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ENHANCED CONVOLUTION APPROACH FOR CAC IN ATM NETWORKS, AN ANALYTICAL STUDY AND IMPLEMENTATION

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"ENHANCED CONVOLUTION APPROACH FOR CAC IN ATM NETWORKS, AN ANALYTICAL STUDY AND IMPLEMENTATION".

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GLOSSARY

ABR	Available Bit Rate
ABT	ATM Block Transfer
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Service Digital Network
BT	Burst Tolerance
CAC	Connection Admission Control
CBR	Continuous Bit Rate
CD	Cell Delay
CDV	Cell Delay Variation
CLP	Cell Loss Priority
CLR	Cell Loss Ratio
CTD	Cell Transfer Delay
DTR	Deterministic Bit Rate
ECA	Enhanced Convolution Approach
FCAC	Fuzzy CAC
FCFS	First Come First Served
FF	Fluid-Flow model
GCRA	Generic Cell Rate Algorithm
GMDP	General Modulated Deterministic Process
GVP	Group Virtual Path
ISDN	Integrated Service Digital Network
ITU	International Telecommunication Union
ITU-T	ITU Telecommunication Standardization Sector
LAN	Local Area Networks
MBS	Maximum Burst Size
MCR	Minimum Cell Rate
MDF	Multinomial Distribution Function
MMDP	Markov Modulated Deterministic Process
MMPP	Markov Modulated Poisson Process
NP	Network Provisioning
NRM	Network Resource Management
PC	Probability of Congestion
PCR	Peak Cell Rate
PTM	Packet Transfer Mode
QOS	Quality of Service
RACE	Research and development in Advanced Communication technologies in Europe
RM	Resource Management
SBR	Statistical Bit Rate
SCR	Sustainable Cell Rate
STM	Synchronous Transfer Mode
STR	Statistical Bit Rate

TAT	Theoretical Arrival Time
UBR	Unspecified Bit Rate
UPC	Usage Parameter Control
VBR	Variable Bit Rate
VBRV	Variable Bit Rate Video
VC	Virtual Channel
VP	Virtual Path
VPN	Virtual Path Network

NOTATION

e	Maximum admitted Probability of Congestion.
l	arrival rate in cells
r	utilisation (number of carried cells per time slot)
l **i**	Call rate
a(k)	mean of the arrival distribution
C	Capacity of the link
$\mathbf{h_i}$	Average Holding time
i	i-state
j	j-type traffic
k	Number of active sources
k	number of arrival cells
L	Number of traffic classes
$\mathbf{l_i}$	Mean Burst Length
M	Number of elements
N	Number of sources
P	Probablility
$\mathbf{P_k}$	Probability that k sources are active
$\mathbf{R_i}$	Constant bit rate
$\mathbf{Rm_i}$	Peak Cell Rate
$\mathbf{Rp_i}$	Mean Cell Rate
S	Number of possible states
s(k)	state probability
SMX	Sub-Matrix (homogeneous traffic)
Source-SV	Source Status Vector
SSM	System Status Matrix
System-SV	System Status Vector
V	Velocity (rate)
W	Offered traffic
W'	Carried traffic

1. Introduction

1.1 ATM TRANSPORT NETWORK OVERVIEW

The Asynchronous Transfer Mode (ATM) transport network is based on fast packet switching using small fixed-size packets called cells. ATM permits flexible bandwidth allocation, so an important objective is to obtain the maximum statistical gain on a shared resource: the physical link.

The characteristics of service independence and flexibility associated with ATM networks make the control problems of such networks critical. One of the main challenges in an ATM network is to design traffic control mechanisms that simultaneously enable economically efficient use of network resources and provide the desired quality of service to higher layer applications. Window flow control mechanisms of traditional packet switched networks (at the speeds envisaged for the B-ISDN network) are not well suited to real time services.

Two major control problems can be identified in ATM networks: Call Admission and Source Policing. This work focuses on Call Admission Control and related aspects: Bandwidth Allocation, Traffic characterisation and Quality of Service parameters.

1.2 BASIC ATM ISSUES

1.2.1 Current Situation

Today's telecommunications networks are characterised by specialisation. Most networks are dedicated to specific purposes such as telephone systems, TV distribution, circuit-switched or packed data transfer. Some applications, such as facsimile, make use of the widespread telephone network. Using pre-existing networks for new applications may lead to characteristic shortcomings. However, as such networks are not usually tailored to the needs of services that were unknown when the networks were implemented. So data transfer over the telephone network is confined by a lack of bandwidth, flexibility and quality of analogue voice transmission equipment. As telephone networks were engineered for a constant bandwidth service, using them for variable bit rate data traffic requires costly adaptation.

It can be concluded that the current networks are very specialised and suffer a large number of disadvantages, the most important are: Service Dependence (each network is only capable of transporting one specific service) ; Inflexibility (today's network technology is influenced by the current service requirements); and Inefficiency (resources which are available in one network cannot be made available to other networks) [PRY91].

Since, in general, the public telephone network was unable effectively to support non-voice services to the extent required by the customer, other dedicated networks arose, such as public data networks or private data networks connecting, say, a large company's plants or several research institutes. An example of a large data network is Internet. Private networks often deploy equipment, interfaces and protocols which are unable to offer access to other networks and users. If in such an environment gateways are required to the outside world, their implementation may be tedious and costly [HAN94].

1.2.2 B-ISDN

In 1984 the CCITT adopted a series of recommendations dealing with integrated service digital network (ISDN) matters. ISDN is a network that provides end-to-end connectivity to support a wide range of services, including voice and non-voice services. An ISDN standard interface was defined, called basic access, comprising of two 64 kbit/s **B** channels and a 16 kbit/s **D** channel. Another interface, the primary rate access with an aggregate bit rate of around 1.5 Mbit/s (or 2 Mbit/s) offers flexibility to allocate a set of **H** high speed channels. ISDNs are being integrated in this decade. Major benefits [HAN94] and [KUH94] are:

- Arbitrary transmission speeds of several hundred Mbit/s,
- Low information loss and delay,
- Common user-network interface for a variety of different services,
- Service Integration,
- Provision of new and improved services,
- Enhanced signalling capabilities (e.g. Out-of-band)

The highest bit rate, that the 64 kbit/s based ISDN, can offer to the user is 1.5 Mbit/s (or 2 Mbit/s). However, connection of local area networks (LAN's) or transmission of moving images with acceptable resolution, may require considerably higher bit rates. Considerable technological progress has occurred in recent years and the development of electronics and optical technologies now enables telecommunications networks to operate at very high speeds.

Consequently, the conception and realisation of a Broadband ISDN (B-ISDN) was desirable. ITU-T Recommendation [I.113] ("Vocabulary of Terms for Broadband Aspects of ISDN") defines **broadband** as: "... a service or system requiring transmission channels capable of supporting rates greater than the primary rate." This definition of broadband does not indicate anything about its technical conception. Whereas this definition was settled at the beginning, the final technical conception of B-ISDN only emerged after long and controversial discussions within the standardisation bodies, reflecting the different backgrounds and intentions of the participants.

The first concrete idea of B-ISDN was simply to:

- add new high-speed channels to the existing (ISDN) channel spectrum
- define new broadband user-network interfaces
- rely on existing 64 kbit/s ISDN protocols and only to modify or enhance them when absolutely unavoidable

Service	*Definition*	*Examples*
Conversational Services	Bi-directional (dialogue) communication between terminals without intermediate buffering of user information	– video telephone – video conferencing – high speed interactive data communication – file transfer
Messaging Services	User-to-User communication stored in public domain on demand of individual users	– message handling – voice/video mail – document mail – high resolution images
Retrieval Services	Retrieval of information stored in public domain on demand of individual users	– Hi-fi audio retrieval – video programs – mixed mode document retrieval – data retrieval

Table 1-1 Interactive Services for B_ISDN.

The recent directions taken by B-ISDN are influenced by a large number of the emerging tele-services with different, and sometimes still unknown, requirements. Customers are requesting an ever increasing number of new services ; example of proposed tele-services to appear in the future include: High Definition TV, Video Conference, Videophony, Home Education and Video on Demand. Each of these services make their own demands on the B-ISDN which introduces the need for one universal network flexible enough to provide such services [PRY91].

Table 1-1 and Table 1-2 give an overview of the two major service types (interactive and distribution) and the corresponding sub-classes [KUH94].

Service	*Definition*	*Examples*
Broadcast Services without individual presentation control	Continuous information flow distributed from a central source accessed by an unlimited number of users without control of the start and order of presentation	– television programs – audio programs
Broadcast Services with individual presentation control	Individual access to broadcast services with control of the start and presentation of the information by the user.	– interactive videotex – videography

Table 1-2 Distribution Services for B_ISDN

Services are characterised by their natural information rate. This rate is the rate at which the source generates information if no limitations in terms of functionality or transportation are present. This natural information rate can be represented by a stochastic process. Two important values can be used to broadly characterise this process: the maximum (peak) bit rate **P** and the average bit rate **A**. The ratio between maximum and average (P/A) bit rates is called burstiness **B**. The following table gives example values for some broadband services.

Service	*A*	*B*
Voice	32 kbit/s	2
Interactive data	1-100 kbit/s	10
Bulk data	1-10 Mbit/s	1-10
Standard quality video	20-30 Mbit/s	2-3
High definition TV	100-150 Mbit/s	1-2
High quality video telephony	2 Mbit/s	5

Table 1-3 Broadband Services Characteristics

From Table 1-3 some conclusions can be obtained. There is not a typical service description: all services have different characteristics. None of the services has a burstiness equal to one. A transfer mode able to transport all these described services must be very flexible in the sense that it must transport a wide range of bit rates and should to be able to cope with fluctuating bit rate services [PRY91].

1.2.3 Networking Concepts: The Asynchronous Transfer Mode

Mode	*Description*	*Characteristics*
STM	Synchronous Transfer Mode	– Multiplexing of physical connections based on STD[1] – Circuit switching principle – Constant bit rate traffic streams – Integration of traffic with different but Constant bit rates.
PTM	Packet Transfer Mode	– Multiplexing of Virtual Connections based on ATD[2] – Packet switching principle and variable block length (Packets) – Bursty Traffic streams – Flow control, Error recovery – Special case: Connection less transfer (datagram)
ATM	Asynchronous Transfer Mode	– Multiplexing of Virtual Connections based on ATD – Packetised switching principle of fixed sized packets (cells) – Bursty Traffic streams – Lightweight Protocols: No Flow control – Integration of traffic with arbitrary cell rates

Table 1-4 Main Transfer modes.

In the past, various switching architectures have been developed for different applications, such as voice and data, based on transfer modes. With these notions, three main transfer modes may be defined, Synchronous Transfer Mode (STM), Packet Transfer Mode (PTM) and Asynchronous Transfer Mode (ATM) [KUH94].

[1] Synchronous Time Division.

[2] Asynchronous Time Division.

These switching architectures have been adapted to existing services by using different technologies (from mechanical, semi-electronic to fully electronic switches). But, switching architectures developed for STM and PTM are not directly applicable to B-ISDN [PRY91].

The asynchronous transfer mode is considered to be basis on which B-ISDN is to be built: "Asynchronous transfer mode (ATM) is the transfer mode for implementing B-ISDN ..." [I.121]. The term **transfer** comprises both transmission and switching aspects, so a **transfer mode** is a specific way of transmitting and switching information in a network.

In ATM, all information to be transferred is packed into fixed-size slots called **cells**. These cells have a 48 octet information field and a 5 octet header. Whereas the information field is available for the user, the header field carries information that pertains to ATM layer functionality, i.e. the identification of cells.

Header	*Information field*
5 octets	48 octets

Fig. 1-1 ATM cell structure

ATM uses a label field inside each cell header to define and recognise individual communications. In this respect, ATM resembles conventional packet transfer modes. Like packet switching techniques, ATM can provide a communication with a bit rate that is individually tailored to actual need, including time-variant bit rates.

The term **asynchronous** refers to the fact that, in the context of multiplexed transmission, cells allocated to the same connection may exhibit an irregular recurrence pattern as they are filled according to actual demand.

In ATM-based networks the multiplexing and switching of cells is independent of the actual application. Thus the same piece of equipment can, in principle, handle a low bit rate connection as well as a high bit rate connection, be it of contimous stream or burst in nature. Dynamic bandwidth allocation on demand with a fine degree of granularity is provided. The flexibility of the ATM-based B-ISDN network access resulting from the cell transport concept strongly supports the idea of a unique interface which can be employed by a variety of customers with quite different service needs.

In summary, today's networks are characterised by the coexistence of circuit switching and packet switching, whereas B-ISDN will rely on a single new method called ATM which combines advantageous features of both circuit - and packet - oriented techniques. The former requires only overhead and processing, and, once a circuit-switched connection is established, the transfer delay of the information being carried is low and constant. The latter is much more flexible in terms of the bit rate assigned to individual (virtual) connections.

The use of short cells in ATM and the high transfer rates involved result in transfer delays and delay variations which are sufficiently small to enable it to be applied to a wide range of services, including real-time services such as voice and video. The ability of ATM to multiplex and switch on the cell level supports flexible bit rate allocation, as known from packet networks. Another important feature of ATM networks is the possibility of grouping several virtual channels into one so-called virtual path. The impact of this technique on the B-ISDN structure will be presented in section 2.6.1.

1.3 DOCUMENT OVERVIEW

This work deals with the problem of bandwidth allocation in an ATM link.

A brief discussion to basic ATM issues is presented in Chapter 1.

In Chapter 2 this thesis examines different mechanisms to prevent congestion in ATM networks and related aspects. A brief survey of the related work appearing in the literature is also presented.

Chapter 3 is a presentation of the issues related to congestion control in high-speed networks. The chapter focuses on the utilisation of the Probability of Congestion as decision criteria for Bandwidth Allocation. Emphasis is given to some aspects concerning buffer size and burst length in presence of bursty traffic in an ATM link. Some experiments are presented to illustrate the behaviour in the defined scenario.

Chapter 4 presents in detail the Enhanced Convolution Approach (ECA). First, the basic convolution approach is analysed pointing out the drawbacks. A solution is then proposed: the ECA is defined and it is related how such an algorithm can overcome the draw backs associated with the basic approach. The cost evaluation is addressed and some calculation algorithms are shown.

Chapter 5 addresses the problem of efficient evaluation of network resources by using CAC schemes based on Convolution. Several implementation issues are detailed and a real-time approach as a one-level CAC based on convolution is proposed. The reduction of cost, in terms of storage requirements and calculation amount, is analysed.

Chapter 6 presents a comparison between ECA and other evaluation techniques focusing on bandwidth allocation methods. Measurements obtained in the Basel test-bed corresponding to RACE programme are also included. A comparison with other analytical methods for bandwidth allocation, such as Fluid Flow, Gaussian and linear, is made in the last section.

Finally, the conclusions and possible future research are presented in Chapter 7.

2. Traffic Control and Bandwidth Allocation in ATM networks

The basic objective of a bandwidth management and traffic control strategy in an ATM network is to enable high utilisation of network resources, while sustaining an acceptable QOS for all connections. However, the bursty nature of the ATM traffic imposes strict requirements for traffic control. Traffic control is necessary to avoid congestion. Traffic control methods can be divided in two categories: reactive control and preventive control. Reactive control methods regulate the traffic flow at the access points based on current traffic levels within the network. Preventive control methods provide a fair allocation of bandwidth by requiring, at times of high network load, that each connection's traffic flow remains within specified bounds appropriate for the supported service [R2059]. Due to real time constraints preventive control is more suitable than reactive control in high speed networks [LET94].

2.1 TRAFFIC & CONGESTION CONTROL FUNCTIONS

ITU-T and the ATM Forum have defined a range of traffic and congestion control functions to maintain the QOS of ATM connections.

Traffic Control Functions [PRY91] and [STA95]:

- Network Resource Management
- Connection Admission Control
- Usage Parameter Control
- Priority Control
- Traffic Shaping
- Fast Resource Management

Congestion Control Functions [STA95] and [DEC92]:

- Selective Cell Discarding
- Explicit Forward Congestion Indication

The Table 2-1 shows the hierarchy of the various Traffic and Congestion control functions.

Function	*Traffic description level*	*Protocol Layer*	*Time scale*
NRM	Service, group of connections	Physical, ATM layer (Virtual Path[3])	>= Connection inter-arrival time
CAC	Single connection	ATM layer (Virtual Channel[3])	Connection inter-arrival time
UPC/NPC	Burst, cell	ATM layer (Virtual Path, Virtual Channel)	Inter-burst, inter-cell distance
Shaping	Burst, cell	ATM layer (Virtual Path, Virtual Channel) AAL layer	Inter-burst, inter-cell distance
Priorities	Virtual Channel, Burst, cell	Virtual Channel or lower	

Table 2-1 Hierarchy of Traffic and Congestion control functions.

Network Provisioning (NP) is the set of long term control actions that determine the physical quantities of the resources to be placed in the network. The Network Resource Management (NRM) is the set of control functions related to resource configuration and allocation, which are performed by the network in order to achieve the basic performance and utilisation objectives. Routing is applied in parallel with, or part of, the above three control actions. These relationships are given schematically in Fig. 2-1.

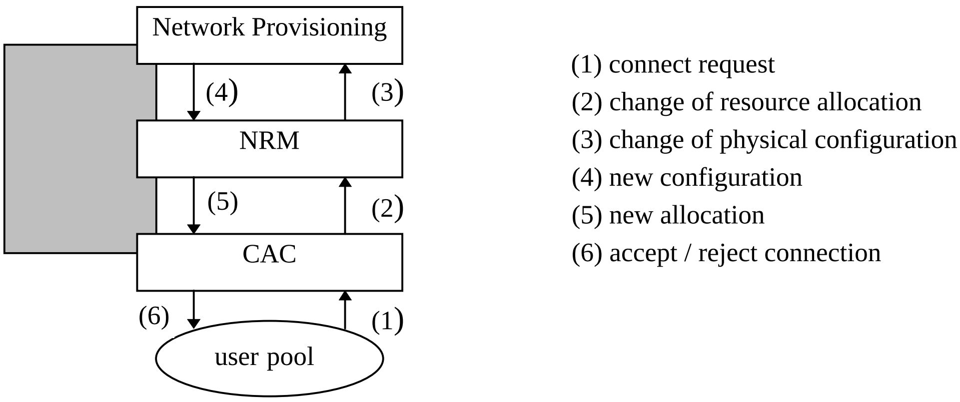

Fig. 2-1Relationship between NP, NRM, CAC and Routing.

This study focuses on preventive control: Connection Admission Control. Related aspects, such as Resource Allocation and the Virtual Path concept, will also be studied.

[3] The Virtual Path and Virtual Channel concepts are described in section 2.6.

2.2 BANDWIDTH ALLOCATION

The major benefit of a broadband integrated ATM network is flexible and efficient allocation of communications bandwidth for communications services [HUI88]. In an ATM network, several sources will be combined in a single link. In a STM network, the required bandwidth on a link will simply be the mathematical sum of all individual fixed bit rates. However, in an ATM network sources are multiplexed and the evaluation of the demanded Bandwidth is more complex; in section 2.6 previous work and several evaluation methods are presented. Bandwidth allocation is a preventive control mechanism [DEC92].

2.3 CONNECTION ADMISSION CONTROL

ATM networks are connection oriented in the sense that, before two systems on the network can communicate, they should inform all intermediate switches about their service requirements and traffic parameters. This is similar to telephone networks where a circuit is set-up between the calling party and the called party. The connections allow the network to guarantee the quality of service by limiting the number of connections.

Connection Admission Control (CAC) is a procedure responsible for determining whether a connection request is admitted or denied [MCD94]. The procedure is based on resource allocation schemes applied to each link and switching unit.

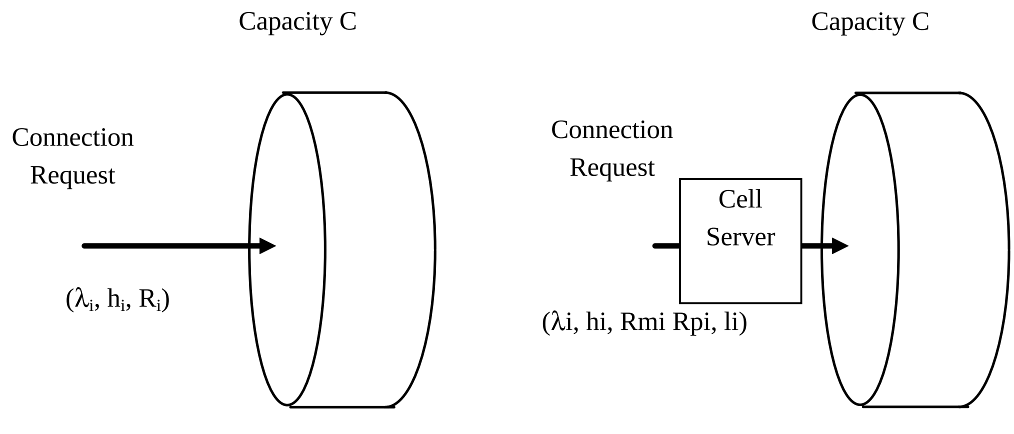

Fig. 2-2 CAC for STM-based Networks and ATM-based Network

λ_i	Call rate
h_i	Average Holding time
R_i	Constant bit rate
Rm_i	Peak Cell Rate
Rp_i	Mean Cell Rate
l_i	Mean Burst Length

In the first phase of B-ISDN, only peak rate CAC algorithms will be applied. The Fig. 2-2 and Fig. 2-4 illustrate the problem of CAC[4]:

The main aspects in the design of a CAC algorithm are speed and simplicity. On the other hand, the bandwidth allocation to the various connections requests must be done in a flexible way that maximises the statistical multiplexing gain [R2059]. Typically, a user declares key service requirements at the time of connection set up and declares the traffic parameters dynamically as demanded by the network.

The main requirements in a CAC procedure can be summarised as follows:[KLE91]

- The network must be protected from overload.
- Resources must be allocated in such away that the QOS Requirements are met for all established connections.
- Maximal statistical multiplexing gain should be obtained.
- The required real-time processing should be reasonable.

The determination of a simple and efficient CAC policy is one of the major challenges in the design and implementation of an ATM-based B-ISDN [YAN 93].

2.4 TRAFFIC CHARACTERISATION

The maximum statistical multiplexing gain can only be achieved if the network knows the probability distribution density function of the individual sources. The network needs a complete characterisation of sources with a known behaviour in statistical terms.

Initially, three different types of sources have been considered: continuous bit rate (CBR) or stream sources, bursty sources (ON-OFF), and variable bit rate sources (VBR). A CBR stream source is modelled as a continuous flow of cells at a constant rate. A bursty source consists of active and idle states. The active state generates a continuous stream of cells at an uniform rate: the idle state transmits no cells. A variable bit rate source consists of a multiple state model, for example variable bit rate video [DEC92]. Each state generates a continuous stream of cells at a constant rate, this model allows to estimate the multiplexer performance with a generic bit rate distribution.

Latterly, the ITU and the ATM Forum are defining new ATM bearer capabilities and associated traffic-control mechanisms [BER95]. Both the ITU and the ATM Forum have chosen a set of ATM Layers, which have different traffic control schemes associated with

[4] This notation of Service Characteristics is detailed in [CAS95 p 545] and [KUH94 p 259 + fig 31 & 32]

them [BER95]. Five ATM Layer bearer capabilities are defined [ATM Forum], [I.371], [CIS95] and [BER95]:

DTR. Deterministic Bit Rate bearer capability, corresponds to CBR sources. DTR is used by connections that request a static volume of bandwidth during the life of connection, typically for circuit emulation.

SBR. Statistical Bit Rate bearer capability, corresponds to VBR sources. A set of standardised parameters describes the behaviour of the connection in statistical terms. This parameters are: Peak Cell Rate (PCR), Sustainable Cell Rate (SCR) and Burst Tolerance (BT) (see details in Table 2-5). The SBR bearer capability can be partitioned in two types: a) real-time (VBR/RT) that requires tightly constrained delay and delay variation, (as voice and video interactive applications), and b) non-real-time (VBR/NRT) where only a maximum cell transfer delay is considered (e.g. data transmissions with QOS guaranteed).

ABR. Available Bit Rate. ABR is designed for applications that can adapt their information transfer rate based on feedback information from the network. This category is designed for normal data traffic such as file transfer and e-mail. The users are allowed to declare a minimum cell rate, which is guaranteed to the connection by the network.

ABT. ATM Block Transfer bearer capability. ABT introduces the concept of a block of cells and transports complete blocks with low cell loss and cell delay variation. In ABT connection establishment no bandwidth is allocated to the connection. Two ABT bearer capabilities are defined: with delayed transmission (ABT/DT), the connection waits for a response of the network before sending a block, and with immediate transmission (ABT/IT), the connection sends the block without waiting for a response, in this case if the network can not support the request rate, the block may be discarded.

UBR. Unspecified Bit Rate bearer capability. The UBR service does not offer any service guarantees. UBR is intended for delay tolerant applications. In contrast to ABR, UBR does not use a feed-back traffic control mechanism, it is a concept similar to the 'best-effort-service' approach. The UBR service is currently the best match to LAN protocols given that the ABR specification has still to be completed.

Focusing On The Problem

Here we point out the CAC problems corresponding to each described ATM bearer. The CAC aspects involved related to the above traffic classes are summarised in the following table:

ATM bearer	*CAC*
DTR	Peak Allocation
SBR	Difficult Bandwidth Allocation
ABR	No Bandwidth Allocation
ABT	Bandwidth Allocation per Bloc
UBR	No Bandwidth Allocation

Table 2-2 ATM bearer capabilities vs. CAC.

DTR connections are allocated using a single peak rate parameter, which implies a very simple CAC strategy. ABR and UBR traffic are not kept down by a CAC method. ABT has special characteristics, we assume that no CAC control is applied in an ABT connection and admission control is only considered per bloc transmitted. This thesis focuses on CAC aspects relating to SBR traffic management.

Adequate traffic characterisation is required to properly design and operate the ATM network, but the wide range of possible future services make this task very complex [KLE91]. Inevitably, any characterisation of traffic must be in terms of the specific times at which cells are generated by the traffic source. The classification of traffic sources in this manner leads to the question: Which parameters should be used by the CAC during call establishment? [BUR91].

QOS REQUIREMENTS & TRAFFIC DESCRIPTORS

Quality of Service (QOS) is defined by specific parameters for cells that are conforming to the traffic contract [MCD94]. QOS is defined on an end-to-end basis, and the main terms of the measurements are illustrated in Table 2-3.

The mean transfer delay is the average of the random component of the Processing Delay and the fixed propagation delay. Cell Delay, Cell Delay Variation and Cell Loss are impacted by buffer size and buffering strategy [MCD94].

Among the above network performance parameters the Cell Loss Ratio (CLR) is the most crucial one in the case of small buffers [R2061/8]. A cell loss rate around 10^{-8} and 10^{-9} for an end-to-end virtual connection is feasible. This value is acceptable for most services, but services that do not support any loss at all, electronic fund transfer is an example, need an end-to-end protection based on a transport layer of the OSI model. Another important issue are time constraints. From this perspective there are two sort of services: Services which have no real time constraints, (data transmission) and services with real time constraints (video, voice, etc.) delay and jitter are new Quality of Service parameters adequate for last [PRY91].

QOS parameter	*Description*	*Components*	*Sub-Components*	
CLR	Cell Loss Ratio	Lost Cells / Transmitted Cells		
CD	Cell Delay	1 Coding and Decoding Delay		
CDV	Cell Delay Variation	2 Segmentation and Re-assembly		
		3 Cell Transfer Delay	Transmission	fixed
			Processing	random[5]

Table 2-3 Main QOS Parameters

In order to make things simpler to users, a small number of pre-defined QOS classes are defined.

These classes are associated with corresponding services. For those connections which do not specify traffic parameters and QOS class, there is a capability defined by the ATM Forum as “best effort” where no guarantees are made and no specific traffic parameters need be stated [MCD94].

QOS Class	*QOS Parameters*	*Application*	*ITU Class*	*ITU ATM layer bearer capabilities*
0	Unspecified	Best Effort Data	-	UBR
1	Specified	Circuit Emulation, CBR	1 A	DBR
2	Specified	Video Audio, VBR	2 B	SBR,ABT
3	Specified	Connection Oriented Data	3 C	ABR
4	Specified	Connection-less Data	4 D	ABR

Table 2-4 ATM Forum QOS classes

[5] Queuing, switching, routing, ..

