance," in Proc. ITC Specialist Seminar, Adelaide, Australia, 1989.

[3] Z. Dziong, J. Choquette, K.-Q. Liao, and L. Mason, "Admission control and routing in ATM networks," in Proc. ITC Specialist Seminar, Adelaide, Sept. 1989.

[4] R. Guerin, H. Ahmadi, and M. Naghshineh, "Equivalent capacity and its application to bandwidth allocation in high-speed networks," IEEE J. Select. Areas Commun., vol. 9, 1991.

[5] F. Kelly, "Effective bandwidths at multi-class queues," Queueing Syst., vol. 9, pp. 515,1991.

[6] R. Jain, "Congestion control and traffic management in ATM networks: recent advances and a survey," Computer Networks ISDN Syst., Jan. 1995.

[7] D. Mitra, "Asymptotically optimal design of congestion control for high speed data networks," IEEE Trans. Commun., vol. 40, pp. 301–311, Feb. 1992.

[8] K.-Q. Liao and L. G. Mason, "A congestion control framework for broadband ISDN using selective window Control," in Proc. Broadband Communications'94, Paris, France, Mar.1994.

[9] T. E. Tedijanto and L. Gun, "Effectiveness of dynamic bandwidth management mechanisms in ATM networks," Proc. INFOCOM'93, San Francisco, CA, 1993.

[10] H Saito., "Dynamic call admission control in ATM networks," IEEE J. Select. Areas Commun., vol. 9, Dec. 1991.

[11] H. Kroner and T. Renger, and R. Knobling, "Performance modeling of an adaptive CAC strategy for ATM networks," in Proc. ITC-14, Paris, France, June 1994.

[12] R. Griffiths and P. Key, "Adaptive call admission control in ATM networks," in Proc. ITC-14, Paris, France, June 1994.

R. J. Gibbens, F. P. Kelly, and P. B. Key, "A decision-theoretic approach to call
ission control in ATM networks," IEEE J. Select. Areas Commun., vol. 13, pp. 1101–
4, Aug. 1995.

S.-Q. Li, S. Chong, and C. L. Hwang, "Link capacity allocation and network control by
red input rate in high-speed networks," IEEE/ACM Trans. Networking, vol. 3, Feb. 1995.

] C. D. Pack and B. A. Whitaker, "Kalman filter models for network forecasting," Bell
t. Tech. J., vol. 61, no. 1, pp. 1–14, 1982.

] P. Chemouil and J. Filipiak, "Kalman filtering of traffic uctuations for real-time network
nagement," Annales des Telecommun., vol. 44, nos. 11/12, pp. 633–640, 1989.

7] Z. Dziong and L. Mason, "Call admission and routing in multi-service loss networks,"
EE Trans. Commun., vol. 42, Apr. 1994.

8] P. E. Boyer and D. P. Tranchier, "A reservation principle with applica-tions to the ATM
ffic control," Computer Networks ISDN Syst., vol. 24, pp. 321–334, 1992.

9] J.W. Roberts, "Performance evaluation and design of multi-service networks," COST
24 final report Oct. 1992.

0] G. Gallassi, G. Rigolio, and L. Fratta, "Bandwidth assignment and bandwidth
nforcement policies," in Proc. GLOBECOM'89, Dallas, TX, 1989.

21] Z. Dziong, K-Q. Liao, and L. Mason, "Effective bandwidth alloca-tion and buffer
imensioning in atm based networks with priorities," Computer Networks ISDN-Systems,
ol. 25, pp. 1065–1078, May 1993.

22] M. Veeraraghavan, T. F. La Porta, and W. S. Lai, "An alternative approach to
all/connection control in broadband switching systems," in IEEE BSS'95, Poznan, Poland,
Apr. 1995.

23] A. Gelb, Applied Optimal Estimation. Cambridge, MA: MIT Press, 1974.

[24] A. Gach, C. Mialaret, and P.E. Allard, "An Experimental Evaluation of Call Acceptance
Management Algorithms in ATM Based Networks," in Proc. CCECE-92, Toronto, Ont.,
Canada, Sept. 1992.

5] M. Ritter and P. Tran-Gia, Eds., COST 242 Mid Term Seminar, l'Aquila, 27 Sept.1994.

6] H. Yamada and S. Sumita, "A Traffic Measurement Method and its Application for Cell oss Probability Estimation in ATM Networks," IEEE J. Select. Areas Commun., vol. 9, pp. 5–324, Dec. 1991.

7] F. Kelly, "Tariffs and effective bandwidths in multiservice networks," in Proc. ITC-14, ntibes Juan-les-Pins, France, June 1994.

8] K-Q Liao., Z. Dziong, L. Mason, and N. Tetreault, "Effectiveness of leaky bucket olicing mechanism, " in Proc. ICC'92, Chicago, IL, 1992.

29] Z. Dziong, M. Juda, A. Jajszczyk, and L. G. Mason, "In uence of UPC on connection dmission based on measurements in ATM switch output ports," in 1st Workshop ATM raffic Management WATM'95, Paris, France, 1995, pp. 77–84.