The ATM Forum traffic parameters (descriptors) are presented in the following table [STA95] and [MCD94]:

Parameter	*Description*	*Metric*
PCR	Peak Cell Rate is an upper bound on the traffic submitted on a connection.	Cells/second
CDV	Cell Delay Variation. This is a tolerance on the variability in the pattern of cell arrivals. This traffic parameter cannot be specified by the user, but the network may to set it.	Seconds
SCR	Sustainable Cell Rate is the maximum average rate of an ATM connection calculated over the duration of the connection.	Cells/second
MBS	Maximum Burst Size is the maximum number of cells that can be sent at the peak rate.[6]	Cells

Table 2-5 ATM Forum traffic parameters

The Concept Of Traffic Contract

A traffic contract is an agreement between the user and the network which is negotiated at connection time prior to traffic entering in the network. A traffic contract exists for every existing connection on the link.

- For permanent circuits, the traffic contract is agreed when the circuit is provisioned.
- For switched virtual circuits, the contract is negotiated using the signalling protocol during the call set-up. Of particular relevance to traffic policing is the negotiation of elements of the connection traffic descriptor.

The Connection Traffic Descriptor specifies the traffic characteristics of a connection. The Connection Traffic Descriptor consists of a Source Traffic Descriptor, CDV tolerance and Conformance Definition.

The Source Traffic Descriptor is the subset of traffic parameters requested by the source (user) which characterise the traffic that will (or should) be submitted during the connection. The Source Traffic Descriptor includes parameters such as the PCR, SCR and BT.

The CDV tolerance is specified indirectly by the QOS class which is another negotiated traffic contract parameter. The various QOS classes are differentiated by their performance

[6] The relationship between MBS and Burst Tolerance (BT) is given in formulae (1) and (2), page 19.

parameters: CDV, Cell Transfer Delay and Cell Loss Ratio. The specific network provider quantifies these network performance parameters for each class. Thus, for each specified QOS class, the network provider will specify the CDV objective as one of the performance parameters.

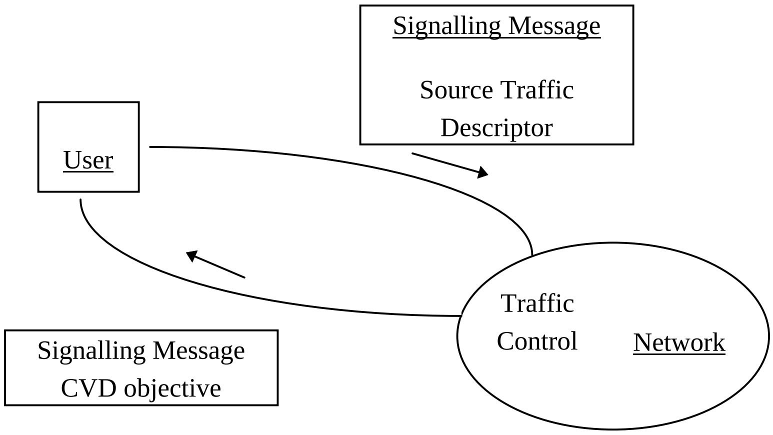

Fig. 2-3 Call set-up: Signalling protocol.

The negotiation of parameters depends on the ATM bearer capabilities. For DTR traffic only the PCR is specified with a tolerance (CDV). An obvious application for the DTR bearer capability is to support circuit-switch emulation. Therefore, SCR and MBS parameters are not needed.

In an ABR service, the cell rate provided by the network can change during the connection. At connection time, the user will negotiate a PCR along with a minimum cell rate and resource management (RM) cells are used as feedback information to control this traffic.

The concept of a bloc of cells is introduced to manage ABT traffic, RM cells are also used for each block transfer request. Policing ABT services is still under discussion. The UBR traffic does not use a feedback traffic control mechanism. Finally, SBR services need all the traffic parameters described in Table 2-5 in [CIS95] and Table 1 in [JAI96]. The following table summarises the necessary parameters during the contract negotiation depending on the ATM bearer capability.

Parameter	*Attribute*	*DTR*	*SBR / RT*	*SBR / NRT*	*ABR*	*ABT*	*UBR*
QOS	CLR[7]	✔	✔	✔	✔	✘	✘
	Delay	Max CTD CDV	Max CTD CDV	Mean CTD	✘	✘	✘
Traffic	PCR	✔	✔	✔	✔	✔	[✔]
	CDV	✔	✔	✔	-	✔	[✔]
	SCR	-	✔	✔	MCR*	**	-
	MBS	-	✔	✔	-	**	-
	Congestion Control	✘	✘	✘	✔	✘	✘

Table 2-6 Parameters negotiation

✔ Specified.
✘ Unspecified.
[✔] Not subject to CAC and UPC procedures.
* Optionally a minimum cell rate (MCR) control is used instead.
** Per bloc strategy.

2.5 USAGE PARAMETER CONTROL

The absence of a channel structure in ATM networks offers greater access and transfer flexibility, but requires a mechanmism to check that the terminal is conforming to the agreed traffic contract [PRY91]. Once a connection has been accepted by the CAC, the Usage Parameter Control (UPC) function monitors the connection to determine whether the traffic conforms to the traffic contract [STA95]. The UPC is often referred to as traffic policing.

The main function of the UPC is to monitor the traffic cell rate for contract violations. However, the UPC also ensures that the traffic submitted has adequate call identifiers[8], as cells with invalid values could cause erroneous cells to be inserted into another connection. The UPC must pass or reschedule (traffic shape) conforming traffic. Any traffic shaped must still conform to the QOS objectives specified in the contract. By discarding or tagging cells, UPC protects the network by ensuring that network resources are available for all users. The negotiated traffic descriptor (see Table 2-5) is used to parameterise the UPC mechanisms for the connection. Depending on the type of service desired, different

[7] For all service categories, the CLR may be unspecified for priority traffic.(Cell Loss Priority bit active: CLP = 1).

[8] See the Virtual Path and Virtual Channel concept in 2.6.1

parameters are exchanged between the user and the network; the PCR plus a CDV and the SCR plus a MBS.

THE GENERIC CELL RATE ALGORITHM (GCRA)

The algorithm defining conformance with the traffic parameters contained in the traffic contract is the Generic Cell Rate Algorithm (GCRA) - or continuous state Leaky Bucket - [ATM Forum] and [I.371]. Using appropiate traffic descriptors the GCRA can be applied as an UPC function.

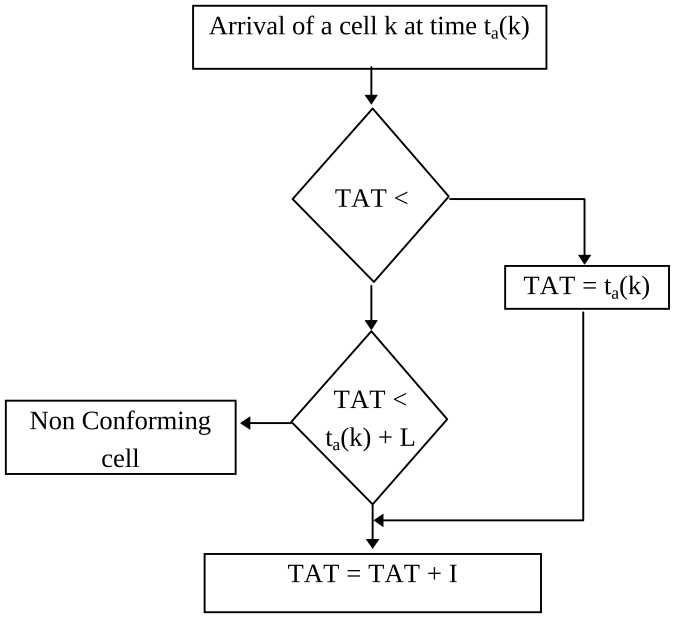

Fig. 2-4 The Generic Cell Rate Algorithm (GCRA).

The GCRA depends on two parameters: increment **I** and limit **L** (GCRA(I,L)). **I** characterises the drain rate of the bucket and **L** characterises the height of the bucket. The greater the height of the bucket, the more cells the bucket can buffer. If the cells are pouring too quickly into the bucket, the bucket will overflow and cells will be lost. In the figure TAT means Theoretical Arrival Time and $t_a(k)$ refers to the arrival time of a cell. According to the ATM Forum, it is necessary to police the PCR, using I = 1 / PCR and L = CDV as GRCA parameters [9].

Specifying a SCR is a further restriction on the submitted traffic and should require less network resources. Specifying a SCR and BT would only make sense for VBR services. These two variables are sufficient to parametrise another GCRA. When the GCRA is applied for SCR policing **I** = 1 / SCR and **L** = BT, but, policing the SCR is optional. The

[9] A different PCR can be stated for the CLP=0 and the CLP=0+1 streams.

following expression relates all described parameters. Actually, the MBS (in cells) and not the BT is sent in the signalling message which must be translated into the BT parameter. This is done using the following equation:

$$BT = (MBS - 1)\left(\frac{1}{SCR} + \frac{1}{PCR}\right) \tag{1}$$

The inverse equation relating BT and MBS is given by the following expression:

$$MBS = 1 + \left\lfloor \frac{BT \bullet PCR \bullet SCR}{PCR - SCR} \right\rfloor \tag{2}$$

Where $\lfloor x \rfloor$ denotes the integer part of the real number x.

In practice, UPC can be quite complex because of the different flows of traffic within a link. The Cell Loss Priority (CLP) bit in the ATM header creates high and low priority flows for a connection. CLP set to 1 means a high priority cell. The PCR and SCR can separate monitoring of the CLP = 0+1 flow (that means the aggregate CLP = 0 and CLP = 1 streams). Additionally, there is a separate set of parameters for each direction of cell flow.

2.6 BANDWIDTH ALLOCATION FOR CAC

2.6.1 Virtual Path Management

Vp Capacity Reservation

Control actions can also be applied to groups of connections sharing common paths through the network. These groups of connections are referred to as Virtual Paths (VP). Each VP may support many Virtual Channel (VC) connections. Allocation of capacity to VPs also has the potential to simplify node structure, node processing, and control of routing and bandwidth [OHT92], [BUR91] and [MIT94].

The role of VPs in traffic management can be summarised, by quoting from [I.371], in the following utilisations:

- To simplify CAC (by reserving resources on a VP basis).
- To implement a form of priority control by segregating traffic requiring different QOS.
- Efficiently distribute messages for Traffic and Control functions by collectively addressing all the Virtual Channel connections with a VP.

To aggregate user-to-user services such that Traffic and Control functions can be applied to the aggregate traffic.

A possible drawback of using VPs is due to the possible waste of bandwidth involved. A number of bandwidth allocation strategies exist and these are described in detail in the literature [HUB93]:

Dedicated bandwidth allocation (non-sharing).

> The VP bandwidth is fixed and constant until the VP is released. This strategy results in low link utilisation, coupled with minimum processing load. Moreover, the bandwidth of each VP can not be shared with other VPs.

Multi-hour engineering.

> The VP bandwidth is not changed at every VC set-up/release, but rather over a period of time.

Statistical multiplexing between VPs (complete-sharing).

> The (unused) bandwidth on a link is completely shared by all VPs on the link [SAT91] and [MIT94]. An advantage is the optimal utilisation of the link bandwidth. This approach reduces the required total path and link capacity because of the statistical multiplexing effects between virtual circuits in path, and between paths in links. The drawback is the bandwidth change required for a VP at every VC set-up and release which causes a high processing load for the VP manager. In [TAK93] a scheme of bandwidth, named Group Virtual Path (GVP), shared among multiple VPs is proposed.

Dynamic bandwidth allocation (partial-sharing).

> Dynamic bandwidth allocation means that the bandwidth allocated to a VP is changed from time to time, in order to adapt to traffic variations and to obtain better resource utilisation [R2061/8], [ARV94], [BUR91], [GER94], [HUB93], [SAT90], [OHT88] and [OHT92]. Since allocation depends on current values of link capacities and offered traffics, changes in these will require the re-evaluation of capacity distributions.

In [OHT92] and [HUB93] similar algorithms to control the VP bandwidth allocation are evaluated: a) Bandwidth allocated to the VP is increased by a specified step if current bandwidth is insufficient for the new VCC; b) if the increase is allowed, then the bandwidth is increased and the VCC is established; if not, the current bandwidth is maintained and the VCC is rejected; and the bandwidth is decreased by a specified step if it is possible, according to the condition of VP utilisation. The [HUB93] paper provides only two bandwidth values for a VP in contrast with [SAT90] which proposes a bandwidth allocation in multiple, load-dependent steps.

As explained in [ANE95], VP capacity allocation algorithms fall into two major categories: synchronous and asynchronous. Synchronous algorithms update the VP capacity based on the observed demand for VCC establishment in real-time. In this context, the bandwidth of the VPs can be expanded or contracted at call arrival or departure times. On the other hand, asynchronous algorithms maintain a fixed VP distribution for a time period T. These algorithms are called asynchronous, because the modifications of VP capacity are not driven by the call arrival or departure process associated with each VP, but are computed at the beginning of the period, and remain fixed during that period. The decision on the VP distribution policy is based on estimates of the offered load during the coming period.

Limiting our consideration to the resource dimensioning aspects, simple models have been proposed to analyse the consequences of managing resources according to VC or VP concepts. A link can be structured as a VP carrying a set of VC each with different requirements depending on whether traffic is integrated or segregated [COST224]. The question of how to multiplex two or more diverse traffic classes while providing different QOS requirements at a switch is a complicated, open problem [HWA94].

When different traffic flows with different QOS requirements, but sharing the same destination node, arrive to a node, as far as local decisions are concerned, two possible simple approaches to virtual traffic management can be taken [COST224], [GUP93], [CHA94], [HWA94], [MOC94] and [FAB95]:

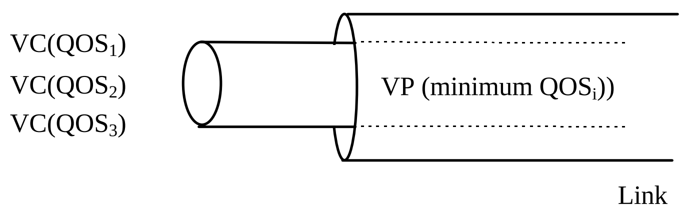

Fig. 2-5. Integration Approach.

In an Integration Approach[10] (sharing) all the traffic from different connections are multiplexed onto one VP, sharing the available bandwidth and buffers. The capacity allocated to the VP should be large enough to satisfy all the individual QOS. Arriving cells from all source types are routed to a single Firtst Come First Served (FCFS) buffer. In absence of any buffer priorities, the same CLP is provided to both traffic types. This implies that the most restrictive QOS requirements must be applied to all services. Therefore, link utilisation will be decreased because unnecessarily stringent QOS criteria are applied to all connections and efficient use of available resources is difficult because source types receive a QOS in excess of the one they demanded. This could produce a lower multiplexing gain than the Segregation Approach. The multiplexing gain is strongly

[10] Aggregation Approach by some authors.

dependent on the ratio between total link bandwidth and source mean bandwidth, concentrating a greater traffic intensity could produce a conceivably higher gain than segregation, even with the most stringent QOS requirements.

With the Segregation Approach (Physical partitioning) the problem can be much simplified if different types of traffic are separated by assigning a VP with dedicated resources (buffers and bandwidth) to each type of traffic. Arriving cells from different source types (or a set of source types) are directed to distinct FCFS buffers, which are then served in accordance to a chosen bandwidth assignment policy. Thus resources may not be efficiently utilised because no sharing of bandwidth can take place across the VP. In this case, advantage can be taken of the less stringent QOS to achieve a high multiplexing gain. As a consequence of the separation, more than one VP may be established between the same source-destination pair, with each carrying different types of traffic. The capacity allocated to each VP should be sufficient to guarantee the required QOS of the traffic classes supported. The sum of all the assigned VP capacities would thus constitute the total capacity required.

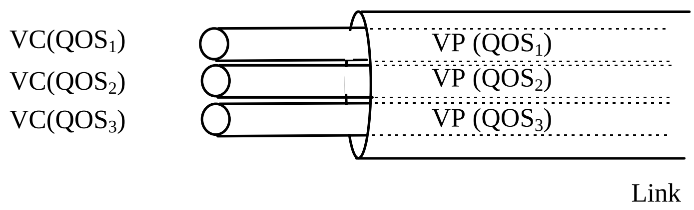

Fig. 2-6 Segregation Approach

Superior performance of one or the other strategy depends on the particular characteristics of the traffic flows involved. There is a need for algorithms that, on the basis of a small set of easily estimated parameters for each traffic flow, can predict accurately the advantages and disadvantages of aggregation and segregation.

The Integration Approach is more adequate in the following cases: (i) the VPs share the same QOS specification ; (ii) the VPs have different QOS specification, but the most stringent ones can be satisfied with a substantial multiplexing gain. On the other hand, segregation is useful in some cases when aggregation produces low multiplexing gain, because the former can take advantage of the less stringent QOS requirement.

If the load of an individual virtual path is not high enough to justify the use of separate links, multiplexing over the same links can save network resources [R2061/8]. Such multiplexing is productive in the following three cases:

- The VPS share the same QOS requirements.
- The VPS have different QOS specifications, but the most stringent ones can be satisfied with a substantial multiplexing gain.
- Priority mechanisms for buffer and/or link rate allocation to different VPS are available that allow for the different QOS required.

Obviously the third case offers the highest possible resource utilisation, but suffers from complexities of implementation and is not always available.

The results obtained in [CHA94] and [FAB95] imply that, when a high peak rate traffic requires less stringent QOS, the segregation approach gives a better performance. On the other hand, when high peak-rate traffic demands the most stringent QOS, the integration approach would be a better choice in terms of less total capacity required.

The Virtual Path Network Model

The concept of a VP is to provide a logical direct route between switching nodes via intermediate cross-connect nodes. It can be considered as the logical equivalent of a link between two switching nodes that are not necessarily directly connected by a single physical link. Thus, VP concept allows a distinction to be made between the physical and logical network structure and gives flexibility for rearranging the logical structure according to traffic requirements.

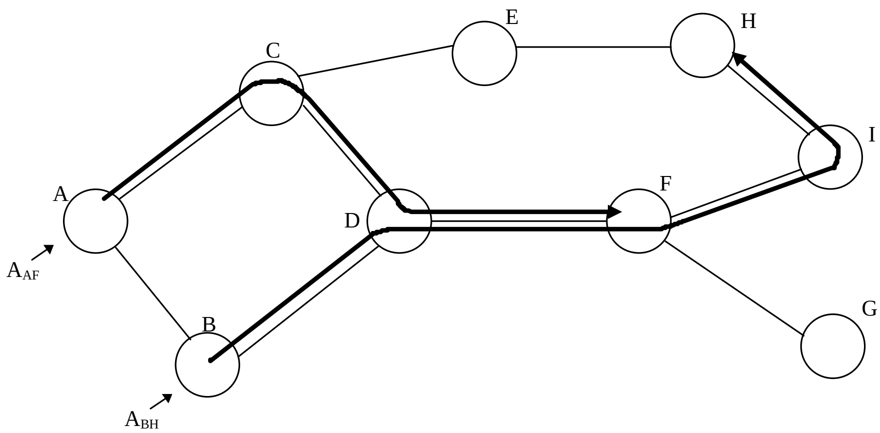

Fig. 2-7 Virtual Path Network

Fig. 2-7 depicts two VPs (AF and BF). Note how a VP can carry connections (VCs) with different QOS requirements which share its networking capacity (DF physical link). VCs within a VP originate at the VP source node and terminate at the VP destination. In this sense, they are "pre-established". Therefore, an incoming call can be assigned one of these VCs at the VP source node, and bypass the call set-up procedure in all intermediate nodes, which would have been required during a normal SVC set-up procedure.

The work discussed in this thesis assumes the Virtual Path Network (VPN) concept. A network of VPs, a VPN, forms a higher layer which is logically independent of the underlying physical network [ARV94b]. VPNs are designed to handle traffic demands with an acceptable grade of service, or, if all demands cannot be accommodated, to maximise some performance metric. For B-ISDN type networks, we may additionally gain simplified statistical multiplexing and grade of service control by grouping services according to their characteristics, e.g. peak rate and burstiness, and demands, e.g. loss and delay, and use separate VPNs for each class. VPN establishment is beyond the scope of this work.

2.6.2 Related work

The exact evaluation of the possible connections onto a link, maximising the statistical multiplexing gain with guaranteed QOS, is a difficult aspect in ATM networks management [CAS91], [HUI88] and [OHT92]. This is due to QOS parameters dependencies: assigned bandwidth and buffer size in a link. Previous work related directly source parameters with both assigned bandwidth and buffer size.

Hierarchical Model

A multilevel congestion and control model mechanism was proposed by J. Hui in [HUI88]. This model defines three different levels : the cell (packet), burst and call level. Those levels are based on the behaviour of the integrated traffic [HUI88]. Different statistical parameters are required to define the traffic at each level [HUI88].

With reference to the QOS parameter Cell Loss Ratio (CLR), we can combine both, cell and burst levels, in a global CLR approximated [11] by:

$$CLR = CLR_{burst} + CLR_{cell} \qquad (3)$$

CLR_{burst} is the dominant factor for large buffers and CLR_{cell} is the dominant factor for small buffers [CAS91], [HAN94] and [MIY93]. It is very interesting to analyse environments with buffers large enough to make CLR_{cell} negligible, but small enough to trail the approximation for CLR_{burst} close to a buffer-less model.

Statistical Multiplexing Gain

Efficiency gain depends on the statistical multiplexing effect of sources, with the condition that enough sources are multiplexed and that they are not correlated. When statistically

[11] The cells which are discarded by the UPC function are not considered.

multiplexing cell loss probability and delay performance do indeed depend on link utilisation and buffer size, as well as traffic characteristics of the connections on the link. This statistical multiplexing gain can only be achieved if the network knows the probability distribution density function of the individual sources. This assumption is not always fulfilled; neither is it an easy job to calculate the convolution at every link where different sources are multiplexed [PRY91].

EVALUATION METHODS

UNI 3.0 allows CAC, traffic shaping and binary feedback. However, the algorithms for CAC are not specified [JAI96 p6]. There are some approximations to evaluate the bandwidth demanded by a set of connections. Some of them are introduced below.

Approximation	*Characteristics*
Equivalent Bandwidth	Each source has an equivalent bandwidth that reflects its characteristics
Fluid Flow	The individual impact of connections is critical
Stationary	The effect of statistical multiplexing is the dominant factor

Table 2-7 Evaluation methods

A brief description of these methods are described below.

<u>Equivalent Bandwidth</u>

It is possible to assign an equivalent bandwidth (effective bandwidth for some authors) to each source which reflects its characteristics. The notion of “effective” bandwidth for each connection aims to summarise in a single parameter the bandwidth and QOS requirements of a connection. The CAC method based on this notion is named Linear CAC and reduces the CAC task to the simple problem of determining whether the sum of the effective bandwidth of each of the connections is greater then the resource capacity; if that is the case the connection is rejected, otherwise it is accepted.

Several network traffic control functions such as congestion control and routing depend on the characterisation of the equivalent bandwidth of individual connections and the resulting load on network links. A major challenge is to provide traffic control functions in real-time. Normally this involves a reduction in the complexity and, of course, in the accuracy of the evaluation models. Verifying that the aggregate equivalent bandwidth remains below the link capacity is typically a very complex task. In heterogeneous scenarios this approximation may be unfeasible [DEC92].

At the burst level, two different approaches for equivalent bandwidth evaluation are studied by [GUE91] and [GAL89], in which different aspects of the behaviour of multiplexed connections are considered and fluid-flow model and stationary bit-rate distributions are presented. The fluid-flow model is also studied by [CAS91].

The linear approximation may be too optimistic. It is observed that the linear approximation gives accurate results provided that the activity of the sources is not too large. In cases where the linear approximation fails (activity factor of one or more types of sources is greater than 0.8), a simple non-linear approximation [LEE94] is suggested.

Fluid-Flow approximation

The fluid-flow model estimates the equivalent bandwidth when the individual impact of connections is critical. This model does not consider any multiplexing aspect. The cell loss probability is a function of the ratio between the buffer capacity and the average number of cells sent during a burst [CAS91].

The fluid-flow model assumes that the information arrives uniformly during a burst and that the server removes the information from the queue in the same manner. A Markov process {X(t),Y(t)} is used, where X(t) is the number of active sources and Y(t) is the Queue length. Differential equations can be derived for the distribution of {X(t),Y(t)}, leading to a system of (N+1) equations, if N is the number of the homogeneous sources [CAS91] and [GUE91].

In general, the fluid-flow model is valid when the buffer capacity is longer than the mean burst duration.

Stationary approximations

In this case the effect of statistical multiplexing is the dominant factor, and it considers that cells are lost when the instantaneous rate is greater than the bandwidth provided by the link. Small buffers are not effective at the burst level. Three methods are introduced below.

- *Binomial*

  The distribution of the aggregate bit rate on a link can be determined from the stationary distribution of the Markov chain formed by the superposition of sources [GUE91]. In case of identical two-state Markov sources, the probability P_k, defining whether k sources are active, is given by a binomial distribution:

$$P_k = \binom{N}{k} \rho^k (1-\rho)^{N-k} \tag{4}$$

Computing the smallest integer **k’** so that

$$\sum_{k=k'+1}^{N} P_k \leq \varepsilon \qquad (5)$$

N	Number of sources
k	Number of active sources
P_k	Probability that k sources are active
ρ	Probability that one source is in active state
ε	Maximum admitted Probability of Congestion.

where the demanded bandwidth is k’·Rpeak[12]. Therefore, the expected traffic characterisation is: ρ, (the probability that a connection is active), and Rpeak per connection [D122].

- *Gaussian*

The two-moment allocation scheme (also referred to as the normal approximation), assumes the independence of the traffic behaviour of the connections and characterises the multiplexed traffic by a normal distribution with parameters given by the sum of the means and the sum of the variances of each connection. A connection is only accepted if the congestion probability derived from the tail of the normal distribution is less than a pre-specified threshold.

The assumption of a Gaussian distribution allows us to use standard approximations to estimate the tail of the bit rate distribution [GUE91]. An approximation is given by:

$$\sum_{j=0}^{N-1} m_j + m_n + f_\varepsilon \sqrt{\sum_{j=0}^{N-1} \mathrm{var}_j + \mathrm{var}_n} \leq C \qquad (6)$$

N	Number of sources
C	Capacity of the link
m_i	Mean rate of i-connection
fε	Epsilon fractile

where m_n is the mean rate of the new connection, var_j is the variance of the j-connection rate, var_n is the variance of the new connection rate, and fε is a constant value depending on the admissible probability of loss. The expected traffic characterisation is mean m_j and variance var_j per connection [D122]. The Gaussian assumption is not applicable when there are small numbers of very bursty connections with high peak rates, low utilisations, and long burst periods [GUE91].

[12] Rpeak corresponds to the maximum instantaneous rate of the source.

- *Convolution*

The exact distribution of the aggregate bit rate on a link can be determined by convolution using the exact bandwidth requirements of each traffic type. This method is based on the formula:

$$P(Y+X=b)=\sum_{k=0}^{b} P(Y=b-k)P(X=k) \qquad (7)$$

X	Bandwidth requirement of the new connection
Y	Bandwidth requirement of existing connections
b	Instantaneous rate on the link

Y refers to the bandwidth requirement of already established connections; **N** is the bandwidth requirement of a new connection, and **b** denotes the instantaneous required bandwidth. The above expression allows the evaluation of the distribution function of the demanded bandwidth on a link; this method is explained in detail in Chapter 3.

The Decision Criterion in order to accept a new connection **X** when convolution is used in CAC is based on the Probability of Congestion (PC):

$$\mathrm{PC(Y + X) = P([Y + X] > C)} = \sum_{b>C} P(Y+X=b) < \varepsilon \qquad (8)$$

ε Admissible probability of congestion.

<u>Heuristic Methods</u>

The other group of CAC approaches is based on heuristics and data modelling techniques. The neural network and fuzzy logic based approaches are example of this kind of approaches. Heuristic approaches provide a mechanism for clustering data obtained from ATM traffic measurements in a structure that constitutes the traffic model, e.g. a net structure composed of a set of neurones and respective connections for neural nets and a rule structure composed of a set of “if-then” associations of variables in the case of fuzzy systems.

2.6.3 Most Common referenced work

Many authors have studied several bandwidth allocation strategies in ATM networks ; some of these studies are presented in this section.

As noted above, in [HUI88] J. Hui studies traffic blocking on a link in a hierarchical multi-level scheme at three levels. These levels are: cell level, burst level and call level, each having an associated time scale parameter. Different approaches are presented at each level to allocate resources guaranteeing no excessive blocking at the next level. Analytical approaches to the blocking probabilities are presented in his work.

At the burst level, two different approaches for effective bandwidth evaluation are studied by [GUE91] and [GAL89], in which different aspects of the behaviour of multiplexed connections, fluid-flow model and stationary bit rate distribution are presented. Guerin et al. offer some approximations for both the effective bandwidth of a single connection and the aggregate bandwidth usage for multiple connections. Those approximations are better if the number of connections is large and homogeneous.

In [OHT92] a bandwidth control method based on the VP concept is presented. The bandwidth of a virtual path is defined by the maximum number of VCs that the VP can carry. The control which dynamically reassigns the virtual path bandwidth, significantly improves transmission efficiency with constant call blocking. The bandwidth control mechanism described in [OHT92] is an efficient method of flexibly reassigning individual virtual path bandwidth. When the connections on one virtual path increase, any available unused link capacity is assigned to the busy virtual path. Thus, transmission efficiency is improved because each virtual path in the link is well utilised. Although such control may increase processing load, some advantage in reduced node processing is expected to be maintained by changing the bandwidth less frequently than call set-up and clearance. Any request for a bandwidth increase or decrease is carried out by the specified steps.

The assumptions used in this work are made for simplicity: every call has an identical bandwidth and no statistical multiplexing effect is considered in allocating the bandwidth for a call or a virtual path, i.e. the bandwidth is allocated deterministically. A problem with this control scheme is that the call loss probability for each virtual path differs depending on the traffic offered to it.

Some Congestion Control mechanisms are presented in [DEC92] and [CHU95]. These papers study the source traffic descriptions, and propose an equivalent bandwidth for bursty data and variable bit rate video (VBRV). A multiple state model is presented for the latter. A rather small buffer size is assumed to guarantee an acceptable maximum delay, e.g. 50 cells in [DEC92] and 100 cells in [CHU95].

An important conclusion from these results is that if the burst length is longer than the buffer length, the equivalent bandwidth measure does not depend on the buffer length, but on the average burst length and the distribution of the activity duration.

In [WOR92] T. Worster points out that the training requirements imposed by a neural network approach, do not map conveniently the CAC fast response requirement. Fuzzy based approaches require a less lengthy training phase than a neural network approach and their operation is easier to understand and update.

The RACE (Research and development in Advanced Communication technologies in Europe) programme of the European Community in R1022 (deliverable 122, 124 and 126) project, and R2061 (deliverable 8, 18 and 46) project, present an exhaustive report focused on control schemes, traffic models and applications.

In deliverable [R2061/7] various Usage Parameters Control (UPC) functions have been validated using artificial sources generated by ATM test equipment. These studies were extended to include, as far as possible, real sources as well as measurements to validate the Connection Admission Control (CAC) function. While both the UPC and the CAC functions were studied separately in previous work, this deliverable focuses on how these two functions are integrated to form a complete and robust control framework.

The UPC is based on an implemented dual leaky bucket to control the PCR and the SCR. The CAC procedure is a convolution-based method in most of the experiments. This procedure is based on a priori knowledge of the stationary cell rate distribution of a source. The implemented CAC function consists of a two-level procedure[13].

Measurements with multiplexed sources whose cells are clumped back-to-back, have been carried out in order to determine the Cell Loss Ratio (CLR) will be in this case. Measurements are compared with theoretical results and the agreement is good. The traffic control framework proves to be consistent as long as the contracted Cell Delay Variation (CDV) tolerance τ is kept small enough to avoid that the cell level queuing effects (not taken into account by the CAC) becoming dominant.

The convolution algorithm always represented a conservative allocation of resources while achieving significant multiplexing gain allowing for the small buffers in the switches at the test-bed. Since none of the two UPC tolerance parameters, Cell Delay Variation (CDV) tolerance and burst tolerance, are taken into account by the CAC function used, a possible impact on the multiplexer performance from sources utilising these UPC tolerances maximally, has been investigated. Hence, the framework has proven to be able to handle the investigated traffic safely and efficiently, based on the contracted PCR and SCR parameter sets in a network with small buffers.

The SCR used by the CAC and the BT are important in order to ensure that SCR really may be used as the mean cell rate in the CAC. To achieve this the BT must be small compared to the connection holding time, otherwise the assumed mean cell rate would not be valid. In most of the experiments performed all connections have been set up for the duration of the experiment and with a BT in accordance with such duration.

[13] See details in Section 4.2.2 ‘Two level CAC’.

2.6.4 Drawbacks

Several limitations have been found in the previous studies: Inter-dependencies, Source model dependencies, heterogeneous environments, calculation effort accuracy AND individual cell loss probability. These limitations are described below.

<u>Inter-dependencies</u>

All studied models describe the behaviour of the sources without considering their interactions inside the network. There could be individual sources which have a typically correlated structure, e.g. multimedia traffic [HER94].

In [AU95] the worst data loss behaviour of a statistical multiplexer with limited buffer capacity is investigated. Buffer overflow and data loss depend only on the available buffer capacity and the degree of correlation among the input traffic. The feasibility of performance objectives in ATM networks with correlated traffic is also studied in [HEE93], the Markov Modulated Poisson Process (MMPP) model is chosen for arrival processes.

<u>Source model dependencies (number of states)</u>

Only sources of two-states (normally ON-OFF traffic) are presented. Multi-state models are often discarded because of the difficulty of evaluation methods.

<u>Heterogeneous environments</u>

Normally, the performance of the evaluation approximations loses accuracy in heterogeneous scenarios. In these environments there are a trade-off between the Integration vs. Segregation approaches. Additional issues, about the traffic model and QoS parameters, have to be considered to choose one of these approaches as an effective method for Bandwidth Allocation.

<u>Calculation effort and accuracy</u>

Accurate evaluations have been simplified in order to reduce the complexity of calculations and the required memory, consequently, a reduction of the accuracy is obtained. In [GUE91], the convolution approach is first applied solely as a binomial distribution over homogeneous sources. Later, the Gaussian distribution is proposed as an approximation to the exact value.

Cell Loss Probability evaluation (individual CLR)

> Normally, different classes of traffic are segregated to different VPs. Therefore, all individual connections have the same QOS. Nevertheless, it may be more efficient to transport different classes of traffic by the same VP; then, connections on the same VP have different CLR, and, thus, implies different QOS for different classes. This individual CLR_i for each class of traffic is difficult, or impossible, to obtain.

These drawbacks are more profoundly analysed in the next chapters. Also, an approach to overcome the above drawbacks is presented in Section 4. This approach, called the Enhanced Convolution Approach (ECA), is based on the convolution algorithm.

3. The Probability of Congestion as BW Allocation Decision Parameter

In this chapter, the utilisation of the Probability of Congestion as a decision parameter is introduced. A detailed analysis of the relationship between PC, CLR, buffer size and other QOS parameters is also presented. The range over which the metric proposed is valid is studied in section 3.4.

3.1 ATM NETWORK MODEL

ATM offers maximum flexibility with an optimal use of available resources and applicability to all kind of services. This flexibility is applicable both in terms of value of the bit rate as in terms of the behaviour in time (from constant to very fluctuating). Therefore, ATM networks introduce some cell loss and delays which the terminals have to solve at application level [PRY91] and [DEC92].

There are some services for which the QOS has real time constraints (i.e. interactive services), for Cell Transfer Delay, and Cell Delay Variation. Therefore, very large buffers cannot be introduced and buffer dimensioning is carried out taking into account the cell level contention. Also, suitable buffer sizes can be selected to ensure that the maximum cell delay is less than a pre-specified limit [YANG93]. Under that premise, Cell Loss Ratio is the major relevant Parameter of QOS [CAST91]. However, QOS parameters also include Cell Transfer Delay (D) and Cell Delay Variation (CDV).

The global network performance, in particular the end-to-end performance observable by the user, depends to a high degree on all network component performance figures and on globally operating strategies such as routing of connections, CAC algorithms bandwidth management, and network management. Of particular interest is the cell delay variation (CDV) of an end-to-end virtual channel connection; the CDV must be kept within reasonable small as buffers have to be applied.

Applications goals corresponding to insensitive (to D, CDV and CLR) traffic; and interactive services are considered. Therefore, the buffer size of the statistical multiplexer is assumed to be small (e.g. 50 cells), in order to guarantee an acceptable maximum delay (e.g. 140 µs. with the link rate L = 150 Mbit/s) [DEC92]. LAN interconnect is an important example of a service which requires both a low loss rate and a low end-to-end delay [WRI89] due in the main to time-outs in the LAN protocols.

Applications must solve the communication problems since the ATM network needs a simple network management ; one single bearer service for all classes of traffic. Terminals and end-to-end protocols must be able cope with cell losses, and thus CLR becomes the critical QOS parameter.

3.2 PC AND SOURCE TRAFFIC CHARACTERISATION

3.2.1 Offered and Carried Traffic

The offered traffic in a link[14] **W** is evaluated by summing the traffic of all active calls.

$$W = \Sigma \text{ Input rate} = \Sigma\, r_i \quad ; r_i = \text{rate of the active call} \qquad (9)$$

Given several sources emitting cells at the corresponding velocity (depending on the current state of each source), the cells are saved temporally in the output buffer. These cells are extracted from the buffer at a constant rate **C**. If cells arrive at a rate greater than **C** (during a relatively long period) the capacity of the buffer may be surpassed and consequently some cells are dropped.

$$W > C \text{ (Available link Capacity)} \qquad (10)$$

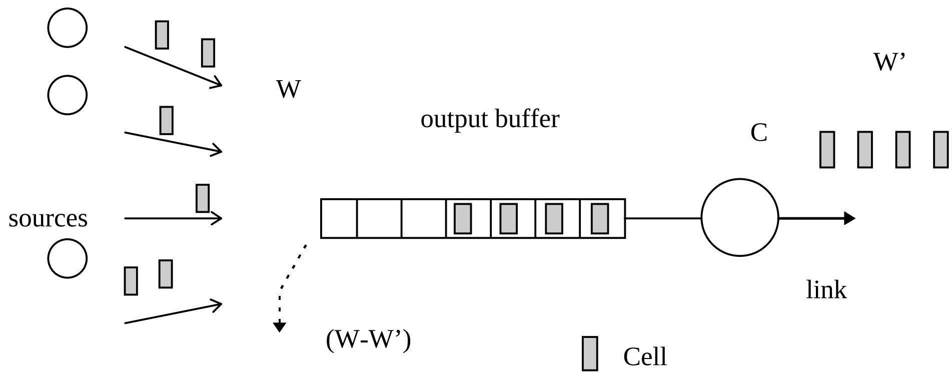

Fig. 3-1 Offered traffic Model

The carried traffic corresponds to the load of the output resources **W'** considering the capacity of the link **C** in terms of velocity. If **W** is greater than **W'**, the link is in congestion and the difference between **W** and **W'** is the lost traffic.

3.2.2 Congestion Control Techniques

Table 3-1 shows how the duration of congestion affects the choice of the method. The best method for networks (that are almost always congested) is to install higher speed links and redesign the topology to match the demanded load. For sporadic congestion, one method is to route according to the load on links and to reject new connections if all paths are highly

[14] In this work, link and VP are interchangeable expressions in the sense that a VP has associated resources (velocity and buffer) as a physical link.

loaded. This is the CAC that is effective for medium duration congestion, since once the connection is admitted the congestion may persist for the duration of the connection.

Congestion Duration	*Congestion Mechanism*
Long	Capacity planning and network design
↓	Connection admission control (CAC)
	Dynamic routing
	Dynamic compression
	End-to-end feedback
	Link-by-link feed back
Short	Buffering

Table 3-1 Congestion techniques for various congestion durations.

For congestions persisting for less than the duration of the connection, an end-to-end control scheme can be used, i.e. during the connection set-up, the sustained and peak rate may be negotiated. A leaky bucket algorithm may be used by the source or the network to ensure that the input meets the negotiated parameters. In a closed loop scheme sources are informed dynamically about the congestion state of the network and are asked to increase or decrease their input rate. The feedback may be used in a hop-by-hop fashion at datalink layer, or in an end-to-end basis at transport layer. For very short variations in traffic load, providing sufficient buffers in switches is the best solution.

Notice that solutions that are good for short term congestion are not good for long-term overload and vice-versa. A combination of various techniques rather than just one technique is used, since overloads of various durations are experienced on all networks.

3.2.3 The Probability of Congestion

This work focuses on the congestion prevention of call congestion duration; the other shorter term congestion techniques presented in Table 3-1 are very difficult to use for traffic with real-time constraints. Only small buffers are considered for short duration congestion in this chapter. Cells are multiplexed statistically among the links for transmission, so a buffer is needed. If the buffer is full, some cells may be lost. By using buffers, a delay is introduced which has a random behaviour depending on the characteristics of the traffic. The following figure shows the instantaneous aggregated rate of all sources connected against time.

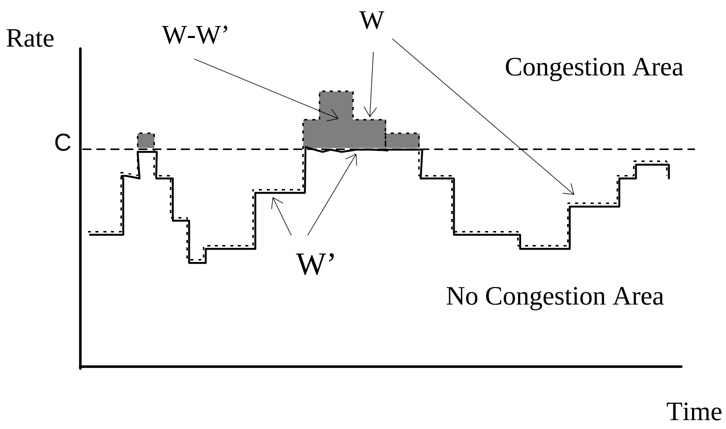

Fig. 3-2 Congestion: Time scale

The following figure shows the probability associated to a given instantaneous aggregated bit rate of all sources. The figure shows that all situations corresponding to rates greater than **C** (at right of **C**) are in a congestion state.

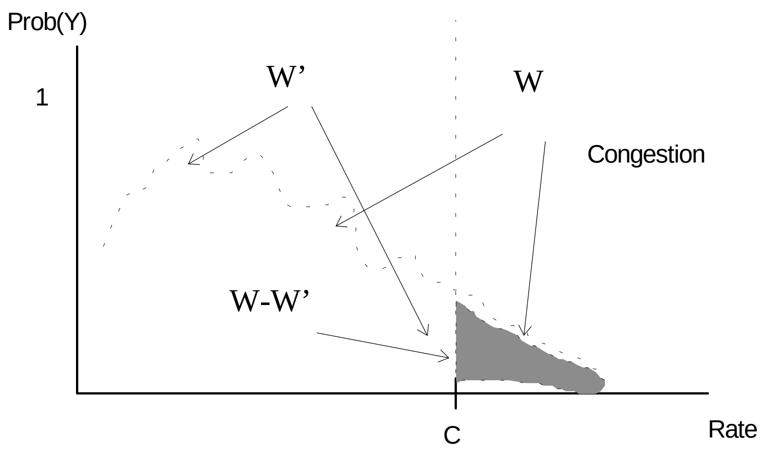

Fig. 3-3 Congestion: Probabilistic scale

The probability of congestion (PC) is the sum of probabilities corresponding to rates greater than C, which is the shadow area (see Fig. 3-3). The probability of congestion does neither state how many cells are lost nor the duration of the congestion state, but only that there is cell loss [IVE91].

3.2.4 The PC evaluation

There are two ways to evaluate the PC in an ATM link: empirical methods based on the declared statistical traffic parameters, and methods based on measures acquired directly from the system. From the point of view of the QOS, learning methods have a important problem: normally connections do not use all their assigned bandwidth, but it is possible that all connections, or part of them, suddenly become active so that the traffic control functions cannot avoid an excessive loss of cells. Consequently, empirical methods are more widely used in this work.

The load region admissible is approximated using parameters [KLE91] such as; the mean load, the congestion probability, and the ratio of cells exceeding the link capacity for the total cell stream and for each individual connection: CLRi.

The mean bandwidth required by a link is evaluated by applying:

$$E(Y) = \sum_{i=1}^{S} N_i \bullet E(X_i) \qquad (11)$$

Y	Rate distribution of the instantaneous offered rate
N_i	Rate distribution of the source i.
S	Number of sources.

For arbitrary mixes, the load of the link may provide only little information about the cell loss probabilities [KLE91]. Cell losses are quite likely if the bandwidth required at the burst level exceeds the capacity of the link. These events are taken into account by the probability of congestion (PC) [KLE91].

$$PC(Y) = P(Y > C) = \sum_{L>C} P(Y = L) \qquad (12)$$

C Capacity of the link.

However, the PC does not give any information about the number of cells lost in case of congestion unlike Cell Loss Ratio (CLR). In a short congestion state, all cells may be buffered with no cell losses occurring. Nevertheless, when a burst's duration is longer than the size of the buffer, then almost all cells exceeding the link capacity are lost. In this case, the relation between PC and CLR is approximated by:

$$CLR(Y) = \frac{\sum_{L>C} (L - C) P(Y = L)}{E(Y)} \qquad (13)$$

If the buffer size is sufficient for cell contention, the evaluated CLR provides an upper bound to the total cell loss probabilities [KLE91]. The PC model is an stationary approximation, in other words, a probabilistic scheme.

3.2.5 The small buffer assumption

To work with small buffers implies that the traffic bursts cannot be saved in the buffer. Therefore, the burst length is irrelevant because all cells will be lost. Cell scale losses are the object of the buffer dimensioning problem and the burst level approximation concerns the Call Admission Control [CAST91]. On the other hand, it is not likely that users will be able to supply information about the burst length at connection set-up. Moreover, it might be difficult to guarantee a certain burst length distribution for policing functions [KLE91].

3.2.6 Source parameters

The CAC software translates the converted (PCR, SCR) pair into a cell rate distribution of a matching ON/OFF source [Del126] and [R2061/46]. Source parameters Peak Rate, Mean Rate and Burst Length will be identified as Apeak, Amean and Bmean respectively. The statistical distributions - Velocity and Probability - for each state, ((V_{off},P_{off}), (V_{on}, P_{on})), will be obtained as follows:

Velocity (V)	*Probability (P)*
$V_{off} = 0$	$P_{off} = 1- (Amean /Apeak)$
$V_{on} = Apeak$	$P_{on} = Amean /Apeak$

Table 3-2 On-off source model parameters

Corresponding to the following graph:

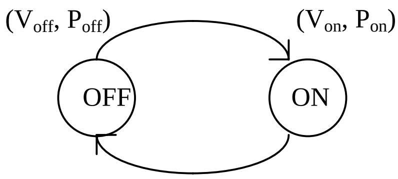

Fig. 3-4 On-off source model state diagram

From more recent traffic descriptors CDV, PCR, SCR, and BT[15], PCR can be derived from the Peak Bit Rate and SCR may be approximated by the Mean Bit Rate. The SCR must assume at least slightly above the MCR of the source [Del126]. For a small BT, such as real-time traffic, the SCR is usually larger than the mean rate for long term, this implies that to approximate SCR to mean rate is accurate enough in small buffer environments [R2061/7].

In the above scenario, temporal references are not taken into account and this is a stationary model. Results are insensitive to burst length. But burst length is a traffic parameter that is difficult to declare and police [LET94]. The presented model allows **n** possible states, but the disadvantage is that no policy function is available when sources have 3, or more, possible states.

3.2.7 PC and QOS parameters relationship

Based on the previous premise, CLR is considered as the prime QOS parameter, and can be represented by the PC in a buffer-less model. This means that no-buffering at burst level must be considered and consequently, cells sent are estimated lost during the congestion period. Therefore, the upper limit for CLR is evaluated.

[15] These traffic descriptors are detailed in Section 1.

The accuracy of this assumption depends on both buffer size and burst length. A set of simulations are presented to illustrate how close the actual CLR is to the evaluated CLR. It is known that an increase in burst length leads to an increase in the CLR. Results are shown in the next section which demonstrate this point.

Results provided by both analysis and simulation will be presented. The CLR is the single metric considered. A set of experiments (considering different number of sources, buffer size and output link rate) are presented. Sources are grouped into classes with identical traffic parameters.

In the analysis, a convolution approach is used to obtain the corresponding CLR evaluated on the basis of Probability of Congestion. This evaluation ignores both buffer-size and burst length. Note that the convolution approach is a stationary model. The simulator can support three parameters for the traffic model [GUE91] giving mean rate, peak rate and mean burst length. Burst length is considered as a negative exponential distribution which is generated for the sojourn-time. The parameters used for simulation correspond to the source model as explained in 3.3.3 (see also Table 3-4). These parameters are V_{off}, P_{off}, V_{on} and P_{on} based upon mean rate, peak rate and mean burst length value as shown in Table 3-2. The simulator directly gives CLR as a final result.

3.3 SIMULATION VERSUS ANALYSIS RESULTS

3.3.1 Objective

Although the addressed traffic engineering items are by no means exhaustive, the dimension of the problem may be anticipated, methods and assumptions must be validated by measurements. But, many new applications or services specific to ATM networks do not yet exist. Therefore, validation of traffic assumptions and analysis procedures must be done. One method of validation is simulation. In a computer simulation the simplifying assumptions of analytic techniques may be validated [KUH94].

The objective of this section is to obtain the buffer-size range in which the stationary model is sufficiently accurate. In next section, the minimum buffer size necessary for cell contention is analysed. On the other hand, large buffers with respect to the mean burst length causes the stationary model to represent too conservative a position. A range of values for the maximum buffer size (which is adequate to apply the analytical-based buffer-less model) are discussed. Finally, the accuracy of the presented method is compared with the Fluid-Flow approximation when varying source the utilisation.

3.3.2 Environment and Statistical Considerations

In this section, different experiments have been carried out to obtain a set of CLR results by varying both buffer-size and mean burst length. The analysis method using convolution gives exact values assuming no buffers at burst level.

Metric	Evaluation	Value
Total slots	Pre-fixed	10^8
Link utilisation	Pre-fixed	0.2
Total sent cells	Link utilisation $\cdot$ Total slots	$0.2 \cdot 10^8$
CLR goal	Pre-fixed	$\approx 5 \cdot 10^{-4}$
Total cell lost	Total sent cells * CLR goal[16]	$\approx 10^4$

Table 3-3 Simulation Metrics for each experiment

Simulations have been carried out by defining a number of experiments according to the confidence interval desired. For each point plotted in the following figures a set of 5 experiments, each consisting of 10^8 slots, have been analysed at the output buffer. Given a load equal to 20%, if the CLR is fixed to. 5 10^{-4} ; about 10^4 lost cells are counted in each experiment. This value is considered representative.

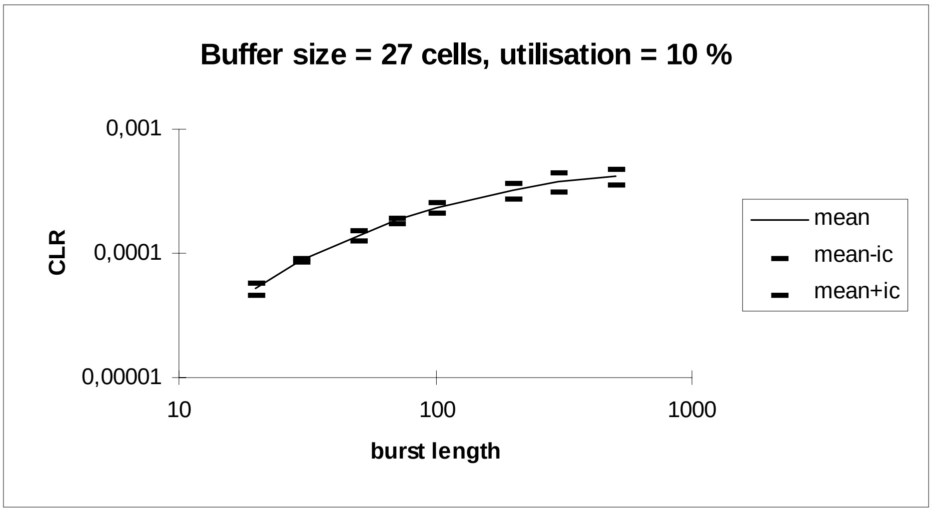

Fig. 3-5 Confidence interval.

For each traffic scenario the width of the 95% interval of confidence (**ic**) is less than about 10% of the estimated probability of cell loss. The above figure shows the width of the 95% confidence interval for a given set of experiments

[16] Cells sent $\cdot$ CLR goal = Link utilization $\cdot$ Total slots $\cdot$ CLR goal = $0.2 \cdot 10^8 \cdot 5 \cdot 10^{-4}$

The Fig. 3-6 plots two series of values corresponding to both theoretical and simulated distributions for a given experiment. The x-axis corresponds to the number of active sources (from 0 up to 40) and the y-axis corresponds to the probability that **n** sources are emitting cells (in activity). While the associated probability is less than 10^{-8} approx. the values evaluated by both are satisfactory. Due to limitations of the simulation when it continues for a long period two effects can be observed close to point **U**: first, there are slight deviations from the theoretical predictions and secondly no values are obtained by the simulation. Deviations are not very important in absolute terms, moreover, they can be observed in the figure because a logarithmic scale is used for probabilities. The simulation does not settle in these states a sufficient number of times, therefore, the effects which are mentioned come into play.

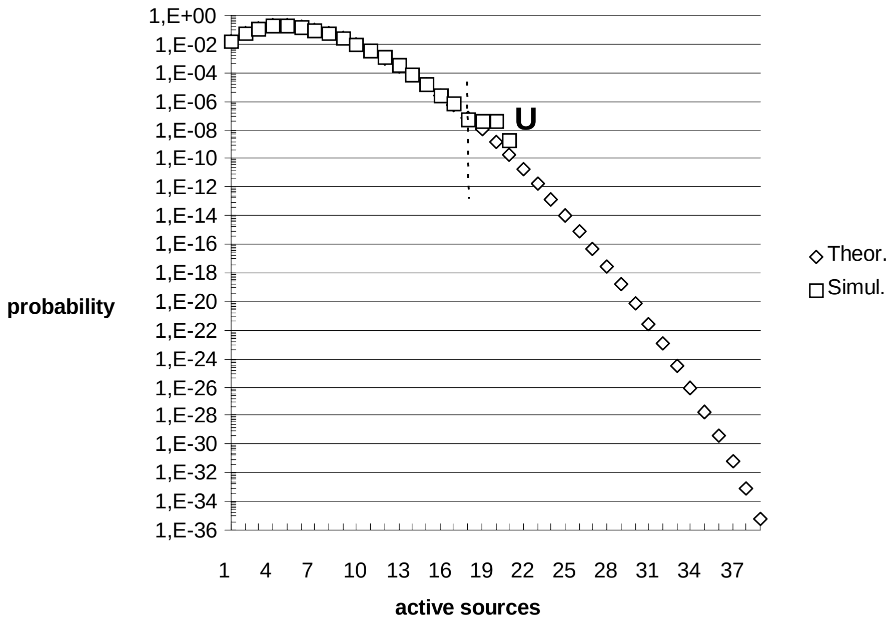

Fig. 3-6 Theoretical vs. Simulated Distribution

Cell and burst behaviour

Congestion, leading to cell delay and cell loss, within an ATM link is caused by two different effects. The asynchronous arrival of cells from different connections cause a short-term (cell level scale) congestion, because several cells may arrive almost simultaneously. Also, if the offered bit rate exceeds the link rate, a long term (burst level scale) congestion occurs because of the duration of the active state of the connections. A distinction between these behaviours on the output buffer of the link can be observed in the following figure where CLR is plotted against the buffer size.

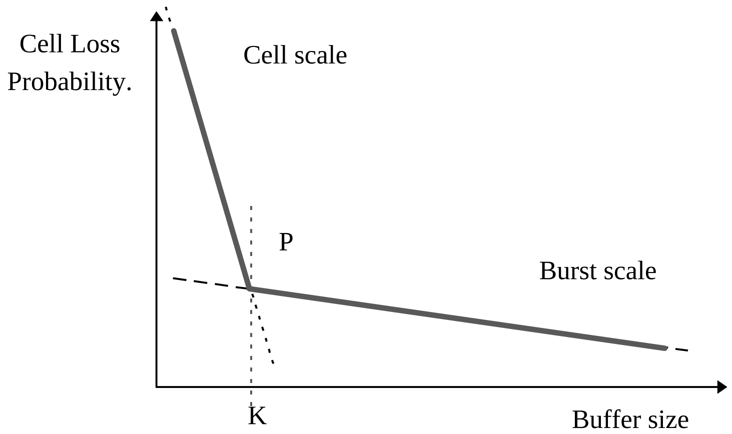

Fig. 3-7 Cell and burst time scale behaviour

The queuing behaviours are different and can be modelled separately. A point **P** is given where the cell loss probability obtained from two behaviours is the same, this occurs for a buffer size equal to **K**. For a buffer size less than **K** the cell-level-scale is the dominant factor (the slop depends on the load), if the buffer size is greater than **K** the burst-level-scale is the dominant factor (depending on the mean burst length). Results from both models can be combined to obtain the overall cell loss probability [CUT93].

3.3.3 Experiments

To validate the proposed model, a set of experiments have been carried out. The performance for both cell-level and burst-level is analysed using an extreme range of values. The scenario of these experiments corresponds to several sources which are emitting cells onto a single link (see model in Fig. 3-1).

Cell &Burst behaviour test

The number of sources used in the simulations is a trade-off between an adequate multiplexing behaviour (large number of sources) and a reasonable duration of simulations. 40 sources is a number of connections which achieves both objectives. A value equal to $5{\cdot}10^{-4}$ as CLR has been chosen as a reference because it allows an adequate range of the ratio of the cells lost against the emitted cells (see details in section 3.3.2). The utilisation varies from 5% (high burstiness) to 40% (moderate burstiness), this range covers the majority of the traffic classes used for other well known experiences (see Chapter 5). The mean burst length is chosen to analyse extreme situations. For example 500 cells correspond to a 2 kbits burst length which is an exaggerated case for real-time traffic. The buffer size changes from the minimum number of cells (1 cell) where only one cell is saved, up to 128 cells which corresponds to a delay equal to 140 μs when using a link of 150 Mbit/s. Finally, the capacity of the link is determined by a given CLR and utilisation, the presented experiments constrain the values of **C** from 30 to 70 Mbit/s.

The following table summarises a set of simulation parameters and its corresponding range of values.

Parameter	*Units*	*Value/Range*	*Reason*
Number of sources	none	40	Good Statistical Multiplexing Gain
CLR	cells lost / cells sent	5 E-4.	Trade-off between realistic CLR (smaller) and feasible simulation
Utilisation	percentage	(5,40)	High Burstiness (Real-time services)
Burst-Length	cells	(20, 500)	From a minimum information block (about 1 Kbyte) to a maximum length
Buffer-size	cells	(1, 128)	(1, Max. Admissible delay)
Capacity of the Link.	Mbit/s	(30, 70)	Depending on (CLR and utilisation)

Table 3-4 Experiments: Environment and parameter values justification

Cell level queuing models

Analysis of the queuing process is a fundamental part of performance evaluation, because queues form in telecommunications systems whenever customers contend for limited resources. In ATM, customers can be cells, bursts, or connections. They arrive at a queuing system; they wait in a storage area, if service is not immediately available.

Queuing systems are widely studied in a general manner. The classical book [KLE75], discusses general queuing theory. More specifically, books which are related to telecommunications, also include queuing systems theory applied to the behaviour at cell level ([MCD94], [PIT96], [SAI91] and [SAI92] are good examples). A great number of papers related with bandwidth allocation and CAC schemes use the queuing theory to solve CLP and delay in their respective environments.

The first assumption is that the cell arrival pattern to the output buffer in an ATM link can be approximated by negative exponential inter-arrival times. This is the same as saying that the arrivals are described by a Poisson process, which counts the number of arrivals in a time interval.

The distribution of arrivals per slot (corresponding with the time to service one cell) by a Poisson process is:

$$\Pr\{\text{k arrivals in a cell slot}\} = a(k) = \frac{\lambda^k}{k!} e^{-\lambda} \quad (14)$$

λ	arrival rate in cells
k	number of arrival cells

The mean of the arrival distribution a(k) is:

$$E(a) = \rho \quad (15)$$

ρ	utilisation (number of carried cells per time slot)

The state probability, i.e. the probability of being in state k, is defined as:

$$s(k) = \Pr\{\text{there are k cells in the buffer in a slot}\} \quad (16)$$

For the M/M/1 system the service time is described by a negative exponential distribution. By considering a fixed service time (in accord with cell size) the M/D/1 queue has to be considered.

The analysis for infinite queues is widely introduced, [KLE75], [PIT96] and [MCD94] are examples. The evaluation of s(k) is based on the exact value of s(0):

$$s(0) = 1 - E(a) \quad (17)$$

With an infinite buffer, there will no cell loss. In the following paragraphs the effect of a finite buffer is considered. Knowledge of how the CLP varies with the buffer capacity **X** is needed.

With finite buffers there will be lost traffic; therefore, the evaluation of s(0) is unknown. Temporarily ignoring this fact, a new variable u(k) is defined:

$$u(k) = \frac{s(k)}{s(0)} \quad (18)$$

so

$$u(0) = 1 \quad (19)$$

Then

$$u(1) = \frac{1 - a(0)}{a(0)} \quad (20)$$

$$u(k) = \frac{u(k-1) - a(k-1) - \sum_{i=1}^{k-1} u(i)a(k-i)}{a(0)} \quad (21)$$

and all the values of u(k) , 0 <= k <= X, can be evaluated.. The expression for s(0) can be evaluated as:

$$s(0) = \frac{1}{\sum_{i=0}^{X} u(i)} \qquad (22)$$

Finally, by using previous equations corresponding with u(j) and E(a) the cell loss probability is just the ratio of lost traffic to offered traffic:

$$CLP = \frac{E(a) - (1 - s(0))}{E(a)} \qquad (23)$$

The following figure plots the Probability of Overflow of the output link at cell level that corresponds to the evaluated CLP for the different link load used in the simulations (the corresponding value for λ has been applied).

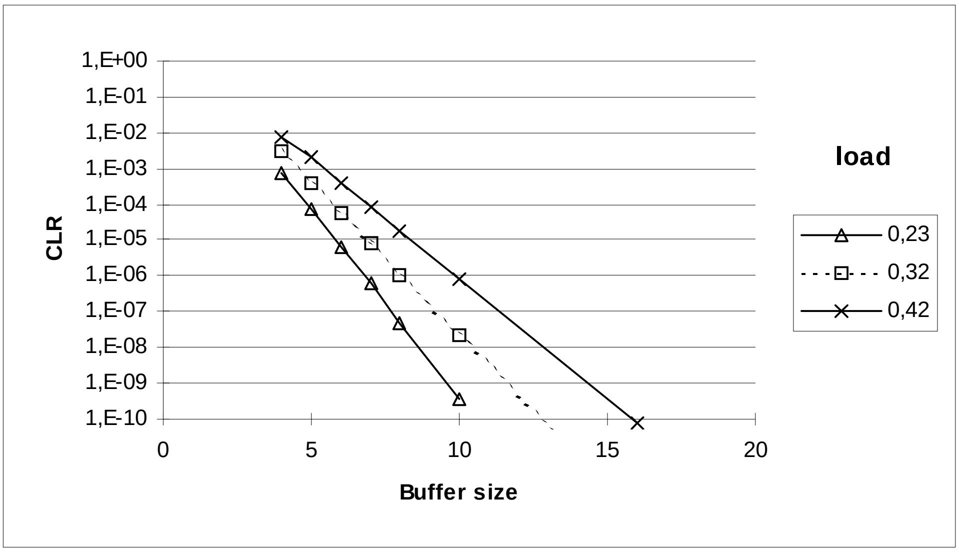

Fig. 3-8 Cell Loss Ratio of the output link at cell level based on M/D/1/K queue system..

Minimum Buffer size considerations

The following three experiments are presented with the buffer size as the x-axis, different curves plot Cell Loss Ratio corresponding to the mean burst length pre-set values. The capacity of the link varies from 70 Mbit/s to 38 Mbit/s to hold the CLR about 5 10^{-4} , as shown in Table 3-4.

Experiment	*Source Utilisation*	*Link Capacity*	*Link Load*
1	5 %	70 Mbit/s	23 %
2	10 %	50 Mbit/s	32 %
3	20 %	38 Mbit/s	42 %

Table 3-5 Definition of the experiments

Results:

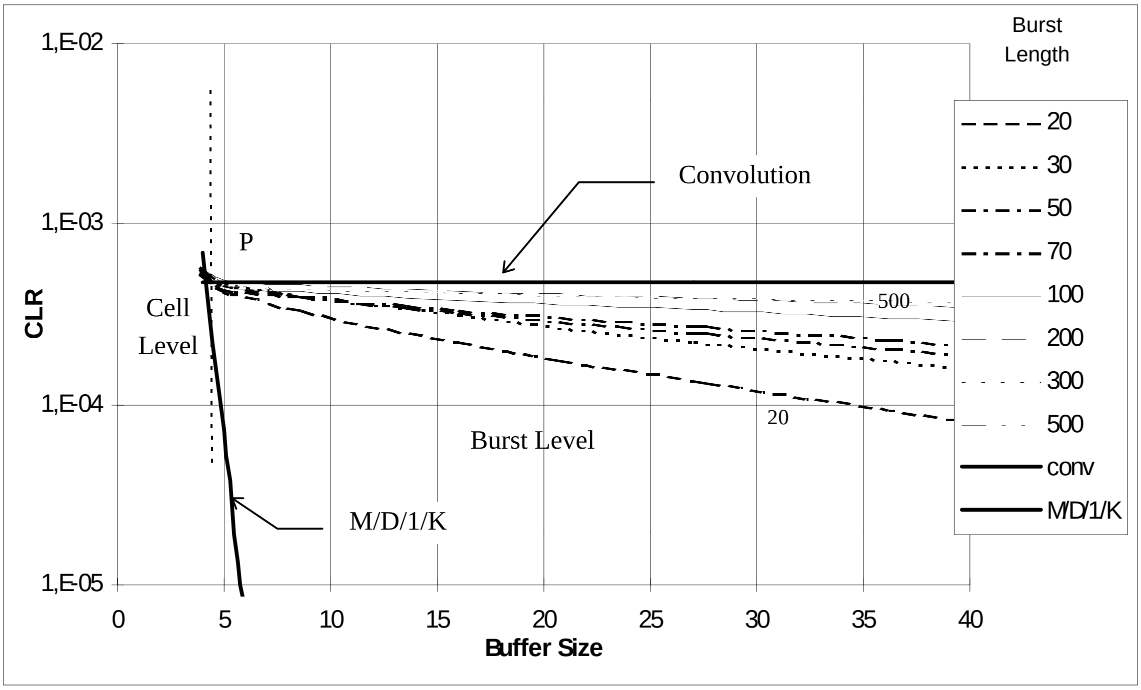

Fig. 3-9 Experiment 1: CLR against the buffer size for different mean burst length.

As we expect, this sort of graphics show clearly the behaviour of the buffer occupation at cell-level (left-side) and burst level (right-side). General results have been obtained for a wide range of values for the buffer size and for the burst length. Note that the behaviour at cell-level does not depend on the mean burst length.

The lower limit to dimension the buffer size corresponds to point P (see Fig 4, Fig 5 and Fig 6) where all curves plot the same CLR. Therefore, the corresponding minimum buffer size is clearly fixed by the x-coordinate of point P. On the presented experiments the minimum buffer size is 4 for a 5% of utilisation, 5 for a 10% of utilisation and 7 for a 20% of utilisation. Buffer size for cell-level contention is rather small.

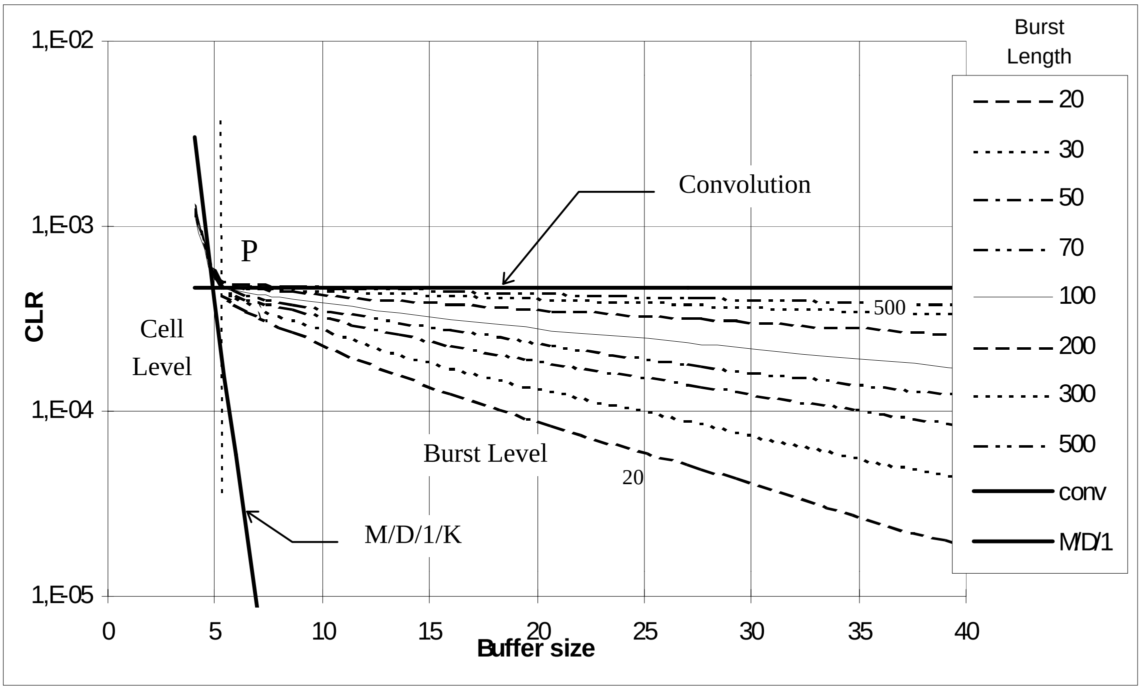

Fig. 3-10 Experiment 2: CLR against the buffer size for different mean burst length.

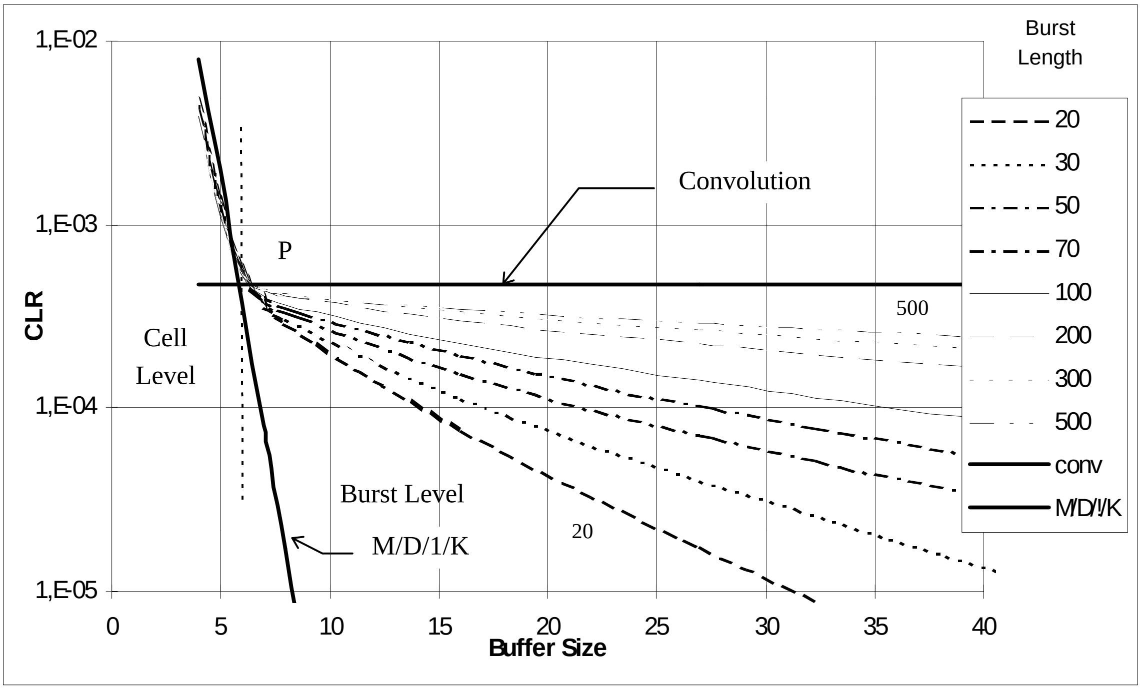

Fig. 3-11 CLR against the buffer size for different mean burst length.

Approximate analysis

An approximate analysis of the M/D/1 system produces the following equation:

$$CLP(x) \approx Q(x) = e^{-2x\left(\frac{1-\rho}{\rho}\right)} \quad (24)$$

This can be rearranged to give

$$x = -\frac{1}{2}\ln(CLP)\left(\frac{\rho}{1-\rho}\right) \quad (25)$$

For these equations to be accurate, the utilisation must be high. Therefore, this approximation is not sufficient accurate for the presented environment.

By using the equation for the exact evaluation of the CLR the minimum buffer size for a given CLR can be obtained. Set the CLR equal to $5{\cdot}10^{-4}$, for ρ = {0.23, 0.32, 0.42}, the corresponding buffer size B is {4, 5, 6}, that as we expect is a upper bound for the buffer size to achieve adequate cell contention. As shown in Fig. 3-9, Fig. 3-10 and Fig. 3-11 the accuracy of this approximation against simulation depends on the link utilisation.

In general, the expression (23) can be viewed in the following figure:

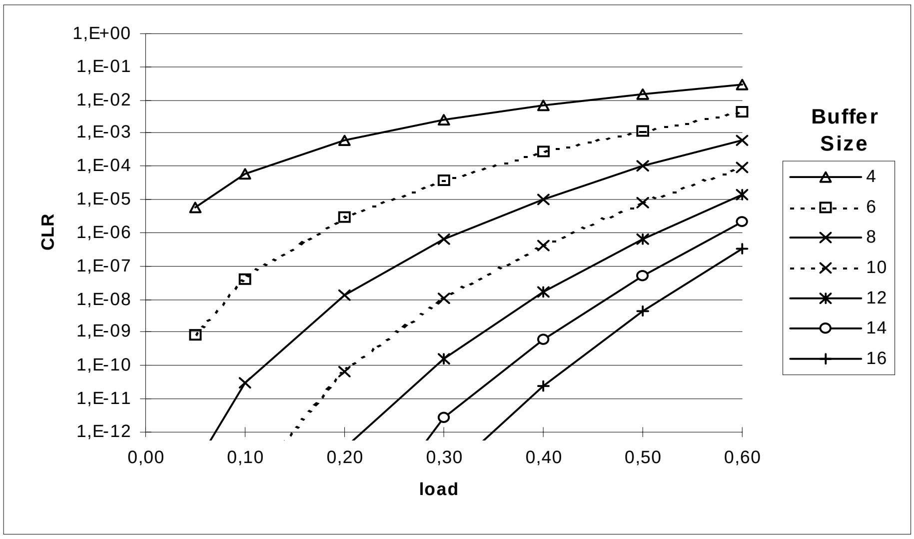

Fig. 3-12 CLR vs. Load varying Buffer size.

The previous figure can be viewed in a alternative manner. For a given buffer size and a CLP objective, a maximum admissible load is shown by the Fig. 3-13. For example, if the buffer of the output link is equal to 12 cells, and the desired QOS is a maximum cell loss probability less than $1{\cdot}10^{-6}$ the maximum admissible load is about 0.52.

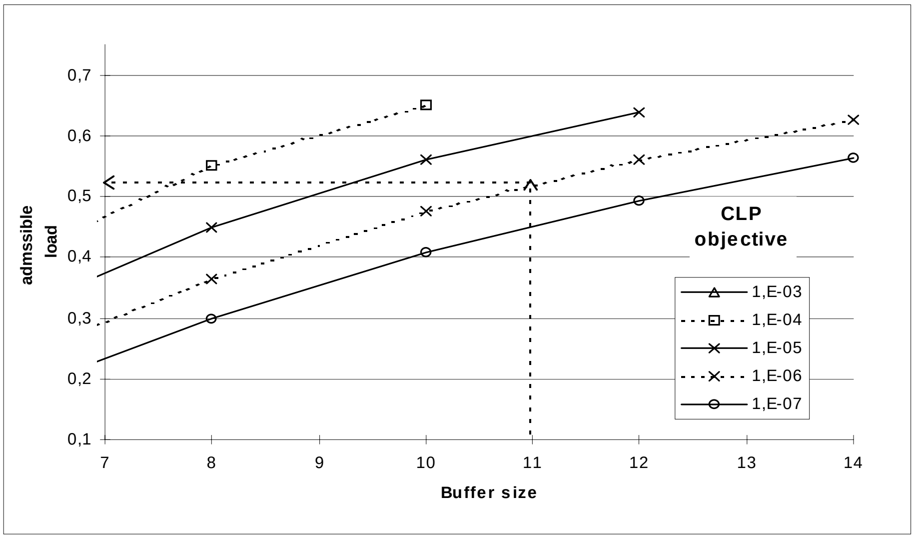

Fig. 3-13 Buffer size against load, varying the CLR objective

Maximum Buffer size considerations

This experiment concerns a study about the accuracy of the method when large buffer versus the mean burst length is considered, which is a exaggerate situation for the buffer-less assumption. The experiment parameters are showed on the table below.

Parameter	*Value*
Capacity of the link	50 Mbit/s
Connection Peak Rate	4 Mbit/s.
Connection Mean Rate	0.4 Mbit/s
Number of connections	40

Table 3-6 Experiment Parameters

A ratio of peak rate/link capacity equal to 4/50 = 0.08 is chosen (in order to achieve a reasonable multiplexing gain). As in experiment 2, a number of 40 connections emitting cells in a link is analysed.

The following figure is an example that shows the behaviour of the cell stream in the output buffer in a ATM switch. This aspect is illustrated in Fig. 3-14 which plots the Cell-Loss Ratio against mean burst length when varying the buffer size.

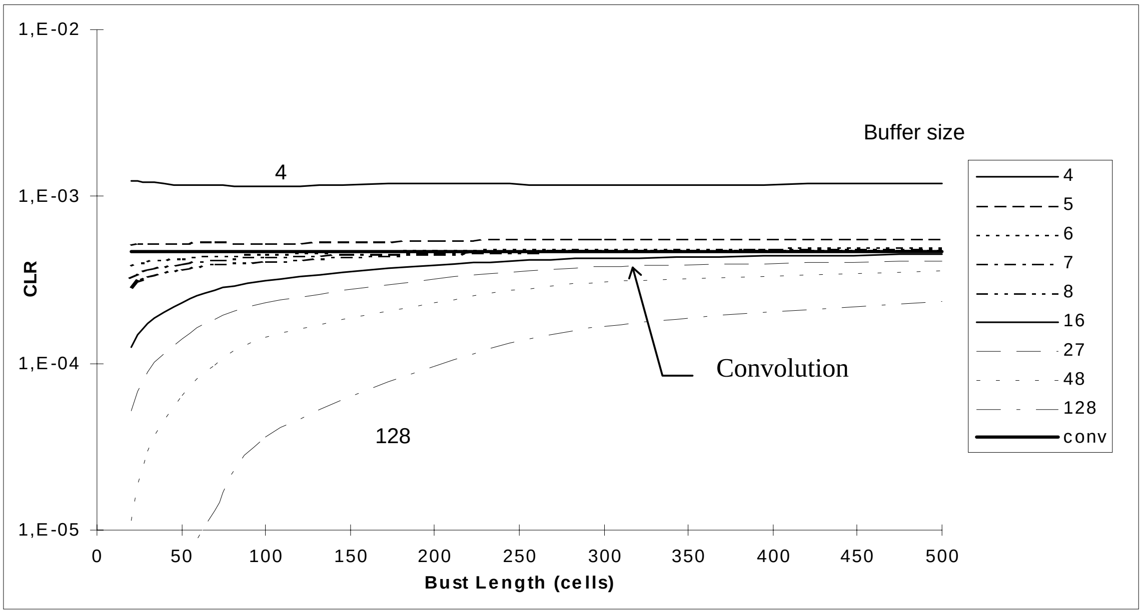

Fig. 3-14 CLR vs. Burst Length

As we expect the CLR evaluated by the convolution approach is an upper limit for all curves corresponding to buffer size greater than 5 cells. From this result we can conclude that, in this scenario, the minimum buffer size for cell-level contention is equal to 6.

A more accurate study of the above figure provides further conclusions about the relationship between the (mean burst length/buffer size) ratio and the CLR predictions based on convolution. By combining these results it is possible determine the buffer size or the mean burst length for a given environment.

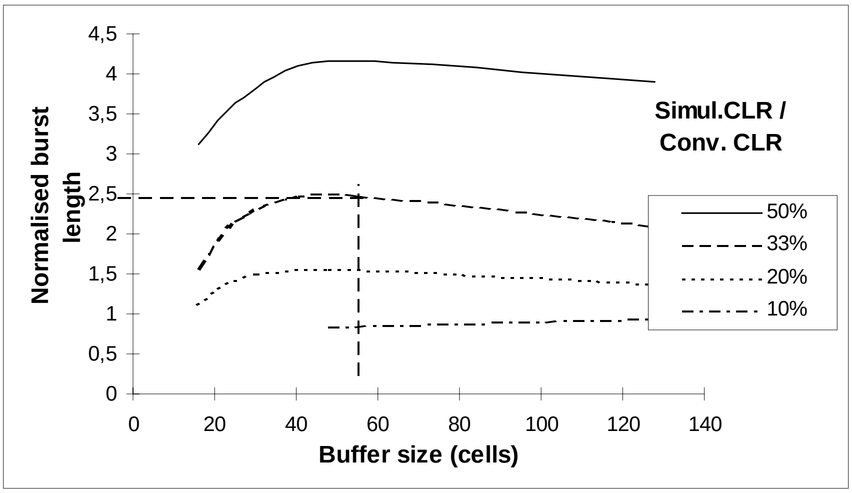

Fig. 3-15 Burst Length/Buffer size Ratio vs. Buffer size varying load.

Fig. 3-15 shows the normalised[17] mean burst length against buffer size. For a mean burst length of about 3 to 4 times larger than the buffer size, the CLR obtained by simulation decreases to half the CLR obtained by convolution (solid line in Fig. 3-15), and for a mean burst length of about 2 to 2.5 times larger than the buffer size, it decreases to a third.

For a given buffer size (60 cells in the example), an accuracy equal to 33 % (which means that the CLR obtained by simulation is 33% of the convolution prediction) is achieved when the mean burst length is about 2.5 (that means a mean burst length equal to $2.5 \cdot 60 =$ 150 cells). A similar method can be applied to other buffer sizes and accuracy goals.

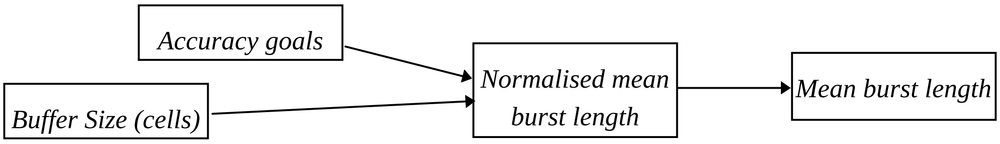

Fig. 3-16 Mean burst length determination

A second possibility is to provide an adequate buffer size by knowing the mean burst length of the traffic for a given scenario. The same results as before can be used, but the buffer size amount against burst length is plotted, see Fig. 3-17.

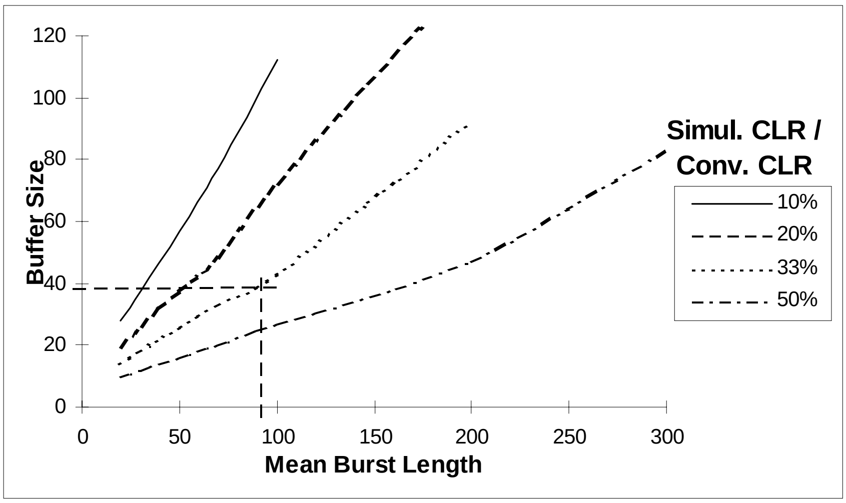

Fig. 3-17 Buffer size vs. Mean Burst Length

In the above figure, we can examine the necessary buffer size for a mean burst length by choosing the desired accuracy for CLR. For example, for bursts of about 100 cells a buffer able to store 40 cells is needed if the actual CLR corresponds to 33% of the evaluated CLR. Note the linearity of the different curves.

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[17] Mean burst length / buffer size

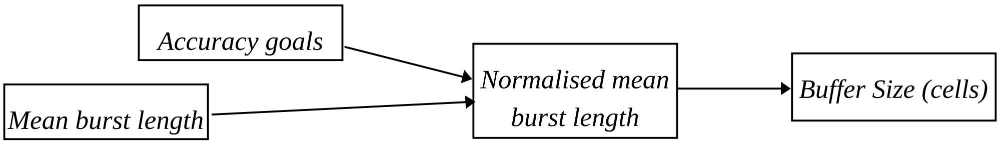

Fig. 3-18 Buffer size determination.

There is no point in evaluating a maximum buffer size, the evaluation method should be more or less conservative corresponding to the buffer size chosen. Therefore, the maximum buffer size depends on the accuracy desired. Given a traffic descriptor and the desired accuracy, the minimum and maximum buffer size can be determined.

Fluid-flow model vs. stationary model

Finally, in order to determine the accuracy of the convolution approach a brief comparison between a stationary method and a Fluid-Flow (FF) model is shown. The major objective is to establish, whenever the convolution is accurate enough when approaching the worst case situation. This is done by increasing the buffer size in comparison to the mean burst length and source utilisation.

The same scenario, used in the previous section, is analysed. The metric for this section is the demanded bandwidth, which is evaluated by fixing the CLR used in the previous experiments (about 0.5 E-4, as shown in Table 3-4).

	Source Utilisation (%)			
Method	*5*	*10*	*20*	*40*
Fluid-Flow	400	200	100	45
Convolution	70	50	38	30

Table 3-7 Demanded Bandwidth (Mbit/s)

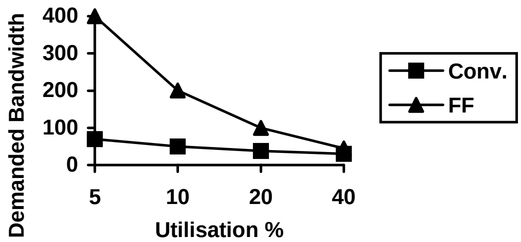

Fig. 3-19 Demanded bandwidth against. utilisation (Fluid Flow versus Convolution)

Fig. 3-20 shows the Fluid-Flow model results for different burst lengths when varying the buffer size. As the figure shows, there are some variations in the Demanded Bandwidth when burst length is small. The Demanded Bandwidth remains constant for large burst length. Moreover, these results show that the Demanded Bandwidth is independent of the buffer size.

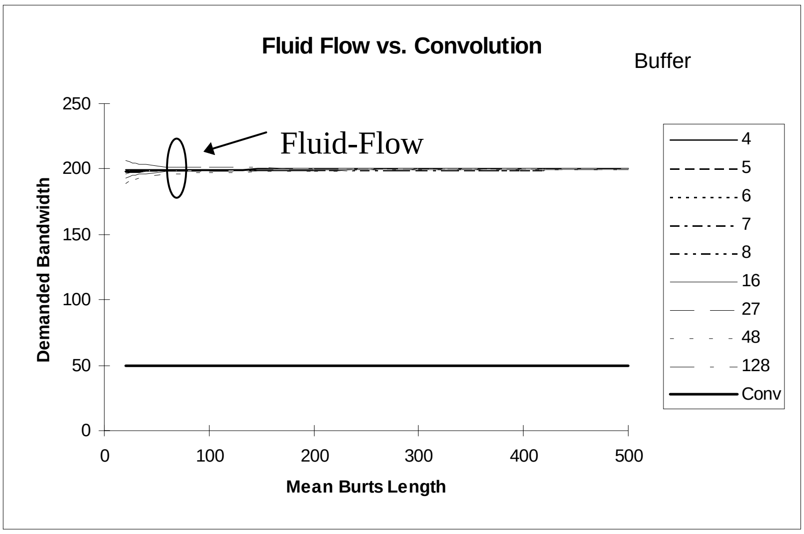

Fig. 3-20 Demanded Bandwidth vs. Mean Burst Length (source utilisation = 10 %).

The utilisation applied to obtain Fig. 3-20 is equal to 10 %, (similar results have been obtained for the other values for utilisation but have been omitted for simplicity). When the utilisation decreases the difference for small burst length becomes negligible, so these figures are omitted. We can conclude that the mean burst length is not relevant for the presented scenarios.

3.4 SUMMARY

In this Chapter the utilisation of the Probability of Congestion as bandwidth decision parameter has been presented. The validity of using PC is compared in situations when only the CLR parameter is relevant, i.e. QOS parameters for a bufferless scenarios

It can be summarised that the convolution algorithm seems to be a good solution for CAC in ATM networks with relatively small buffers. If the source characteristics are known actual cell loss ratio can be very well estimated. Furthermore, this estimate is always conservative allowing to keep the network performance guarantees [R2061/8].

The PC is evaluated by convolution and a upper limit for CLR is obtained. This evaluation ignores both buffer size and burst length. Therefore, several experiments have been carried out and investigated to explain the deviation between the proposed method and the simulation[18]. Time parameters for burst length and different buffer sizes have been considered.

Experiments to confine the limits of the burst length with respect to the buffer size conclude that a minimum buffer size is necessary to achieve adequate cell contention. Normally, small values for the buffer size are involved (4 to 7 in the presented scenario). On the other hand, by increasing the buffer size the CLR, as shown by analysis becomes too conservative. Then results are obtained by analysis become inaccurate, as the mean burst length approaches the buffer size.

Finally, a comparison with non-stationary methods is presented. It is seen that convolution approach obtains a more accurate CLR result in the presence of bursty traffic and by using relatively small buffers.

[18] Results which are obtained by real measurements given by an ATM network are also presented in chapter 5.

4. Bandwidth Allocation based on the Probability of Congestion

In this section, different methods to obtain Probability of Congestion (PC) on a link based on the convolution function are presented. The cost involved in the evaluation of the PC is also analysed. The study focuses on the stationary model. In this case the effect of statistical multiplexing is the dominant factor and cells are considered lost when the instantaneous rate is greater than the bandwidth provided by the link.

4.1 WORKING HYPOTHESIS

This chapter has been carried out under the following assumptions:

1. Small buffer size. We assume that the buffer length is assumed to be small in order to guarantee an acceptable maximum delay. The size of the buffer is not studied; it will be dimensioned by imposing the requirement that the cell loss probability due to contention at cell level be negligible. If the burst length is greater than the buffer length, several cells of the burst are lost, so only the burst level statistics are of interest for the traffic control issues [MIT94].

2. Statistical multiplexing. The Statistical multiplexing gain is a dominant factor. The activity of established connections is assumed to be statistically independent.

3. Traffic descriptors. This work uses the source characterisation based on the GMDP model[19]. The parameters used are rate and source state probability. In ON-OFF traffic models the parameters can easily be obtained from mean and peak rate (see section 3.2.6)

4. Stationary method: Convolution algorithm. In line with previous assumptions the stationary model will be used in order to evaluate the bandwidth offered by all connections [GUE91] and [OHT92]. More specifically, the convolution approach is chosen to compute the PC. This method involves computation complexity which makes real-time responses difficult. In order to reduce this high cost, new methods for evaluating the PC are presented.

5. Heterogeneous Traffic: Individual Cell Loss. ATM must provide, proper QOS performance for different service classes. QOS parameters include cell loss probability (CLR), maximum cell delay, maximum cell delay variation, etc. Different QOS performance can be achieved using priority levels for the cells of each service class by introducing explicit indicators, e.g. cell loss priority bit [MIT90], [KRO91],

[19] The GMDP model is described later in section 5.3.1

[KRO92] and [DEC92]. Multiple priority levels implicit indicators, e.g. virtual path indicator, VPI, can be used, but this requires a more complex congestion control structure (buffer allocation, scheduling and selective discarding mechanisms). These aspects are not considered in this work, although in compensation an accurate analysis of the individual CLR for heterogeneous traffic is presented.

4.2 CONVOLUTION APPROACH

This section concerns the evaluation of the basic Convolution Approach, the calculation algorithm is detailed pointing out the problems found.

The Convolution Approach evaluates the statistical distribution of the instantaneous offered rate **Y** by all established connections on a link. This distribution allows the evaluation of the Probability of Congestion (PC) on the link. The PC is easily evaluated by summing all terms (that is probabilities) corresponding to rates greater than the capacity of the link.

4.2.1 Analysis of the Convolution Approach

This section contains the calculation of the bandwidth requirements of the superposition of several sources. This approach is based on the well known expression of the convolution procedure denoted by:

$$Q = Y * X \qquad (26)$$

which is evaluated by the following expression:

$$P(Y + X = b) = \sum_{k=0}^{b} P(Y = b - k) P(X = k) \qquad (27)$$

where **Q** is the bandwidth requirement of all established connections including the new connection; **Y** is the bandwidth requirement of the already established connections; **X** is the bandwidth requirement of a new connection, and **b** denotes the instantaneous required bandwidth. In fact, the convolution approach obtains a probability density function for the offered link load, expressed as the probability that all traffic sources together are emitting at a given rate **b**. We take into account that the evaluated offered load is not the link load itself, but the load generated by all the traffic sources intended to be carried by the link. The link carries this load in non-congestion state only[20].

[20] See Figures 2.1 and 2.2.

4.2.2 Calculation algorithm

The direct application of the expression (26) in order to evaluate the convolution is difficult in practice. In this section, a detailed algorithm is explained. For an effective evaluation two data structures are necessary. Each source has an associated Source Status Vector (Source-SV); this vector has two fields for each element, rate and its associated probability, for each possible state.

State	*Rate*	*Probability*
0	r_0	p_0
1	r_1	p_1
...	...	...
k	r_k	p_k
...	...	...
M-1	R_{M-1}	p_{M-1}
		$\Sigma p_k = 1$

Table 4-1 Source Status Vector

M Number of possible states.

The corresponding probabilistic scale is shown in the following figure:

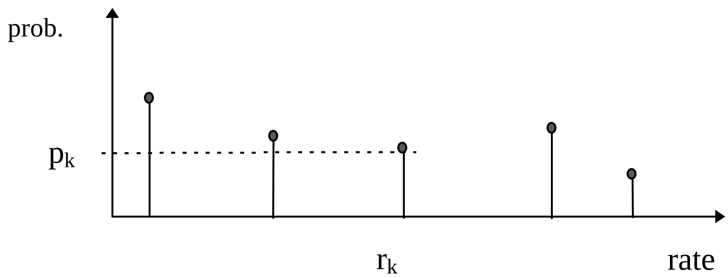

Fig. 4-1 Distribution of rates.

For On/Off sources, it is simple to see that the maximum number of states **M** is equal to 2, with the corresponding values for r_0 (off state) and r_1 (on state) being 0 and the peak rate respectively. To store the transmission rates possible for all the connections established at a given moment and to store the probability that the sources will emit at those rates, a System Status Vector (System-SV) is needed. Each element of this vector has two fields, rate and probability, as the Source-SV (see also Table 4-1 Source Status Vector).

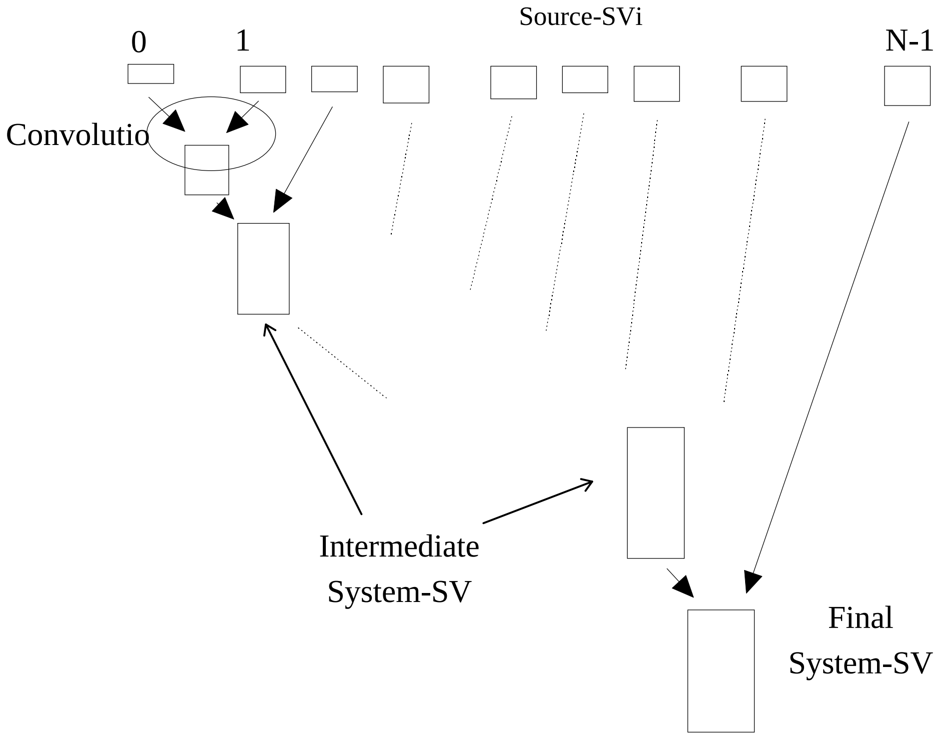

Fig. 4-2 Evaluation scheme

In order to obtain the probabilistic distribution for the System-SV, the following process is carried out: whenever a new connection demand arrives the System-SV must be updated ; the corresponding Source-SV is used to do this update, and for each old System-SV element a set of new System-SV elements is generated. The number of these elements is determined by the number of states for the new connection. The *rate* of each new element is the sum of the existing rate and the rate corresponding to the state of the new source. The *probability* of each new element is the product of the existing probability and the new probability corresponding to the state of the new source[21]. By using this method N-1 convolutions are needed for each new connection.

The expression (25) can be re-written as [IVE90]:

$$Q_n = Q_{n-1} * X_n \; ; \; n \in \{1,2, ..., N-1) \qquad (28)$$

Considering $Q_0 = X_0$. Clearly, we should carry out N-1 convolutions to obtain the global distribution.

4.2.3 De-convolution issues

At any point in time, an ATM link can carry several thousand connections. As described in a previous section, when a connection is accepted, the new connection is convoluted with the global steady-state probabilities of all existing connections. When a connection terminates it would be preferable to deconvolute all existing connections. So the

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[21] The details of this mechanism can be viewed on the example in Appendix 3-A.

feasibility of the deconvolution operation is very important. The problem is that the global steady-state has now changed; this means that some previously calculated values are lost. The reason for this is that a) the accumulation of probabilities corresponding to the same rate and b) very small intermediate values are not considered [IVE90].

From a numerical point of view, the accuracy of the evaluation obtained by repeatedly convoluting and deconvoluting means we lose accuracy. Furthermore, by not truncating the state, the space required for storage increases; the number of arithmetic operations further increases. So, in general, it is not possible to implement the deconvolution [IVE90].

These aspects, relating to accuracy and cost, are more widely developed in Chapter 5.

4.2.4 Cost Issues

Some considerations about storage requirements and calculation cost are presented in order to analyse the complexity of the method. Sources are grouped in types. Let **L** represent source types, which are classified by a mark ranging from 0 to L-1. All connections within the same type have identical traffic parameters and identical QOS requirements.

Storage requirements

The storage requirement cost is independent of the above implementation and it is calculated as follows: let **L** equal the number of types of sources, let n_j equal the number of j-type connections; and S_j be the number of states of each j-type source. The number of connections of all types is **N**, which can be evaluated by the following expression:

$$N = \sum_{j=0}^{L-1} nj \qquad (29)$$

The number of elements **M** necessary to store the System-SV is [22]:

$$M_N = \prod_{j=0}^{L-1} (S_j)^{n_j} \qquad (30)$$

The same rate may appear more than once in System-SV. The size of the System-SV may be reduced by sorting and combining the repeated rate values. This is carried out by accumulation of probabilities of the elements with identical rate; the amount of

[22] In [LEE94b p 81] some simplifications can be applied whenever Sj = 2 ∀ j, (i.e. only On-Off sources).

reduction which can be achieved depends on the rates $r_{j,i}$ (rate of a j-type source emitting in i-state) of the source types and the number of states S_j.

The following expression gives the number of elements necessary to store the System-SV if only one type of source j (homogeneous traffic, L=1) with three states (S=3) is assumed. Therefore, the System-SV is reduced :

$$M_n = \begin{cases} \sum_{k=1}^{n+1} k = \frac{(n+2)(n+1)}{2} & If \ n \in [1, K-1] \\ M_{K-1} + K(n-K+1) & If \ n \in [K, \infty] \end{cases} \quad (31)$$

where

$$K = \frac{r_2 - r_0}{h.c.d(r_2 - r_0, r_2 - r_1, r_1 - r_0)} . \quad (32)$$

h.c.d. Highest Common Denominator

Calculation cost

The evaluation cost is proportional to the size **M** of the vector. **M** products, **M** additions and the sorting of **M** elements are necessary. Additional cost to compact elements with identical rate is needed, and the probability of those elements occuring must be added.

First Improvements

There are two possibilities for improving the implementation of the above algorithm:

- With storage: To overcome the computing time problem, one possibility is to store the bandwidth distribution of the total traffic stream. In this case only one convolution for a connection set-up is needed. One de-convolution for a connection release is needed [KAL92] and [D122].

- Without storage: The distribution is not stored. For a connection set-up many convolutions (equal to the number of existing connections) have to be calculated but no de-convolution for connection release is needed [D122].

Some approximations to the bandwidth requirements have been investigated in [KLE91] and [JOO89] to overcome the above limitations. In [IVE87] the state space is truncated to a bound N and the number of arithmetic operations is reduced, all probabilities less than N are ignored. In [IVE90] an overview of the performance of three algorithms for the evaluation of the total load is presented. In [KAS89] the normal approximation is used, except where the number of calls per traffic types is too small; in this case the convolution approach is used.

Drawbacks

In [JOO89], [IVE90], [KRO90], [IVE91], [KAL92], [R1022], [D122], [MIA94] and [RAM94] some limits of the Convolution Approach are pointed out:

- High cost in terms of storage requirements. The size of the required storage presents a big problem. Note That a huge amount of memory storage M is required by the System-SV. This requirement increases dramatically with the number of connections N and source states S_j.

- High cost in terms of calculation. The computing time depends on the complexity of the distribution itself. The time needed for the convolution increases with the number of states per connection. In any implementation without storage this time is greater because many convolutions have to be performed.

- Accumulated calculations. Since the probability expressed in the System-SV is the result of a large number of previous calculations, results may be inaccurate after a great number of evaluations. Due to rounding errors. An alternative is to update all the System-SV from scratch periodically.

- De-convolution. When the implementation is carried out without storage and there is a disconnection, no de-convolution is feasible, because computing the System-SV from scratch is necessary. When the bandwidth distribution is implemented with storage a de-convolution is not easy (and often impossible) [WIL90] and [IVE90] because of accumulated calculations rounding errors.

- Individual QOS are not considered. The evaluated link status using a convolution approach makes no distinction between individual connection. Thus, the individual QOS for each type of source is not available.

4.3 THE EXTENDED CONVOLUTION APPROACH

4.3.1 Motivation

To overcome the above drawbacks associated with the PC calculation, a new method of evaluation is analysed. The multi-nomial distribution function is applied to groups of the same type of sources, and the global state probabilities are evaluated by convolution of the partial results obtained from the different existing groups of sources [FAB95] and [MAR93].

4.3.2 Overview of the method

The state of the link can be expressed as a function of the number of active connections of each service type (n_0, n_1, ... n_j , ... , n_{L-1}). This is because the state of the link depends only upon a service’s occupancy.

The figure below, shows an overview of the method. First the multinomial function is applied to homogenous sources producing intermediate results. Finally, from these intermediate results a final result is obtained by convoluting one element of a given class of traffic with one from each of the other classes. This process is called *multi-convolution* in this study.

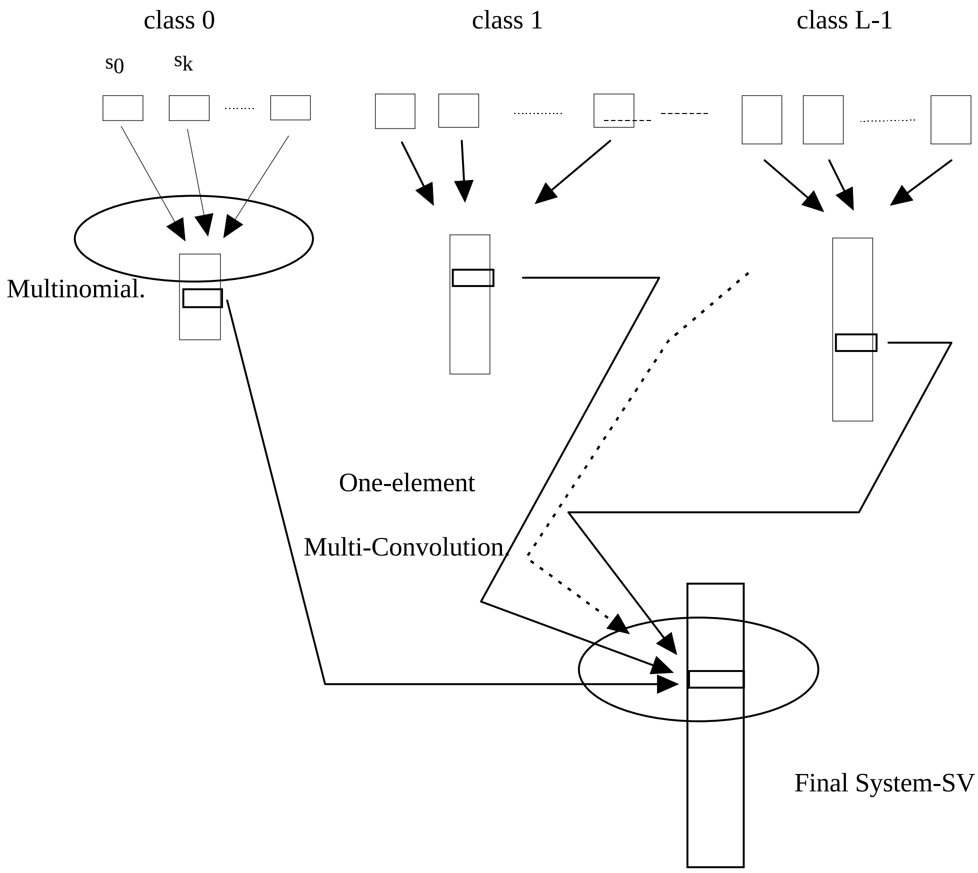


Fig. 4-3 Overview of the method.

The following sections describe both phases: Multinomial evaluation and Multi-convolution procedure.

4.3.3 The Multinomial Distribution Function

A faster method for overcoming the drawbacks of the Convolution Approach has been studied. As seen in the previous section, after computing the convolution the same rate may appear more than once in the System-SV. Which elements are repeated? How many times?

The Multinomial Distribution Function (MDF) will be studied to evaluate directly the number of those repetitions [ASH69], [RAD73], [HAG89] and [MEN95].

4.3.4 Calculation Algorithm

Homogeneous traffic case

In this section, only one type of source is assumed, emitting in **T** states. Each state-i has an associated rate r_i and probability p_i. Therefore, for **N** connections n_0 sources are in state s_0; n_1 sources are in state S_1; and n_{T-1} sources are in state s_{T-1}.

Consider a **S** dimensional random variable denoted by

$$(s_0, s_1, ..., s_{T-1}) \tag{33}$$

Therefore, a random event that has been repeated **N** times (considering all the connections at a instant) has the characteristic:

$$(n_0, n_1, ..., n_{T-1}) \tag{34}$$

It is necessary to calculate the probability of s_0 occurring n_0 times, s_1 occurring n_1 times , and s_{T-1} occurring n_{T-1} times. For this purpose, now consider generalised Bernoulli trials. As in the previous situation, the point:

$$(s_0, s_0, ..., s_0, \quad ,s_1, s_1, ..., s_1, ..., s_{T-1}, s_{T-1}, ..., s_{T-1}) \tag{35}$$

with $(n_0, n_1, ..., n_{T-1})$ connections is assigned the probability :

$$p_0^{n_0} \cdot p_1^{n_1} \cdots p_{S-1}^{n_{S-1}} \tag{36}$$

Where $p_0+p_1+...+p_{S-1}= 1$.

The above expression is the probability assigned to any specific sequence having n_i occurrences of S_i, varying i = 0, 1, ,..., n_{j-1}. Thus, the number of sequences having exactly n_0 connections in state s_0, n_1 connections in state s_1,.... and n_{T-1} connections in state s_{T-1} is

$$\binom{N}{n_0}\binom{N - n_0}{n_1}\cdots\binom{n_{S-1}}{n_{S-1}} =$$

$$= \frac{N!}{(N - n_0)!\, n_0!}\ \frac{(N - n_0)!}{(N - n_0 - n_1)!\, n_1!} \cdots \frac{(N - n_0 -...-n_{S-2})!}{(N - n_1 -...-n_{S-1})! n_{S-1}!} =$$

$$= \frac{N!}{n_0!\, n_1! \ldots n_{S-1}!} \tag{37}$$

Finally, the probability of all sequences that have this characteristic is:

P(state s_0 occurs n_0 times, ... , state s_{T-1} occurs n_{T-1} times) =

$$= \frac{N!}{n_0!n_1!\dots n_{T-1}!} \quad p_0^{\,n_0} \cdot p_1^{\,n_1} \cdots p_{T-1}^{\,n_{T-1}} \qquad (38)$$

where n_0, n_1, ..., n_{T-1} are non-negative integers whose sum is N.

This is probability corresponding to the Multinomial Distribution Function (MDF) [HAG89], [MEN95] and [RAD73]. Note that the probability of each source being in state s_i is independent of the probability of the other source states.

Some data structures are necessary to evaluate the Enhanced Convolution Approach (ECA). For N connections of the same type, there is an associated Sub-Matrix (SMX).

$$SMX(N) = \begin{bmatrix} n_{0,0} & n_{0,1} & n_{0,j} & n_{0,T-1} \\ \cdots & \cdots & \cdots & \cdots \\ n_{r,0} & n_{r,1} & n_{r,j} & n_{r,T-1} \\ n_{M-1,0} & n_{M-1,1} & n_{M-1,j} & n_{M-1,T-1} \end{bmatrix} \qquad (39)$$

SMX_r is the generic row of SMX. The number of columns in this element is equal to the number of source rates T

$$SMX(N) = \begin{bmatrix} SMX_0 \\ \cdots \\ SMX_r \\ \cdots \\ SMX_{M-1} \end{bmatrix} ; \qquad SMX_r = < n_0, n_1, \dots, n_{T-1} > \qquad (40)$$

The number of rows M is determined by formulae (30). SMX stores the distribution of the connections from each state. The system load density function is obtained directly from the sub-matrix using the MDF expression (37).

Heterogeneous traffic case

When there are different types of sources j (heterogeneous traffic), it is necessary to ‘convolute’ between all source types. To store all possible combinations relating to the system state, a System Status Matrix (SSM) is defined. The generic elements of the SSM, namely the general system status rows SSM_r, are generated each by concatenating all possible combinations between the different sub-matrices rows SMX_r, associated with the L different j-types of sources (j = 0, 1, ..., L-1) :

$$SSM_r =< SMX_{r_0,o}, \dots, SMX_{r_{L-1},L-1} > \quad \forall\ r=0,\dots, M_{j-1} \qquad (41)$$

and from (39)

$$SSM_r =< n_{r_0,o},\ldots,n_{r_0,S_j-1},\ldots,n_{r_{L-1},L-1},\ldots,n_{r_{L-1},L_j-1} > \qquad (42)$$

Individual CLR evaluation

In this section, the role that the Enhanced Convolution Approach (ECA) plays in computing the individual cell loss probabilities. Based on the ECA algorithm, grouping connection in types, the following expression for the cell loss probability of the type-j traffic proposed:

$$CLR_j = \frac{\sum_{W>C} \frac{W_j}{W}(W-C)P(Y=W)}{E(Y_j)} \qquad (43)$$

Wj is the rate offered by all type-j traffic when the instantaneous offered rate on the link is L and $E(Y_j)$ is the mean rate of all traffic of type-j. Both terms are easily obtained during the evaluation of PC based on the ECA algorithm. To demonstrate formula (43) it is necessary to sum the individual CLR_j for all connections, so CLR can be evaluated by:

$$CLR = \frac{\sum_{\forall j} E(Y_j) CLP_j}{E(Y)} \qquad (44)$$

Inserting the expression for CLRj we get

$$CLR = \frac{1}{E(W)} \sum_{\forall j} E(Y_j) \quad \frac{\sum_{W>C} \frac{W_j}{W}(W-C)P(Y=W)}{E(Y_j)} \qquad (45)$$

$$CLR = \frac{1}{E(Y)} \sum_{\forall j} \sum_{W>C} \frac{W_j}{W}(W-C)P(Y=W) \qquad (46)$$

Interchanging the sums

$$CLR = \frac{1}{E(Y)} \sum_{W>C} \sum_{\forall j} \frac{W_j}{W}(W-C)P(Y=W) \qquad (47)$$

But,

$$\sum_{\forall j} W_j = W \qquad (48)$$

Resulting:

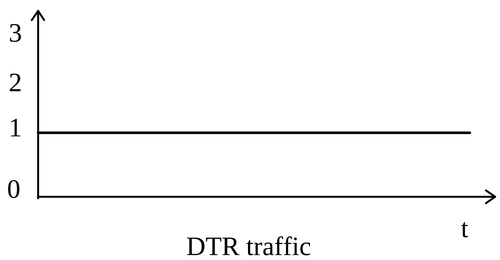
$$CLR = \frac{1}{E(Y)} \sum_{W>C} (W - C) P(Y = W) \qquad (49)$$

This is the same expression for CLR as shows (43), and this implies that expression (49) is correct. Whenever ECA is used for CAC, individual CLR_j's can be used as upper bounds for the bandwidth allocation scheme.

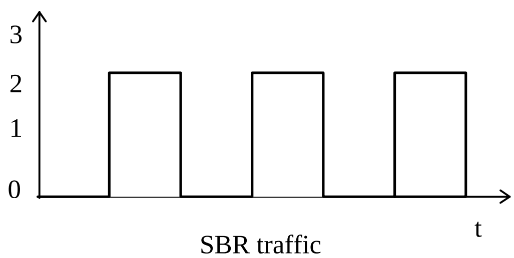
An example is given in the following figures to illustrate the relationship between the PC, CLR and individual CLR_j. The traffic scenario is simple: two sources are emitting cells into a link of capacity **C** equal to 2, sources have different traffic parameters and this is a heterogeneous scenario. The DTR traffic is defined as class j = 0; meaning source with one state (rate $r_0 = 1$, with the associated probability $p_0 = 1$). The SBR traffic is defined as class j = 1, this is an On-Off source;(state 0 : $r_{1,0} = 0$, with the associated probability $p_{1,0} = 0.5$; state 1 : $r_{1,1} = 2$, with the associated probability $p_{1,1} = 2$) giving a 0.5 utilisation. Both have the same mean rate (equal to 1[23]). The description of this traffic is summarised in the following table.

Traffic Class	*Mean rate*		*State 0*	*State 1*
0 (DTR)	1	rate	1	-
		probability	1	-
1 (SBR)	1	rate	0	2
		probability	0.5	0.5

Table 4-2 Traffic description



DTR traffic



SBR traffic

The mix of these sources gives a new System Status Vector, which is obtained by convolution (27).

[23] All rate are expressed in normalised metric.

Rate	*Probability*
1	0.5
3	0.5

Table 4-3 System Status Vector

Where the mean rate is clearly equal to 2 (1·0.5 + 3·0.5)

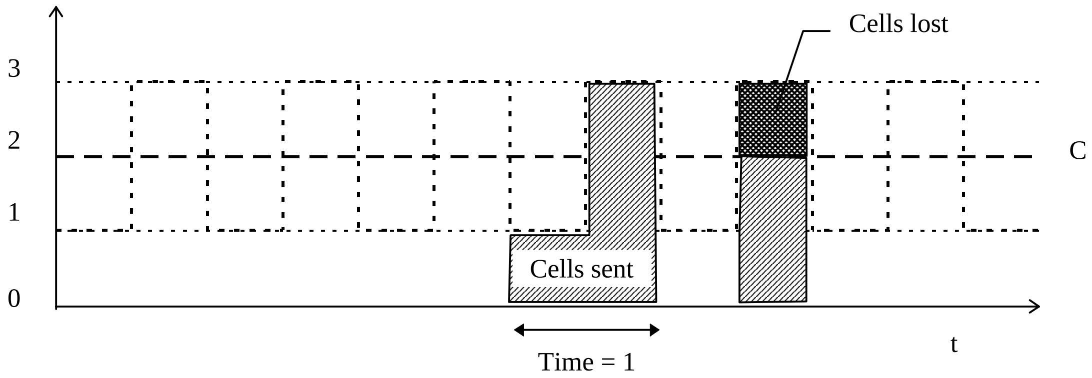


Fig. 4-4 Individual CLR.

As the above figure shows, the probability of congestion, using the formula (12) is

PC = P(Y>L) = P(3)= 0.5

By using the formula (13), assign C = 2, and E(Y) = 2, the corresponding CLR is obtained.

$$CLR(Y) = \frac{\sum_{L>C}(L-C)P(Y=L)}{E(Y)} = \frac{\sum_{L>2}(L-2)P(Y=L)}{2} = \frac{(3-2)P(3)}{2} = \frac{1\bullet 0.5}{2} = 0.25$$

Note that this value corresponds to the cell lost / cells sent ratio (cells sent = 4 units, and cells lost equal to 1 in one unit of time),

The individual CLR for class y is:

$$CLR_j = \frac{\sum_{W>C}\frac{W_j}{W}(W-C)P(Y=W)}{E(Y_j)}$$

For C = 2 the expression is now :

$$CLR_j = \frac{\sum_{W>2} \frac{W_j}{W}(W-2)P(Y=W)}{E(Y_j)}$$

W = 3 is the only state that W > 2, and $E(Y_j) = 1$ corresponding to the mean rate in this scenario, so:

$$CLR_j = \frac{\frac{W_j}{3}(3-2)P(Y=3)}{1} = \frac{W_j}{3}0.5 = \frac{W_j}{6}$$

Now we can apply this expression for j = 0 , $W_0 = 1$ corresponding to the instantaneous rate in this state, resulting $CLP_0 = 1/6$. And for j = 1, W1 = 2 resulting $CLP_1 = 2/6 = 1/3$.

The same results can be observed on Fig. 4-4. An analysis - based on the areas corresponding to cells sent, cells lost and the instantaneous rate for each traffic - leads easily to the previous results obtained by formulation.

Balanced Algorithm

The Balanced Algorithm combines appropriate storage requirements with low calculation cost.

Since some elements and sub-matrices in the System Status Matrix (SSM) are repeated, memory is employed wastefully, and excessive calculation takes place. This repetition is caused by convolution of the different source types. To reduce these drawbacks, partial results can be stored in a set of vectors associated with each type of source. It is therefore not necessary to store all the SSM: storing one Sub-Matrix (SMX) for each type of source is enough.

In the Balanced algorithm each partial probability corresponding to a type of source is evaluated using the MDF and is stored in an associate vector PV. PV stores rate and its associated probability for all possible combinations in a sub-matrix SMX. Therefore the associated probability for any rate is the result obtained from the pre-stored values which were obtained trough the convolution of sub-matrices.

Unfortunately, the multinomial distribution function does not detect identical rates from the additions of different partial values. In order to solve this problem the sub-matrix must be sorted and compacted. Therefore, the PV vector is first sorted and later in a compacted fashion to the convolution algorithm presented in Section 4.2.2. The most important aspect of this algorithm is the fact that the process is applied only to a sub-set of the link status matrix, i.e. a PV vector corresponding to a type of source, and the number of operations is reduced.

Zero storage requirements

The total Probability of Congestion (PC) of the system may be computed without storing pre-evaluated values. The partial probability corresponding to a row of the System Status Matrix is evaluated independently; then this partial value is added to the partial PC. For this evaluation, it is necessary to ensure that all the elements are generated in the appropriate order to obtain all possible combinations[24]. The main drawback of this option is the considerable number of calculations required. Therefore, this option is not recommended for real time evaluation.

4.3.5 Reducing the complexity of the calculation algorithm

Some modifications in order to reduce the number of exponential evaluations are now discussed. A table of factorial n!, from 0 to the maximum of connections, is pre-evaluated and stored once at the beginning of the evaluation of the PC.

Two strategies can be used to reduce the complexity of the evaluation for the expression (38):

$$\frac{N!}{n_0!n_1!\ldots n_{T-1}!}\quad p_0^{n_0}\cdot p_1^{n_1}\cdots p_{T-1}^{n_{T-1}}$$

a) The probabilities of the traffic descriptors for each type of traffic can be saved as its logarithm form. Therefore, only one exponential by element (39) of the SSM is needed:

$$P_R = \prod_{j=0}^{L-1} P_{j,r} = e^{\ln(\prod_{j=0}^{L-1} P_{j,r})} = e^{\sum_{j=0}^{L-1}\ln(P_{j,r})} \tag{50}$$

where

$$\begin{aligned}\ln(P_{j,r}) &= \ln\left(\frac{N!}{n_0!n_1!\ldots n_{SN-1}!}\, p_0^{n_0}\cdot p_1^{n_1}\ldots p_{S-1}^{n_{S-1}}\right) = \\ &= \ln(N!) + \sum_{i=0}^{L-1}\left\{n_i\ln(p_i) - \ln(n_i!)\right\}\end{aligned} \tag{51}$$

b) Only products are needed. Exponential evaluations can be avoided by storing the different combinations of $p_{j,i}^{k}$, varying j = 0 to L-1 , i = 0 to S_j-1 and k = 0 to the maximum of connections for each class (i.e. maximum equal to $2^8 = 256$). Note that

[24] These combinations correspond to the process described in Fig. 4-3

N_k!, and 1/N! are also stored [SLO92]. Therefore, the evaluation of (37) is obtained by:

$$N!\frac{1}{n_0!}\frac{1}{n_1!}\cdots\frac{1}{n_{T-1}!} \quad p_0^{\,n_0}\cdot p_1^{\,n_1}\cdots p_{T-1}^{\,n_{T-1}} \tag{52}$$

where only products are evaluated.

Option b) avoids exponential evaluations. Therefore, this option is chosen for further implementations. In section 5.2, some results applying this complexity-reduction mechanism is shown.

4.4 SUMMARY

Some limits of the formula-based Convolution Approach are presented in this section. To overcome the drawbacks of this method, a new method of evaluation is analysed: the Enhanced Convolution Approach (ECA).

First, convolution is applied to groups of the same type of sources (classes). After computing the convolution the same rate may appear more than once. By using the multinomial distribution function instead of the formula-based convolution, some of these coincidences are evaluated simultaneously. Finally, the global state probabilities are evaluated by convolution of the partial results. This method avoids accumulated calculations and saves storage requirements, particularly in complex scenarios.

The ECA computes the Individual Cell Loss Ratio CLR_j. Based on the ECA algorithm grouping connection in types, a expression for the CLR_j evaluation has been presented.

In order to reduce the number of exponential evaluations in (38) some modifications are discussed. A table of the factorial results, is pre-evaluated and stored once at the beginning of the evaluation of the PC. Furthermore, two strategies can be used to reduce the complexity of the evaluation. First, the probabilities of the traffic descriptors can be saved in its logarithmic form and only one exponential per element is needed. Secondly, exponential evaluations can be avoided by storing the different combinations of primary probabilities and, consequently, only products are needed.

5. CAC Based on the Probability of Congestion

5.1 CAC REQUIREMENTS

In section 2.3, the main CAC requirements have been summarised. QOS is the most important aspect and the statistical multiplexing gain is a desired benefit of ATM, but this section is focused on reasonable real-time processing, storage requirements as well as the maximisation of statistical multiplexing gain. Obviously, all the methods studied intend to guarantee QOS for all established connections.

Note that an important aspect of the CAC system is that only a Boolean response is necessary: Yes, the new connection demand can be accepted, or No, it must be rejected. This means that the exact value of Probability of Congestion (PC), or Cell Loss Ratio (CLR) is not always needed. For example, if during the evaluation of the PC, or CLR, the maximum admissible value is reached, the response is No, and the evaluation process is stopped.

Thus a relaxation of accuracy can be assumed with respect to the CAC requirements and the following discussion involves techniques to reduce the evaluation cost. Some achieve the same exact result whilst others reduce the accuracy, but in an acceptable manner (see experiments of demanded bandwidth in Chapter 3).

5.2 IMPLEMENTATION ISSUES

5.2.1 Cut Off Calculations

In order to evaluate the PC, it is not necessary to completely evaluate the entire distribution of the instantaneous rate.

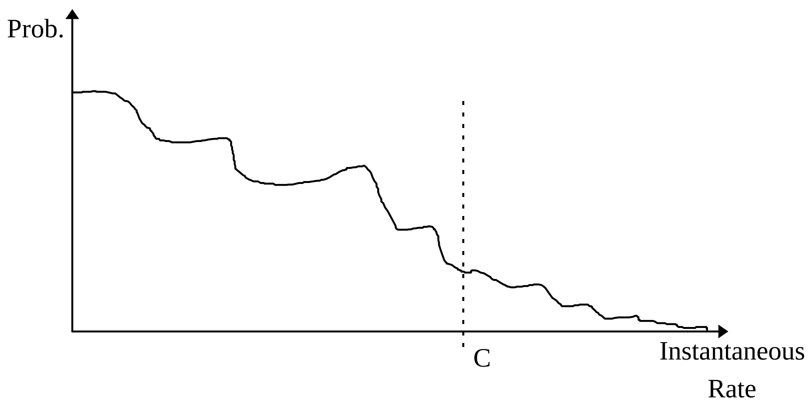


Fig. 5-1 Complete evaluation of the instantaneous bit rate Distribution.

Using ECA it is possible to evaluate a part of the statistical distribution. The techniques presented next are attempt to evaluate only the major relevant part of the system state: Congestion. All five cut-off improvements can be implemented simultaneously.

Link Capacity Cut-off.

A further reduction in calculation cost is obtained as follows: calculation of probability is only carried out in cases where the associated rate exceeds the bandwidth provided **C**. Note that the calculation of probability is considerably more complex than the calculation of rates; therefore, the objects for the calculation are selected a rate basis of the latter.

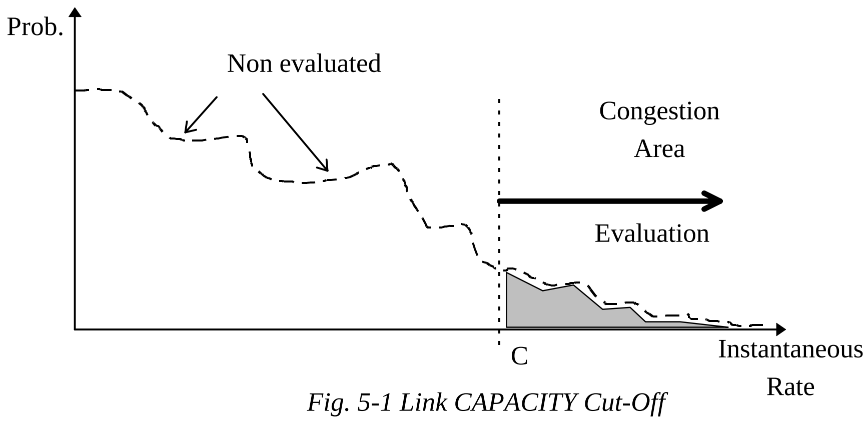


Fig. 5-1 Link CAPACITY Cut-Off

Probability of Congestion Cut-off.

CAC methods have one clear aim: to indicate whether or not a new connection will be accepted. For this decision, the PC[25] of the system is compared with a previously set value in order to guarantee the specific QOS. Therefore, if during the process of calculation the current PC exceeds a pre-set value, the process is stopped and the calculation cost is thus reduced.

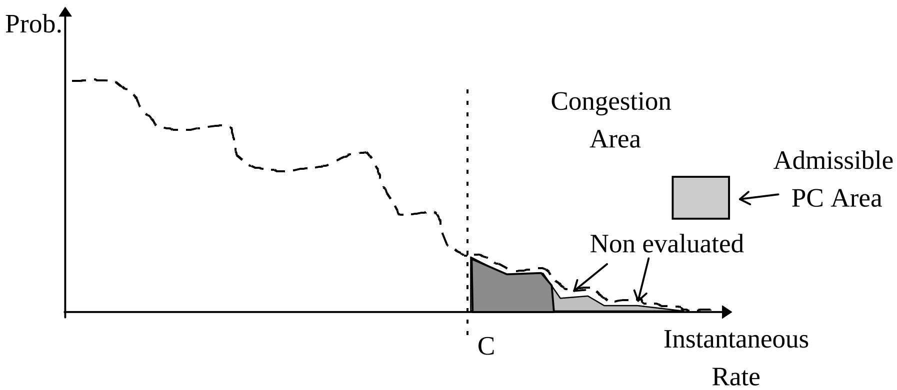


Fig. 5-2 Probability of Congestion Cut-Off

[25] The single metric to be analysed is the PC, but all the following argumentations could be applied to the CLR.

Partial Sorting Cut-off.

Furthermore, in each SSM[26] the rows generated are not examined in an arbitrary order, but are graded according to rate, so when the pre-set minimum rate **C** is reached, the process is terminated and a further saving computation time is achieved.

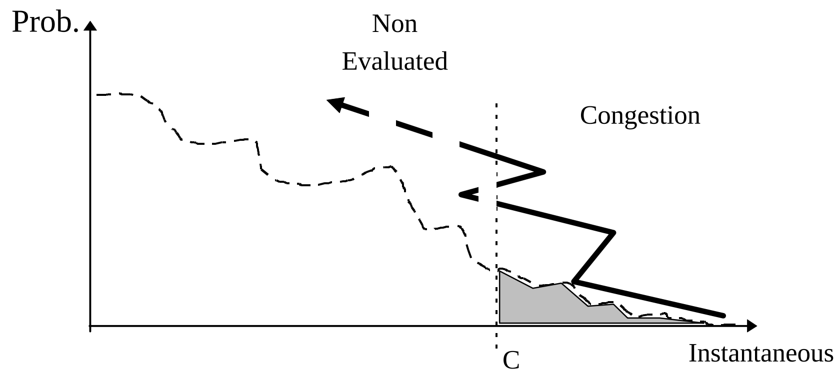


Fig. 5-3 Partial Sorting Cut-Off

Small probabilities Cut-off

When the probability obtained by the Multinomial Distribution Function (MDF), (see section 4.3.3), is less than a pre-set threshold value this result can be ignored. Therefore, the corresponding elements are not stored. This threshold value depends on the maximum congestion probability and on the number of connections. For example: if the admissible PC is equal to 10^{-4} and sources have an associated probability to the 0-state equal to 0.1, when all the connections (say N = 50) are convoluted, the probability of each element is obtained with the MDF.

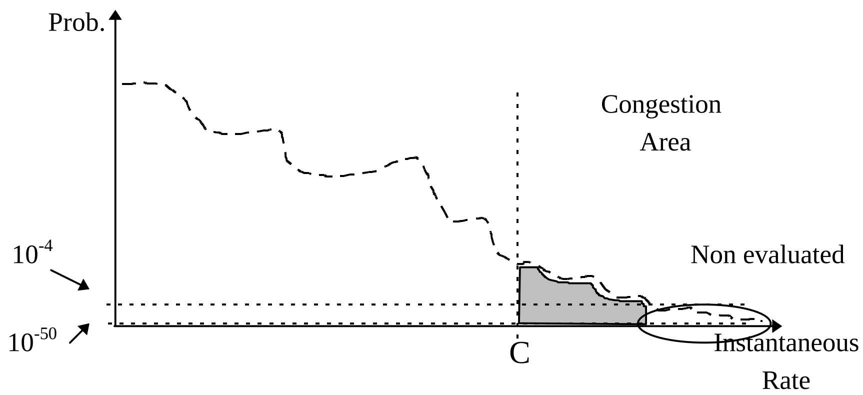


Fig. 5-4 Small probabilities Cut-Off

The probability that all sources are emitting in 0-state is close to 10^{-50}, so that this value is negligible against the admissible PC (10^{-4}); in practice, all probabilities less than say 10^{-20} [IVE90])˙ For each non-considered element the number of operations is decreased,

[26] SSM is the System Status Matrix. The SSM is described in Section 4.3 : The Extended Convolution Approach.

although the magnitude of any improvement depends upon the complexity of the remaining traffic classes.

Grouping states

For those classes with a large number of connections the majority of information enclosed in the sub-matrix may be summarised by grouping states. This mechanism could be applied independently to each class of sources before evaluating the second phase in the calculation of PC.

Set a rate quantum **Q** , all states belonging to the same quantum $\mathbf{Q_r}$ can be grouped into a new state (r^m, p^m), where p^m will always be the sum of the associated state probabilities ; r^m can be evaluated by the function:

$$q\ (r^k\ \forall\ k \mid r^k \in \{ Q_r^k \} \qquad (53)$$

There are two major options for definition the function **q**: the maximum rate of Q_r^k, or the mean rate of Q_r^k. Using the first option, the new rate is set to the maximum value of the current quantum rates, this approximation is pessimistic. The second option implies setting the new rate to the mean of the involved rates; this case is more realistic. However, by reducing states a smoother scenario is considered and consequently a slightly optimistic approach is applied. Note that considering an extreme hypothesis where the quantum is set to infinity and all states are reduced to a single state, corresponding to the mean rate, the system is modelled as a DBR source. The system is emitting at rate equal to the mean with associated probability equal to 1. By choosing adequate values for Q_r this method balances the conservative tendency of the convolution approach with an optimistic deviation.

Note that the aggregate bit rate is distributed according to a function where the tail region is strictly in monotonous decline for each class of traffic.This is true since a single source of each class of traffic is in monotonous decline as well. This is a valid assumption since presented works show this kind of traffic. It must also be noted that for ON-OFF sources all possible rates are multiples of the sources' peak rate.

$$\text{New rate} = g(r_0, r_1, .. r_i, ..., r_q) \qquad (54)$$

$$\text{New probability} = \sum_{i=0}^{q} p_i \qquad (55)$$

A second reduction can be obtained by grouping all states that have an associated probability which is less than a pre-set limit. This limit must be fixed to a value appreciably smaller than the value used for discarding very low probabilities (i.e. as shown by the small probabilities cut-off mechanism.

$$\text{New rate} = h(r_0, r_1, .. r_i, ..., r_s) \qquad (56)$$

$$\text{New probability} = \sum_{i=0}^{s} p_i \qquad (57)$$

Example: Given a Pre-sorted sub-matrix containing **m** elements (meaning that $r_k < r_{k+1} \; \forall$ k = 0 to **m**). Set **Q** equal to 3 elements, i.e. clusters of 3 elements, and set the probability limit to $\mathbf{p_l}$.

	Old Sub Matrix			New Sub Matrix	
	Rate	Prob.		Rate	Prob.
	r_0	p_0			
	r_1	p_1		$g(r_0,r_1,r_2)$	$p_0+p_1+p_2$
Q_0	r_2	p_2			
	r_3	p_3			
	r_4	p_4		$g(r_0,r_1,r_2)$	$p_0+p_1+p_2$
Q_1	r_5	p_5			
	...	...		...	...
	r_k	p_k			
Q_r	r_{k+1}	p_{k+1}		$g(r_k,r_{k+1},r_{k+2})$	$p_k+p_{k+1}+p_{k+2}$
	r_{k+2}	p_{k+2}			
	...	...		...	...
s	r_{m-s}	p_{m-s}	$< p_l$		
elements	...	...	$< p_l$		
	r_{m-2}	p_{m-2}	$< p_l$	$h(r_{m-s}, ..., r_{m-1})$	$p_{m-s+...+}\; p_{m-1}$
	r_{m-1}	p_{m-1}	$< p_l$		

where the function **g** and **h** may be: $\max.(r_k,r_{k+1},r_{k+2}) = r_{k+2}$ (58)

or $\text{mean}\,(r_k,r_{k+1},r_{k+2}) = \dfrac{r_k + r_{k+1} + r_{k+2}}{3}$ (59)

This process corresponds to the following figures:

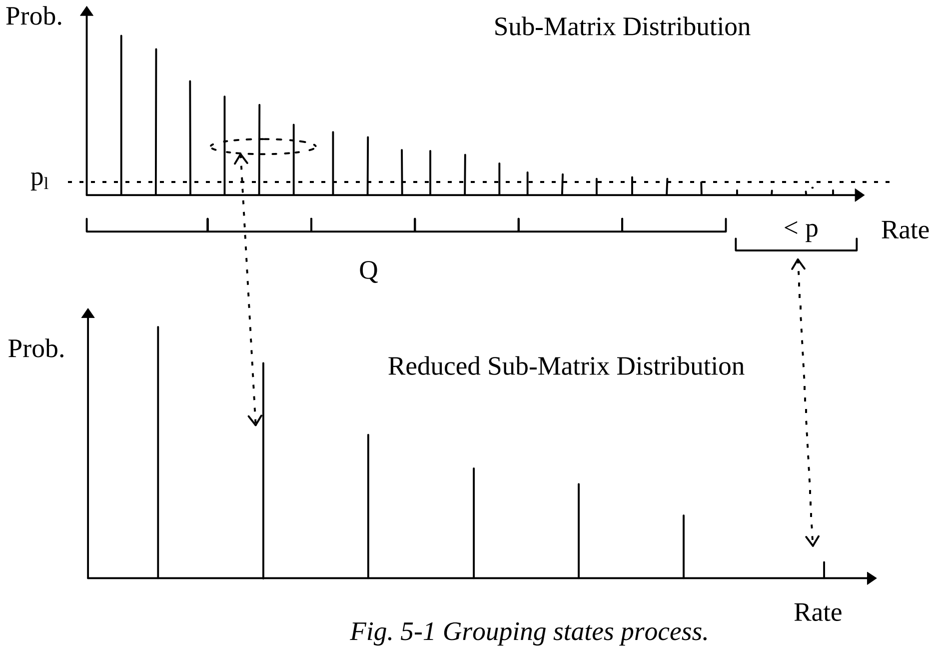


Fig. 5-1 Grouping states process.

In order to group states the expression (55) has been used to obtain the above figure.

5.2.2 The CAC scheme

Basic CAC objectives

In Section 2.3, the general definition and involved aspects about CAC have been presented. In previous work, several CAC algorithms have been proposed with the aim to produce an algorithm that is simple, robust and efficient [CUT93] and [CAST91]. Unfortunately, a trade-off between complexity and accuracy of the different algorithms exists.

Five approaches can be discerned: Peak-rate allocation, Convolution, Two-moment allocation, Linear and Two-level scheme. Peak-rate allocation is the simplest approach, but does not take advantage of the statistical multiplexing gain available in ATM links. Thus such as approach is normally discarded.

Convolution has been widely studied in this document. The convolution approach is the most accurate method but it has a high numerical complexity. In this work the convolution approach is the method chosen to implement the CAC function. The Linear approximation is simple but inaccurate. The two-moment allocation leads to satisfactory results only if the peak cell rate of connections is small compared with the link capacity. Finally, the two-level scheme combines the advantages of a fast real time processing first-level algorithm with more accurate approaches at the second level.

An admission boundary can be pre-evaluated by grouping traffic in classes. Pre-evaluated tables are available before starting the CAC mechanisms. This allows to use an accurate method giving fast response.

The above schemes will be investigate in the next sections.

Two-level CAC

A trade-off between complexity and performance of the different CAC schemes is perceived [R1022]. Thus a hierarchical CAC strategy using two levels has been proposed which combines the fast decision of simple CAC algorithms at first level with the accuracy of the more complex ones at the second level [CUT93]. First level CAC needs real-time response for the connection request. A background second level performs a refinement of this decision in a longer time scale [MIT94b].

Two strategies have been proposed from the European Community RACE programme [R1022]: The first strategy is characterised by using a required cell-rate for all connections in progress. This required cell-rate is updated by the second level algorithm using the convolution approach. The first level adds the peak rate for each connection set-up and subtracts the mean cell-rate for each release. The second level CAC algorithm performs an update of the transmission capacity based on a more complex approach, e.g. a convolution algorithm or a linear approach [CUT93]. The second strategy is based on a table where the admissible number of connections for different traffic classes is fixed and corresponds to a dedicated bandwidth allocation. When the capacity of the link or the traffic characteristics change all the table must be re-evaluated and consequently dynamic bandwidth allocation would be difficult (see section 2.4).

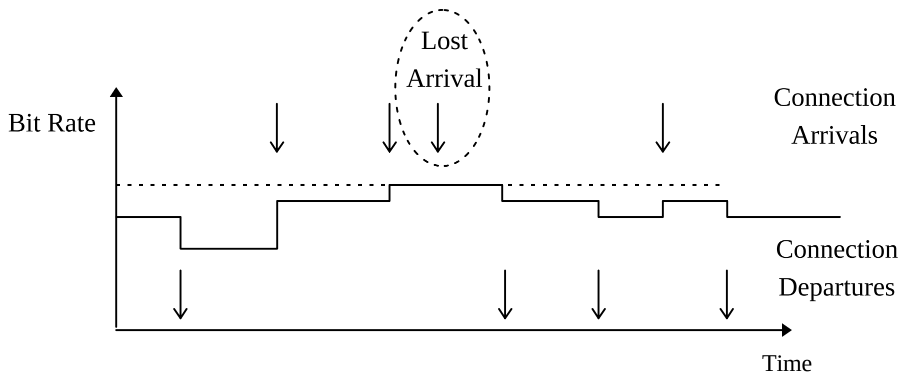


Fig. 5-2 Connection arrivals & departures

Due to its two-level nature, the admission / rejection behaviour of the implemented CAC functions depends on the process of connection arrivals and releases. In this dynamic case (involving connection level fluctuations) the CAC function cannot be described by a single boundary; the number of established connections per traffic class are required. However, the upper bound for these rejection points is given by the static admission boundary obtained when the time between consecutive connection set-up requests is large enough to update the required bandwidth using the second level algorithm, i.e. the convolution. The CAC implementation (two-level by using convolution as second level procedure) [R2061/8] has turned out to be a good compromise between fast response times to connection set-up requests and accuracy in predicting the required bandwidth in order to fulfil the network performance requirements.

First level Algorithms

As explained in Chapter 4, the different connections will be classified according to their bit rate requirements into **L** different service classes. The state vector $N_I = (n_{I,0}, n_{I,j}, \ldots, n_{I,L-1})$ denotes the system state at the considered time instant, where $n_{I,j}$ denotes the number of class **j** connections. Subscripts I and II are the parameters which are associated with the first and the second CAC levels respectively [MIT94b]. The first level algorithm modifies the required bit rate C_I according to the following set of manipulations:

$$C_I = \begin{cases} C_I + R_{t,j} & \text{for a connection set-up of traffic class j} \\ C_I - R_{d,j} & \text{for a connection release of traffic class j} \\ C_{II} & \text{for a parameter update} \end{cases} \qquad (60)$$

The required bit rate C_{II} will be calculated by the background algorithm based on convolution (see the section below). The bit rate increments and decrements of traffic of class **j** have been denoted by $R_{t,j}$ and $R_{d,j}$ and will be chosen according to the three different schemes. The simplest scheme allocates peak bit rate $\mathbf{c_j}$ for each new connection and releases mean bit rate $\mathbf{r_j}$ if a connection terminates.

$$R_{t,j} = r_j \text{ ; peak bit rate of the traffic class j} \qquad (61)$$

$$R_{d,j} = r_j \text{ ; mean bit rate of the traffic class j} \qquad (62)$$

This first level provides an upper bound for the required bit rate ,because peak bit rate will be allocated for each new connection, whereas, the mean bit rate will be released for a terminating connection. A performance improvement is possible, if the bandwidth allocation during an update interval takes account of a compensation of the bandwidth requirements of a connection set-up and a connection release within the same traffic class. For a complete description of this scheme, a differential connection vector is defined:

$$\Delta Nt= (\Delta Nt,0 , \Delta Nt,j,..., \Delta Nt,L-1) \qquad (63)$$

This vector indicates the difference between the actual connection vector and the valid connection vector from the previous time interval when the second level algorithm was called. By using this described improvement, previous expressions for the bit rate increment can be defined:

$$R_{t,j} \begin{cases} ck, \text{ for } \Delta N_{t,j} \geq 0 \\ rk, \text{ for } \Delta N_{t,j} < 0 \end{cases} \qquad (64)$$

Similarly for the bit rate decrement :

$$R_{d,j} \begin{cases} ck, \text{ for } \Delta N_{t,j} > 0 \\ rk, \text{ for } \Delta Nt,j \leq 0 \end{cases} \qquad (65)$$

An effective bit rate will also be defined for each traffic class. This effective bit rate will be adjusted by the background algorithm to the current load vector. Therefore, traffic class j will be removed completely from the system and the required bit rate will be recalculated. The total effective bit rate of traffic class j is defined as the decrease in the required bit rate if all connections of traffic class j are released. Using an effective bit rate in the first level of the CAC algorithm, the performance is improved, but the algorithm may become too optimistic.

Second level algorithms

The fluid flow model is, in principle, a promising candidate for a background algorithm [MIT94b]. But since this model is based on assumptions which will not, in reality, be valid (geometrical distribution burst and silence duration, no correlation among successive burst and silence periods), a bufferless fluid flow model is chosen which requires less information concerning the source characteristics and is much easier to solve. As indicated in Chapter 2, this model assumes that the excess traffic will be lost if the aggregate cell arrival exceeds the link capacity. Computing the aggregate bit rate distribution of all connections in progress can be done by a process of convolution with the bit rate distributions of the individual connections (see section 4.2).

For highly heterogeneous traffic mixes, particularly with respect to the burstiness, the linear scheme is usually optimistic. Logical partitioning offers an alternative solution, but suffers from being pessimistic. In such cases, and in order to operate close to the optimal admissible boundary, sophisticated CAC algorithms are necessary [MIT94b].

Pre-evaluation scheme

In order to offer a no-wait response to a new incoming demand pre-evaluated calculations will be investigated. The inter-arrival time between calls ranges from a few seconds to minutes, during which time the evaluation sub-system may be idle. During this period the status of the system can be dynamically updated for each type of source, and restored to the actual status. After such evaluation the system knows if it can admit a new call for each type of source. A new pre-evaluated vector is used to store results:

$$R_0, R_1, \ldots, R_j, \ldots, R_{L-1} \qquad (66)$$

Where Rj ∈ {YES, NO}.

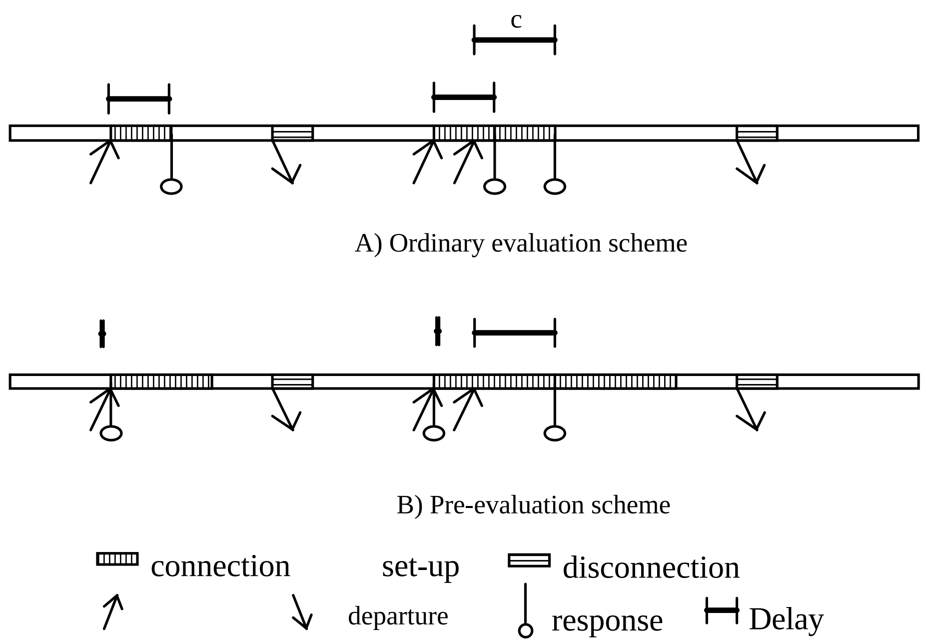


Fig. 5-3 Ordinary and Pre-evaluation CAC schemes

The evaluation sub-system always evaluates the CAC based on the new state of the system. Fig. 5-3.A shows the ordinary scheme. The incoming demands **a**, **b**, and **c** obtain response with delay. The amount of this delay depend on the evaluation cost for this type of source. Fig. 5-3.B shows the scenario with the proposed scheme. In this case, incoming demands **a** and **b** obtain an immediate response; the response to **c** is delayed because the evaluation sub-system is still evaluating the response vector R after the b connection. There is a certain probability of collision between a new connection demand and the last set-up connection process. This situation is analogous to the ordinary case.

The behaviour of this scheme depends on the process of connections arrivals and releases [R2061/8].

This proposal can be enhanced by adding new possibility for the R_j state; the value UNKNOWN. This new response would be given in case of any time-out during the evaluation and means that the CAC has no response at this time.

Now Rj ∈ {Y, NO, UNKNOWN}

That modification can save calculations when the arrival and departure demands are very dynamic. For example, when an incoming call or a disconnection arrives the pre-evaluation sub-system interrupts the background process and processes the new event.

In the re-evaluation procedure, disconnection processes are considered to be all elements of the response vector R that are equal to NO. After the call termination, the responses of R equal to NO are transformed to UNKNOWN to be re-evaluated. Note that YES responses do not need to be updated because the system load has decreased. These responses will continue equal to YES.

However, the connection set-up process, in a re-evaluation consist of all the elements of R that are equal to YES, so that those elements equal to YES are transformed in UNKNOWN. In this case the NO responses do not vary because the system load has increased.

Convolution as first-level CAC

ECA, using cut-off mechanisms, and pre-evaluations can be carried out simultaneously; this leads to proposal for an alternative to two level CAC methods when convolution is used as at first level. In this proposal, a new parameter is introduced: the maximum response time of the CAC algorithm T_0. This parameter to limit that is imposed on the CAC control system to respond to an incoming connection demand. The different possibilities are summarised in Table 5-1. In the first column, the exact response is presented as reference where the evaluation has been done completely without time limitations. The two level scheme has no significant delay since the first level is based on a simple approach (peak/mean cell rate).

The ECA algorithm has satisfactory accuracy. This makes it possible for a YES response in a situation when the two level scheme responds NO, e.g. case **b** and consequently, some statistical multiplexing gain is achieved. But, if the state of the link is complex, the ECA would give a conservative response (NO), because there was not enough time to evaluate the complete state of the link, as in case **c**.

Finally, by combining ECA and Pre-evaluation scheme, new benefits are provided. In case **c,** prolonged evaluation is necessary but the pre-evaluation scheme fulfils the responses within the time demanded. As table 1 shows, cases **d** and **g** give negative response for different situations. However, in case **d** the connection demand should be accepted.

Case	Complete Evaluation		Two level		ECA		ECA & Pre-evaluation	
	Response	Delay	Response	Delay	Response	Delay	Response	Delay
a	YES	$< T_0$	YES	0	YES	$< T_0$	YES	$< T_0$
b	YES	$< T_0$	NO	0	YES	$< T_0$	YES	$< T_0$
c	YES	$> T_0$	NO	0	NO	T_0	YES	$< T_0$
d	YES	$>> T_0$	NO	0	NO	T_0	NO	T_0
e	NO	$< T_0$	NO	0	NO	$< T_0$	NO	$< T_0$
f	NO	$> T_0$	NO	0	NO	T_0	NO	T_0
g	NO	$>> T_0$	NO	0	NO	T_0	NO	T_0

Table 5-1 Comparison between different Two- level CAC schemes.

An ATM connection passes through several multiplexing and switching stages in the network. What is really of concern to the end users is the end-to-end performance, and not just the figures of an individual stage. Modelling the connection by a concatenation of multiplexing stages, as shown in Fig. 5-4, and making the network independence assumption, allows for the derivation of the end-to-end figures in terms of the respective ones of the individual stages.

The accumulative effect of inaccuracy, end-to-end, is showed below.

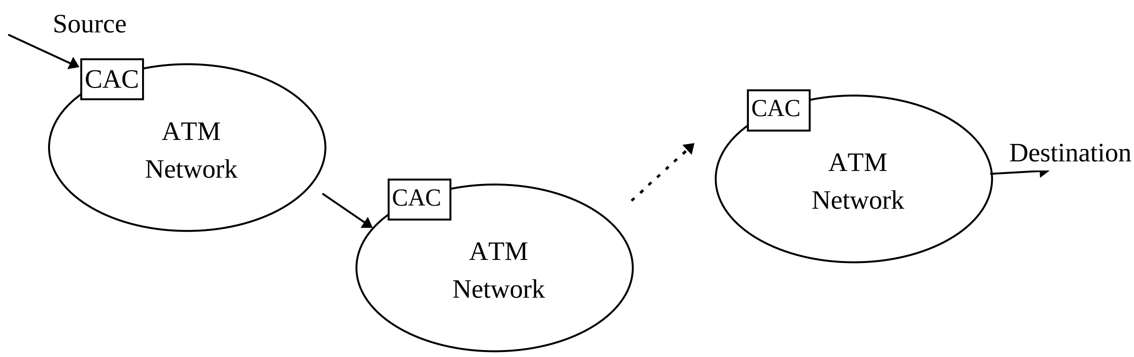


Fig. 5-4 End-to-end CAC problem.

An example is the end-to end CLR, which can be evaluated as the sum of the CLRs corresponding to each stage. Therefore, the accuracy in the CAC scheme for each network has a bearing on the final resource utilisation. For example, the cumulative effect of blocking probability for new calls is found when a new connection demand checks the complete path; if part of this path is rejected by its respective CAC evaluation scheme, the whole connection is rejected and the call demand is lost [MIT94]. In this way, the connection blocking probabilities as function of the bandwidth capacity are obtained in order to compute the necessary total bandwidth capacity for dimensioning purposes. Then, the blocking probabilities for an end-to-end connection of any of the considered service types can be obtained as a function of the blocking probabilities of those switching nodes the connections goes through [CAS95].

5.3 COST EXPERIMENTS

5.3.1 Source Model (Traffic characterisation)

All experiments have been based on a GMDP source model. The GMDP model describes the behaviour of a traffic source at cell and burst level. The number of states for j-type sources is S_j. In each state i (with i = 0, 1, ..., S_{j-1}), during the corresponding sojourn time $Soj_{Tj,i}$, cells are sent with regular inter-arrival times (constant rate $R_{j,i}$,)

The sojourn time $Soj_{Tj,i}$, in a given state $S_{j,i}$, is a random variable. From mean sojourn times the state probabilities for a given set of assumptions can be obtained. In a stationary model, the probability that a source is in state i is evaluated by the ratio between the sojourn time for this state and the sum of all sojourn times.

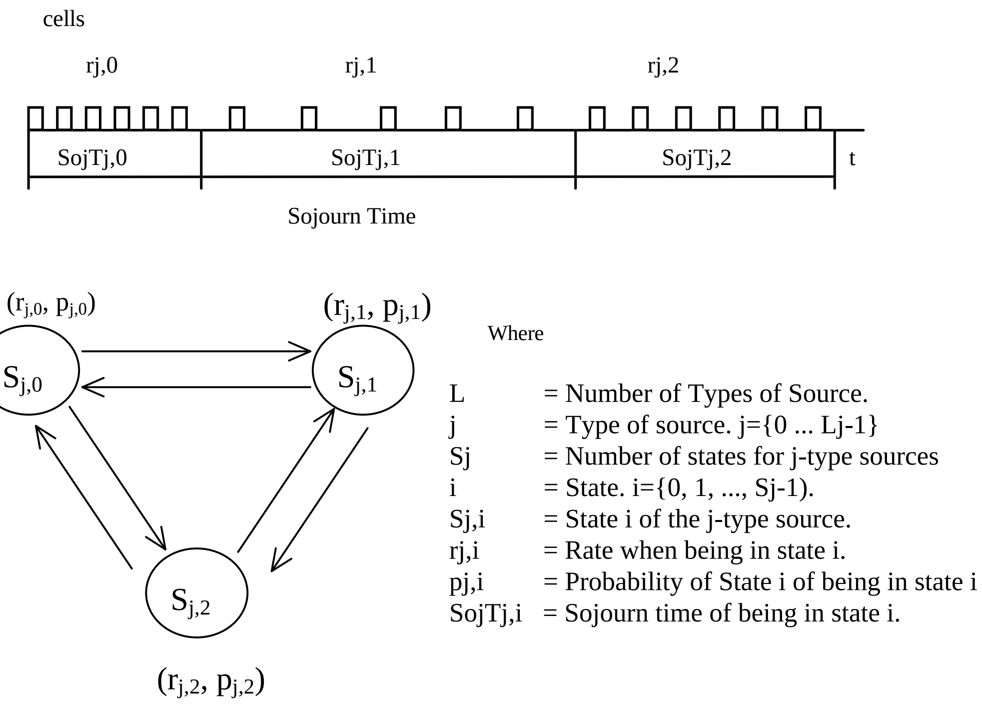


Fig. 5-5 GMDP Source Model

The above figure presents the Source Status Diagram for a generic source of type j. The number of states is 3 (S_j = 3).

5.3.2 Evaluation Cost Experiments

Objectives

A major challenge of the presented method is to obtain real-time response without losing accuracy. Several experiments using different evaluation methods and adding cut-off mechanisms are presented.

Definition of experiments

A set of traffic scenarios are analysed where all traffic is multiplexed onto a single link. Homogenous (Ho) and Heterogeneous (He) traffic is evaluated. The convolution approach formula-based evaluation cost is compared with the Enhanced Convolution Approach (ECA) evaluation cost. The following table shows the used traffic.

			State	
		0	1	2
A2	bit rate (Mbit/s)	0.4	2	
	probability	*0.625*	*0.375*	
A2a	bit rate (Mbit/s)	0.4	2	
	probability	*0.375*	*0.625*	
B2	bit rate (Mbit/s)	2	10	
	probability	*0.625*	*0.375*	
D2	bit rate (Mbit/s)	0	10	
	probability	*0.9*	*0.1*	
A3	bit rate (Mbit/s)	1	2	10
	probability	*0.7*	*0.2*	*0.1*
C3	bit rate (Mbit/s)	3	6	30
	probability	*0.7*	*0.2*	*0.1*

Table 5-2 Source Characteristics

The evaluation cost is measured in terms of a number of different metrics:

- <u>Storage requirements</u> are measured in elements, each element has to store a rate and a probability. Storage requirements corresponds to the amount of utilised memory in order to characterise the state of the link.

- The <u>time</u> parameter corresponds to CPU time and is expressed in normalised time[27]. The time necessary to obtain the final result varies depending on the processing capacity of the computer.

[27] This corresponds to seconds in the experiments presented

- Sorting techniques are required to put the partial status vector in order. The quick sort algorithm is used when necessary. The cost is expressed as $x \cdot \log(x)$, where x is the number of elements to be arranged.

- Calculation cost is expressed as a normalised combination of additions and products. The following expression is used: ((additions + products) / 1000.). For simplicity, the computation cost of additions is considered equal to the computation cost of products [D122]. This expression enables extrapolation of results to faster computation environments.

The computational efficiency has been measured for both the basic convolution (in shadow rows) and the enhanced convolution approach. Different cut-off mechanisms, for the enhanced convolution approach, are also presented. A comparison in time is shown in the following tables in the 'speed-up' column, the time obtained for the basic convolution is set to 1; the evaluations for the enhanced convolution approach are normalised to this value.

In order to extend the cost of the evaluations, a set of complex scenarios have been chosen. Consequently, the capacity of the link is set at least to 600 Mbit/s. The number of connections is given by the given Cell Loss Ratio (in the order of 10^{-4}).

General Remarks

Sorting is the dominant factor for the formula-based convolution, whereas cost evaluation is the dominant factor for the enhanced convolution. Normally, the formula-based convolution needs to convolute a large number of elements, once a new connection is added and a sorting and compacting process is carried out. On the other hand, the enhanced convolution approach requires more previous calculations: ECA generates compacted vectors, via the multinomial distribution function, and as these vectors correspond to each class, the number of elements obtained is generally smaller than the general approach.

With reference to the cut-off mechanisms, the major conclusion is the efficacy of the small probabilities cut-off. The elements necessary to characterise the state of the link achieve a reduction ratio of about 20% of the initial elements. The first direct implication is the reduction in the storage requirements. Moreover, this reduction in the intermediate vectors implies a imminent rise in the evaluation time from 20 up to 200 times more (depending on the heterogeneity of the traffic) in the presented experiments.

Results of Cost Experiments

SCENARIO			Cut-off			RESULTS						
Experiments	*Connections*	*Link*	*PC max.*	*MinVal*	*Reduction*	*PC*	*CLP*	*time*	*Speed-up*	*elem.*	*Sorts*	*Cost*
Ho1a	120	155				3,50E-05	4,26E-07	0,06	**1**	121	72	0,94
2-states						"	"	0,01	**6**	121	0	1,43
Ho1b	480	540				2,34E-04	2,15E-06	0,39	**1**	481	1467	3,93
						"	"	0,03	**11**	481	0	5,86
				1,E-20		"	"	0,03	**11**	194	0	4,49
				1,E-10		"	"	0,03	**11**	132	0	4,23
				1,E-08		"	"	0,02	**23**	116	0	4,17
				1,E-07		"	"	0,02	**23**	107	0	4,13
				1,E-06		2,32E-04	2,05E-06	0,02	**23**	97	0	4,09
Ho2a	40	155				1,67E-04	1,26E-05	0,06	**1**	333	115	3,78
3-states								0,01	**6**	333	1	11,53
Ho2b	160	450				8,86E-04	3,06E-05	0,76	**1**	1413	2594	3,78
						"	"	0,04	**19**	1413	13	162,39
				1,E-10		"	"	0,02	**38**	389	2	146,35
				1,E-08		"	3,05E-05	0,02	**38**	340	2	145,68
				1,E-07		8,83E-04	3,00E-05	0,02	**38**	312	1	145,32
					100,3,2	"	3,07E-05	0,04	**19**	471	13	158,46
					100,5,2	7,46E-04	3,35E-05	0,04	**19**	283	13	157,67
				1,E-10	100,3,2	"	3,07E-05	0,02	**38**	130	2	145,77

Table 5-3 Cost results for Homogeneous traffic

Time is measured in discrete units so that different experiments give the same value (the same can be said for the speed-up column). This effect is perfectly observed for homogeneous traffic when the amount of time required for the complete evaluation is normally short (see ‘time’ column in Table 5-3.

SCENARIO			Cut-off			RESULTS						
Experiments	*Connections*	*Link*	*PC max.*	*MinVal*	*Reduction*	*PC*	*CLP*	*time*	*Speed-up*	*elements*	*Sorts*	*Cost*
He1	100+75	600				1,79E-04	3,46E-06	0,16	**1**	476	323	4
(both 2 states)						"	"	0,01	**16**	177	0	34
						"	"					
He2a	50+22	155				1,57E-04	7,80E-06	0,13	**1**	1223	351	13
(2 and 3 states)						"	"	0,01	**13**	222		46
He2b	160+150	600				3,67E-04	8,22E-06	5,38	**1**	23453	19329	94
(2 and 3 states)						"	"	0,16	**34**	11637	12	1422
				1,E-10		"	"	0,03	**179**	1972	2	
He3	140+80	600				6,28E-04	3,87E-05	4,55	**1**	12746	16254	154
(2 and 3 states)						"	"	0,29	**16**	3462	3	2738
				1,E-10		"	"	0,03	**152**	913	1	250
				1,E-08		"	"	0,02	**228**	719	1	168
				1,E-07		6,80E-04	3,824-5	0,02	**228**	621	1	133
			1,00E-03			x	x	0,29	**16**	3462	3	2738
			1,00E-04			x	x	0,24	**19**	3462	3	2332
			1,00E-05			x	x	0,23	**20**	3462	3	2235
			1,00E-06			x	x	0,22	**21**	3462	3	2138
He4	60+75	600				1,99E-04	1,17E-05	4,86	**1**	13252	17480	162
(3 and 3 states)								0,95	**5**	3439	5	9398
				1,E-10		"	1,16E-05	0,08	**61**	1013	1	700
				1,E-08		"	1,16E-05	0,05	**97**	814	1	440
				1,E-07		1,97E-04	1,13E-05	0,04	**122**	709	1	327
					100,3,2	"	"	0,12	**41**	1147	5	1095
					100,5,2	2,00E-04	1,16E-05	0,06	**81**	689	5	432
				1,E-08	100,3,2	1,97E-04	1,18E-05	0,02	**243**	372	1	97
				1,E-08	100,5,2	2,00E-04	1,16E-05	0,01	**486**	163	1	70

Table 5-4 Heterogeneous scenarios (low heterogeneity: two types of traffic).

x This values have been truncatred by the PC maximum cut-off.

Experiments	Connections	Link	PC max.	MinVal	Reduction	PC	CLP	time	Speed-up	elements	Sorts	Cost
He5	120+60+40	600				1,23E-04	2,66E-05	1,86	**1**	2609	1634	22
(2, 2 and 2 states)						"	"	0,22	**8**	223	0	2170
				1,E-10		"	"	0,04	**47**	133	0	320
				1,E-08		"	"	0,03	**62**	119	0	214
				1,E-07		"	"	0,02	**93**	110	0	153
				1,E-06		1,22E-04	2,61E-05	0,02	**93**	100	0	111
He6	120+80+40	600				3,66E-04	1,57E-05	4,07	**1**	10387	14395	124
(2, 3 and 3 states)						"	"	57,62	**0**	1675	4	598031
				1,E-07		"	"	1,39	**3**	552	1	14756
				1,E-06		"	"	0,87	**5**	483	1	9184
					3	3,70E-04	1,58E-05	2,18	**2**	559	4	22546
				1,E-20	3	3,69E-04	1,58E-05	1,79	**2**	442	2	18620
				1,E-10	3	3,62E-04	1,54E-05	0,42	**10**	292	1	4393
				1,E-15	3	3,68E-04	1,57E-05	1,06	**4**	382	2	11018
				1,E-08	3	3,63E-04	1,56E-05	0,24	**17**	420	1	2448
					5	3,70E-04	1,58E-05	0,5	**8**	337	4	5011
				1,E-20	5	3,67E-04	1,57E-05	0,65	**6**	303	2	6748
				1,E-15	5	3,68E-04	1,57E-05	0,39	**10**	262	2	4009
				1,E-10	5	3,67E-04	1,57E-05	0,16	**25**	202	1	1613
				1,E-08	5	3,65E-04	1,56E-05	0,09	**45**	171	1	908
			1,E-06			x	x	32,48	**0**	1675	4	336875
			1,E-06	1,E-15	3	x	x	0,43	**9**	559	2	4331
			1,E-06	1,E-10	5	x	x	0,05	**81**	202	1	379

Table 5-5 Heterogeneous scenarios (medium heterogeneity: three types of traffic).

x This values have been truncatred by the PC maximum cut-off.

SCENARIO				**RESULTS**						
Experiments	Connections	Link	MinVal	PC	CLP	time	Speed	elements	Sorts	Cost
He7a	60+60+40+50	600		3,11E-04	6,88E-05	3,08	1	5129	3454	21,45
(all 2 states)				"	"	6,31	0	214	0	68043
			1,E-10	"	"	1,07	3	154	0	11779
			1,E-08	"	"	0,65	5	130	0	7173
			1,E-07	"	6,87E-05	0,44	7	128	0	4853
			1,E-06	3,10E-04	6,81E-05	0,30	10	118	0	
He7b	10+160+10+10	1000		5,53E-04	9,19E-05	1,75	1	6909	1995	27,99
(all 2 states)				"	"	0,16	11	194	0	1648
			1,E-10	"	"	0,07	25	109	0	627
			1,E-08	"	"	0,05	35	99	0	438
			1,E-07	"	9,18E-05	0,04	44	94	0	393
			1,E-06	5,52E-04	9,08E-05	0,04	44	87	0	302

Table 5-6 Heterogeneous scenarios (High heterogeneity: four type of traffic).

The Partial Sorting cut-off has not been presented in this work due to the difficulties to implementation; the experiments carried out show that the control to apply the cut-off is sometimes more costly than the benefits obtained by the reduction of evaluated elements.

Homogeneous traffic

In the presence of only one class of traffic the application of the ECA only increases the speed-up factor. However, the number of elements is the same for both methods as expected.

As has been previously noted, the utilisation of the small probabilities cut-off reduces the size of the vector which characterise the state of the link. The evaluation delay obtained in this experiment is enough for real time CAC utilisation, less than 0.04 seconds in the presented scenarios.

Heterogeneous traffic

Sorting cost for convolution based on formula is the predominant factor, this cost is especially large in complex scenarios: heterogeneous and 3-state traffic. However, the dominant factor for ECA is the calculation cost. With a little reduction of accuracy, using small probabilities cut-off, a great improvement in terms of both storage requirements and calculation cost, is obtained.

In Table 5-6, a scenario with high heterogeneity is showed. The main objective of this table is to show that the ECA is faster than the basic convolution when the number of connections for the different classes of traffic is not balanced. For example, if the number of connections is equal to 200, ECA performs better in the case for a distribution of connections equal to 10+160+10+10 than so a distribution equal to 60+60+40+50, (including the direct application of ECA without cut-off mechanisms). This is a reasonable assumption due to the fact that a link, in presence of heterogeneous traffic, a background traffic constitute by a great number of voice connections and other more heavy traffic with a relatively smaller number of connection.

Experiments combining all cut-off mechanisms show that the utilisation of ECA in real-time CAC environments is always possible.

6. Experiments

This Chapter discusses different aspects of the behaviour of cell streams in an output buffer corresponding to an ATM link and is illustrated by experimentation. First, CAC experiments relating to Fuzzy logic and (M+1)-MMDP approaches are presented. Measurements in an ATM test bed are also included. Finally, experiments of bandwidth allocation based on analysis illustrate the main differences between different methods, such as the Gaussian, Fluid Flow and linear approximation.

6.1 CAC EXPERIMENTS

The experiments described in this section refer to a single ATM link and the QOS is expressed in terms of cell loss at the output buffer of an ATM switch. The traffic sources are VBR sources, modelled as On-Off sources described by the peak and mean bit rates and mean burst length.

In the following, (i) a deterministic CAC approach (ECA) based on an enhancement of the convolution algorithm aiming to achieve computation tractability, (ii) a heuristic approach (FCAC) based on a combination of fuzzy logic to represent the traffic knowledge and genetic algorithms to provide an adaptation of the tool to changes in traffic characteristics are presented.

In the following each of the experiments is described and the cell loss results obtained by FCAC and ECA are presented.

6.1.1 Description of Fuzzy-based CAC Approach

The two further CAC approaches are presented, ECA and FCAC, which will be compared in terms of their accuracy for the cell loss prediction for homogeneous and heterogeneous traffic scenarios, their tendency towards pessimism or optimism in the cell loss predictions, and on their ease of development.

A further experiment will assess how well each CAC method optimises the use of network resources in relation to a fixed cell-loss ratio, fixed link capacity and output buffer size. This experiment involves varying the allowed number of accepted connections for homogeneous scenarios and the mean average load, i.e. the sum of the mean bit rates for the accepted connections, for a number of heterogeneous scenarios.

In applying fuzzy logic to CAC it is envisaged that the maximum cell loss per connection can be predicted when a candidate connection is added to a background traffic scenario (made up of the connections already accepted into the network). The Fuzzy based CAC approach [RAM96b] (FCAC) uses the user declared statistical parameters (mean and peak bit rates and mean burst length) for all connections (including the incoming connection), on a node-to-node basis. The cell loss prediction will help the CAC in the decision making in the following manner: if the estimated cell loss value does not violate the cell loss requirements of existing connections and the candidate connection, the connection will be accepted; otherwise it will be rejected.

A basic challenge associated with CAC based on a heuristic method is knowledge elicitation, i.e. the transfer of knowledge from some source into a fuzzy rule base, of the relationship between traffic offered to an ATM switch and obtained network performance, e.g. cell losses. This is because all the knowledge that can be obtained on ATM traffic is expressed in terms of input/output data pairs (examples) collected from measurements. The FCAC approach presented, uses an automatic design of the associated fuzzy system based on a method of learning from examples [HER95].

In previous studies (see also [RAM96a] and [RAM96b]), a learning method that learns the relationship between a specific traffic pattern and a cell loss ratio performance value has been presented. This method is used to automatically design the fuzzy rule base system, that is, to define (a) the fuzzy sets for the fuzzy variables in the antecedent and consequent of each fuzzy rule and (b) a finite set of fuzzy rules able to reproduce the input-output system behaviour.

The learning method uses Genetic Algorithms to determine the membership functions for the fuzzy sets in the domain of the fuzzy variables in the antecedent and consequent of each fuzzy rule and to generate a set of rules able to reproduce the input-output system behaviour, using a set of training examples (ATM traffic scenarios). This process is repeated on-line every time a new set of traffic examples is available, allowing the FCAC to adapt to changes in the traffic patterns.

A rule has the following form

“if X_1 is A_{1j} and ... and X_4 is A_{4j} then Y is B_j“

Where

j	denotes the *j*-the rule of implication
A_{ij} B_j	are the fuzzy sets
i	refers to one of four antecedents
X_i	are input fuzzy variables
Y	is the output fuzzy variable

The fuzzy variables (FV) are defined as follows:

FV	*Description*	*Characteristics*
X_1	Mean Offered Load	linguistically expresses the degree of utilisation of the system
X_2	Mean to Peak Ratio	aggregated and linguistically expresses how close the behaviour of each connection is to that of a constant bit rate (CBR) connection
X_3	Relation between Peak rate and Link Capacity	linguistically expresses how much bandwidth (time resource) the connections require in terms of peak bit rate
X_4	Mean Buffer Allocation	linguistically measures the requirements of the connections in terms of buffer space (space resource)
Y	cell loss ratio (CLR)	represents the maximum predicted ratio of cells lost per cells sent for the connections. The cell loss ratio (CLR) refers to the loss ratio experienced by each connection in a single ATM link and not to the average loss for the multiplexed connections.

Table 6-1 Fuzzy variables

The bell-shaped membership functions [CHW94] were chosen for the definition of the linguistic terms of the antecedents of the rules in order to achieve flexibility in the design of the fuzzy domain set. The linguistic terms for the consequent of the rules are given by singletons instead of fuzzy sets.

The inference algorithm is based on the work by Campos and Gonzalez [CAM91]. The value obtained from the inference process is a crisp output value that represents the negative power of ten of the maximum cell loss ratio to be expected per connection.

CAC methods	*Satisfaction of cell loss requirements*	*Efficient use of network resources*	*Simplicity of parameter determination*	*Speed calculation at network nodes*	*Ability to adapt to new services*	*Simplicity of implementation*
FCAC	yes, if there is previous knowledge of the traffic scenario for which the cell loss prediction is to be made ; otherwise either pessimistic or optimistic	yes	identifies the source traffic behaviour using the parameters declared by the user at connection set-up, that is, mean and peak rates and mean burst length.	inference calculations are fast ; the time spent in “on-line” training depends on required prediction accuracy and how fast the traffic scenarios change in time.	Yes, if the traffic characteristics of new services can be identified using the set of chosen traffic parameters ; otherwise, easy to re-design to cope with modifications.	inference algorithm is very simple, the learning algorithm is complex.
ECA	yes, if the size of the buffer is so that can cope with cell level fluctuations caused by the asynchronous arrival of cells (cell scale influence) ; otherwise, conservative	can be very conservative for network scenarios with large buffer size (the convolution algorithm assumes a bufferless service model).	same as above	fast if traffic sources can be grouped in classes and a smaller number of classes (less than 5) is considered. Several cut-offs mechanisms can be applied.	yes, if the traffic characteristics of new services can be captured using the peak and mean cell rates. Also allows the characterisation in many states	Simple, requires memory and CPU resources.
(M+1)-MMDP	optimist for traffic scenarios that do not exhibit burst scale congestion	yes	sources are modelled as on-off sources defined by the rate in the active state, the mean on and off periods.	is not suitable for real life implementation due to the complexity of calculations	yes, if the new services traffic characteristics can be modelled using the on-off model	simple, requires solving system of linear equations.

Table 6-2 Summary of CAC methods

6.2 EXPERIMENTS IN HOMOGENEOUS SCENARIOS VS. (M+1)-MMDP

Yang and Tsang in [YAN95] describe an approach to estimate the cell loss probability in an ATM multiplexer for traffic scenarios with identical traffic sources (homogeneous traffic) using the *Markov Modulated Deterministic Process* (MMDP) to approximate the actual arrival process and then modelling the ATM multiplexer as an MMDP/D/1/k queuing system. Cells arrive according to a deterministic renewal process whose rate is controlled by a Markov process $\{X(t), t \geq 0\}$. Using queuing analysis, Yang at al derived a formula for the cell loss probability expressed in terms of the limiting probabilities of a Markov chain.

Based upon the results, two approximation methods were proposed: the first method approximates the actual arrival process by an $(M+1)$-state MMDP, where M is the number of identical sources. The modulating Markov process $X(t)$ is the number of sources that are active at time t (hence, the MMDP has $M+1$ states). This approximation provides accurate results "for all cases in which burst-level congestion is the main contributing factor of loss" [YAN95].

The second approximation models the arrival process using a two-state MMDP. The two states of the modulating Markov process $X(t)$. The state 0 characterises the under-load region of the number of sources that are active at time t, i.e. the total cell arrival rate in any state is less than or equal to the link capacity. The state 1 represents the overload region, i.e. the total cell arrival rate in any state is greater than the link capacity. This approximation is "sufficiently accurate for applications where the average burst length is large (such as large file transfers, image retrievals, etc.)" [YAN95]. In the comparative experiments discussed later only the (M+1)-MMDP approximation is considered because it is sufficiently accurate for the scenarios studied.

The traffic sources used in the experiments are On-Off sources defined by parameters: peak and mean bit rate and mean burst length sources (see also Table 6-3) and are in conformance with the traffic characteristics for the On-Off sources presented by [YAN95] in table II, pp. 122.

Traffic Class	*Peak Rate (Mbit/s)*	*Mean Rate (Mbit/s)*	*Burst Length (cells)*
Voice	0.064	0.022	58
Data	10	1	339
Image	2	0.087	2604

Table 6-3 Traffic characteristics of On-Off sources

The results obtained by Yang were checked using an ATM cell rate simulator, LINKSIM, developed by Pitts [PIT93] and used as the training set for the fuzzy tool. The results obtained using ECA and FCAC are plotted in figures Fig. 6-1 to Fig. 6-7. The number of sources and the link capacity values were chosen in order to obtain a range of cell loss values between 10^{-2} and 10^{-8}.

The simulation was implemented assuming that all the On-Off sources were independent. For each traffic scenario, the criterion for stopping the simulation is that the width of the 95% confidence interval should be less than 10% of the estimated cell loss probability. For traffic scenarios with a high number of low rate sources, LINKSIM failed to produce a result satisfying the stopping criterion after two months and the simulations were forced to stop before they met the criterion. For these cases, the average and maximum cell loss values were considered to be the same and its value was read from the simulation result obtained by Yang et al for the specific traffic scenario, see [YAN95] pp.123-124.

The following table describes the Experiments A1 to A6 . The buffer size has been set to 50 cells for all experiments. A range of load and peak were used.

Experiment	*Traffic type*	*# sources*	*load*	*Link Capacity (Mbit/s)*	*Peak/Link capacity*	*Figure*
A1	voice	250-300	0.8-0.95	7	0.00914	Fig. 6-1
A2	voice	15-25	0.05-0.08	0.7	0.0914	Fig. 6-3
A3	data	160-300	0.5-0.85	350	0.28	Fig. 6-4
A4	data	8-26	0.15-0.5	52	0.192	Fig. 6-5
A5	image	4-20	0.05-0.25	30	0.133	Fig. 6-6
A6	image	80-220	0.25-0.65	7	0.285	Fig. 6-7

Table 6-4 Specification of the experiments for homogeneous traffic mixes (mathematical approaches)

Figures Fig. 6-1 to Fig. 6-7 plot the cell loss predictions given by the (M+1)-MMDP, ECA and FCAC approaches and also the reference average and maximum CLP obtained via simulations for experiments A.1 to A.6.

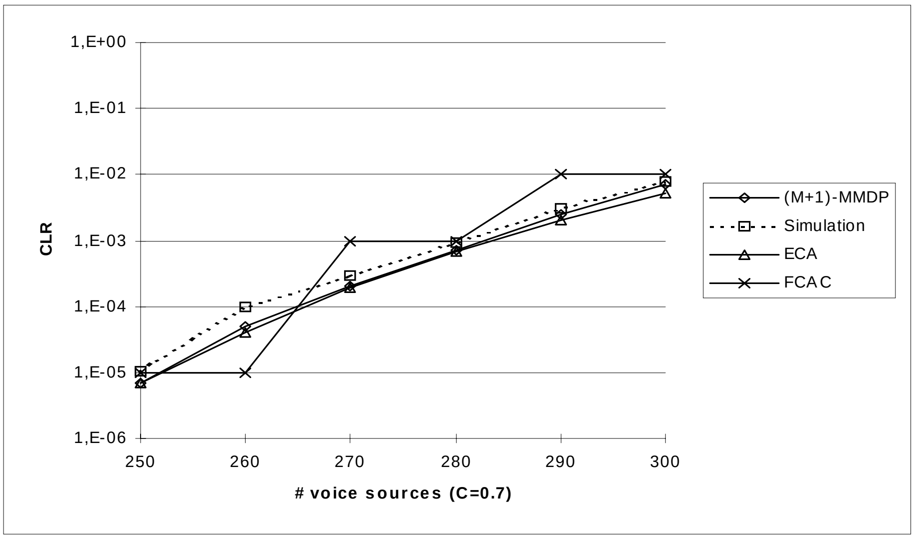


Fig. 6-1 Experiment A1 (fig 2 of [YAN95], pp123).

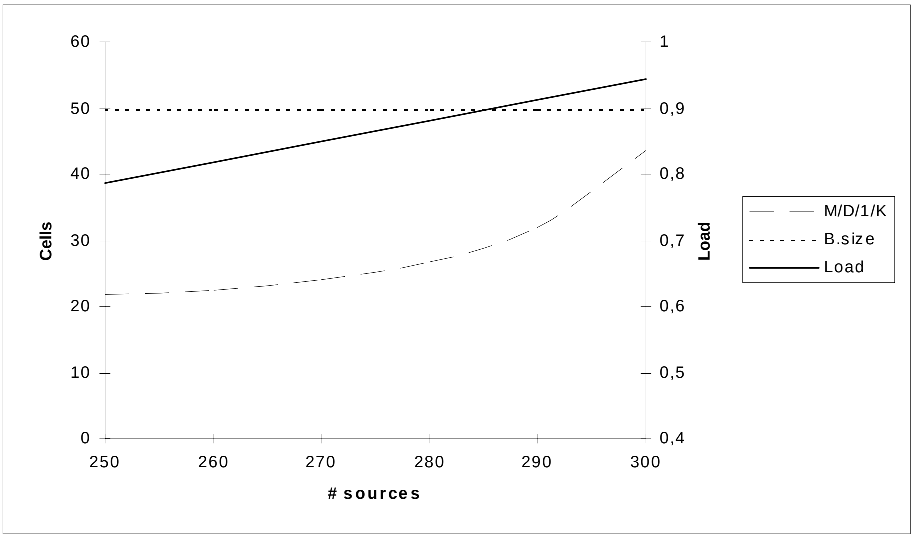


Fig. 6-2 Experiment A1 Load and buffer size against number of sources

In the experiment A1, the number of sources and the corresponding load of the link is high. Fig. 6-2 shows the necessary buffer size evaluated by the M/D/1/K queuing system (see section 3.3.3) against the number of active sources (according to the load of the link). In this situation the buffer size (used in simulation, 27 cells) is close to the evaluated by the M/D/1/K queuing system, since the values obtained by the ECA approach are slightly optimistic. We can conclude that the load in the link is the determinant factor in this scenario. The same argumentation can be said for the (M+1)-MMDP method.

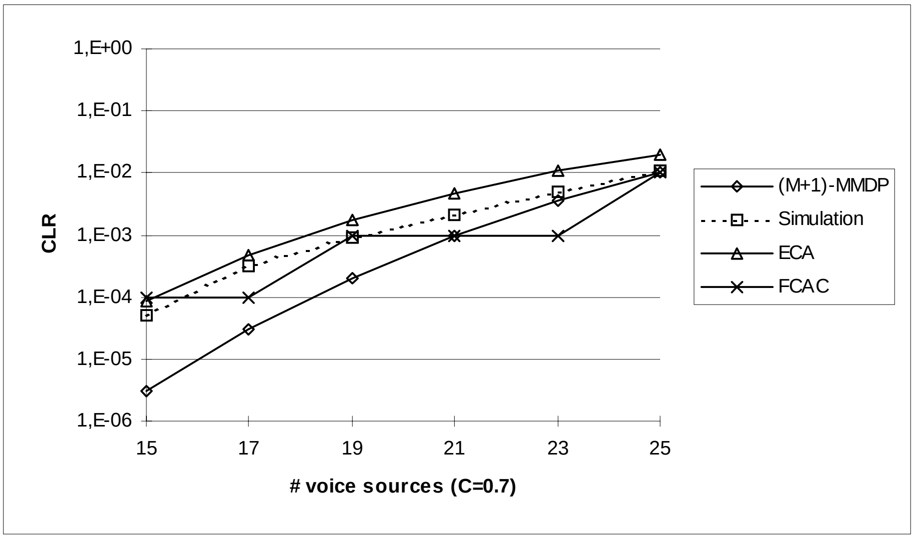


Fig. 6-3 Experiment A2 (fig 8 of [YAN95], pp123).

The load of the link is low, since the size of the buffer is enough to cope with the arrivals at cell level. The experiment A2 ECA and FCAC are more accurate than the (M+1)-MMDP method, which is too optimistic. As expected in this experiment, the convolution approach is an upper limit of the CLR for the link.

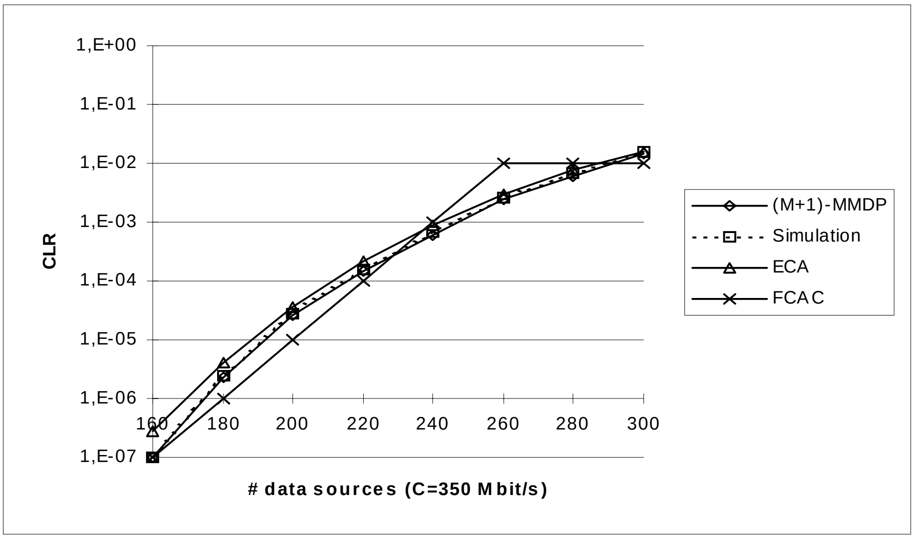


Fig. 6-4 Experiment A3 (fig 3 of [YAN95], pp123).

The medium-high load of the link and the high number of connections in experiment A3 result in good approximation for all methods.

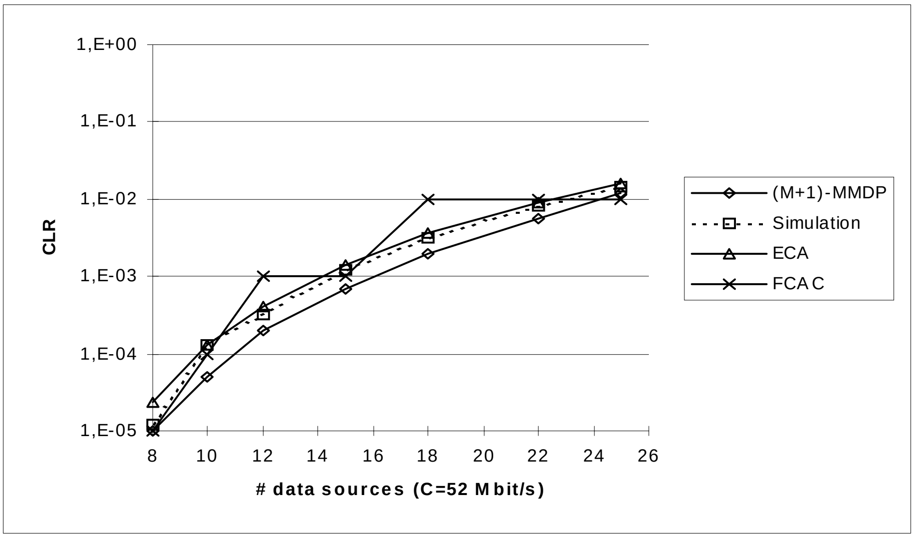


Fig. 6-5 Experiment A4 (fig 9 of [YAN95], pp123).

In this experiment the ECA is very accurate, especially in presence of high load. The (M+1)-MMDP results are slightly conservative. The FCAC method alternates between conservative and optimistic predictions; this is due to the low accuracy in this implementation (one order of magnitude).

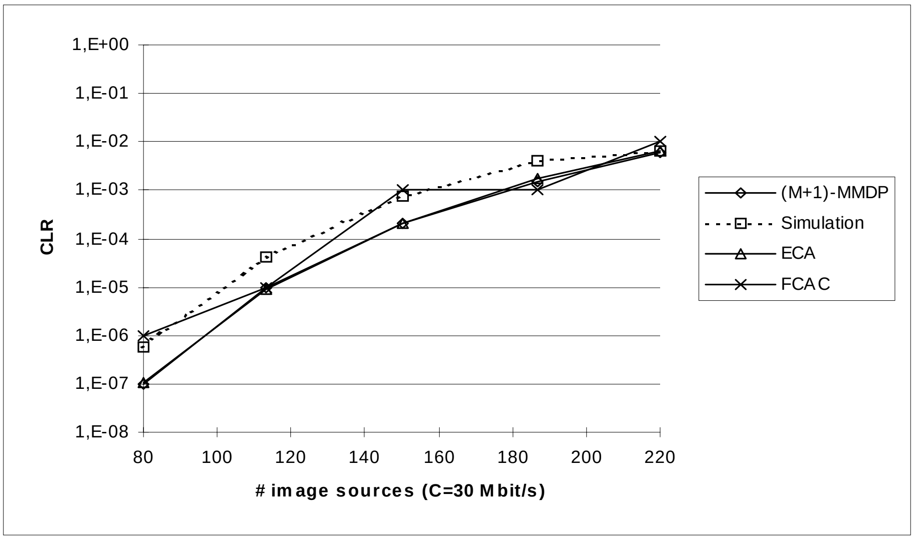


Fig. 6-6 Experiment A5 (fig. 4 of [YAN95], pp. 123).

In experiment A5 the link is loaded with a large number of connections. The results obtained by all approximations are generally optimistic, because of the high load and the large mean burst length of the traffic observed.

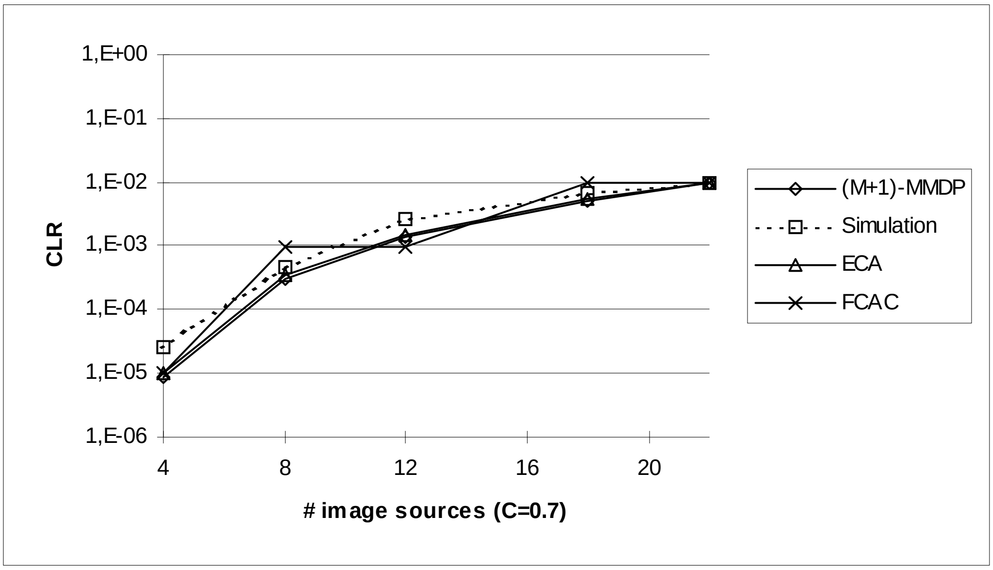


Fig. 6-7 Experiment A6 (fig 10 of [YAN95], pp. 124).

Experiment A6 yields similar conclusions to experiment A5. The single difference observed is for low number of connections where ECA and the (M+1)-MMDP approaches are slightly optimistic. In this situation there is a deficient multiplexing gain.

General Comments

The ECA prediction is quite accurate for the traffic scenarios plotted in Fig. 6-1 and Fig. 6-3 although the average cell loss curve obtained by simulation is above the ECA predicted result in Fig. 6-1. The reason for the further case, in this experiment buffer cannot contain cell level arrivals due to the large amount of active sources with relation to the small buffer size, whilst the effect of the later figure is the gain obtained by queuing part or the whole burst is ignored by the ECA approach.

The cell loss prediction given by the (M+1)-MMDP approximation is slightly more accurate than the ECA prediction in the case of Fig. 6-1 but can be very optimistic in the case of Fig. 6-3 for low average loads. For a traffic scenario of 15 sources, the value predicted for the cell loss by the (M+1)-MMDP approximation is 3.e-6 and the simulation value is $5{\cdot}10^{-5}$.

The ECA prediction provides an upper bound for the average cell loss probability for the traffic scenarios plotted in Fig. 6-4 and Fig. 6-5. ECA is conservative for this scenario, but becomes more accurate when the number of sources increases. The prediction given by the (M+1)-MMDP approach is also accurate and the accuracy increases with the average load as observed previously. The FCAC prediction conforms with the cell loss value given by the simulation. For Fig. 6-5 the values obtained via simulations for the maximum cell loss and average cell loss vary significantly and this is why FCAC predicts a more conservative value.

Yang et al. describes experiment A3 (Fig. 6-4) as a traffic scenario where “burst-level congestion” is predominant and that accounts for the accuracy of the (M+1)-MMDP prediction for this traffic scenario. As can be seen, for the traffic scenario studied in Fig. 6-4, the sources’ mean burst length is quite long (339 cells) and therefore the influence on cell loss caused by queuing part of the buffer is more significant than in the case of voice traffic. Experiment A4, Fig. 6-5, is described by Yang et al. as the traffic scenario equivalent to the one representation of the transfer of large files. For experiment A4, the output buffer only copes with cell level fluctuations and all the loss is caused by lack of bandwidth resources; this constitutes a scenario for which the ECA approach is accurate.

The ECA prediction for both Fig. 6-6 and Fig. 6-7 is below the average cell loss curve obtained via simulations but the difference is not significant to classify the prediction given by ECA as optimistic. The same behaviour can be observed for the prediction given by the (M+1)-MMDP approximation. The FCAC prediction conforms with the cell loss value given by the simulation. Note, that the rounding to the next power of ten can sometimes mislead the observer to think that the approach is either pessimistic or optimistic for a particular traffic study. For the image traffic scenarios studied in experiment A5 and A6, the mean burst length is so long that the size of the buffer is not adequate to cope with fluctuations at cell level and this explains why the ECA approach does not provide an upper bound for the average cell loss value. ECA is also more optimistic in experiment A5 due to the large number of active sources.

For the traffic scenarios studied in Fig. 6-6 to Fig. 6-7 it can be concluded that the cell loss predictions given by the (M+1)-MMDP approximation and the ECA approach are generally accurate. The FCAC predictions are also in the range of the expected cell loss values for these experiments.

6.3 EXPERIMENTS VS. MEASUREMENT (BASEL TEST-BED)

RACE projects provide ATM test-beds on which measurements and tests can be performed [KUH94]. This set of experiments enables comparison of the average cell loss results obtained from on-line measurements in the Exploit ATM test-bed in Basel [R2061/28] with the cell loss predictions given by both the ECA and FCAC approaches for homogeneous and heterogeneous traffic scenarios on a single ATM link. Although FAC attempts to predict the maximum cell loss ratio per connection instead of the average cell loss ratio for the aggregate traffic, for the sake of comparison with the results obtained from on-line measurements, FCAC was trained to predict the average cell loss ratio instead.

In Table 6-5 the traffic sources used for the comparison experiments are described. The link capacity considered is 155.52 Mbit/s and the output buffer size is 27 cells.

Traffic type	*Peak Rate (Mbit/s)*	*Mean Rate (Mbit/s)*	*Mean Burst L. (cells)*
A.31	31.1	6.22	1467
A.32	31.1	1.56	734
B.31	7.78	3.89	917
B.32	7.78	0.39	183
C.31	1.94	0.97	229
C.32	1.94	0.39	91

Table 6-5 Characteristics of the traffic sources

Three pairs of traffic are used for this group of experiments. The mean burst length is reduced for each basic traffic class; accordingly, the mean rate is also reduced for a given peak rate.

Homogeneous scenarios

For the following set of experiments, different scenarios are defined. One type of traffic is multiplexed into the output buffer in an ATM link.

Experiment	*Traffic type*	*# sources*	*load*	*Peak/Link*	*Figure*
B1	A.31	7-16	0.28-0.64	0.200	Fig. 6-8
B2	A.32	14-34	0.14-0.34	0.200	Fig. 6-10
B3	B.31	22-34	0.55-0.85	0.050	Fig. 6-11
B4	B.32	180-280	0.45-0.7	0.050	Fig. 6-12
B5	C.31	115-160	0.7-0.99	0.012	Fig. 6-13
B6	C.32	260-380	0.65-0.99	0.012	Fig. 6-15

Table 6-6 Specification for homogeneous traffic mixes (measurements)

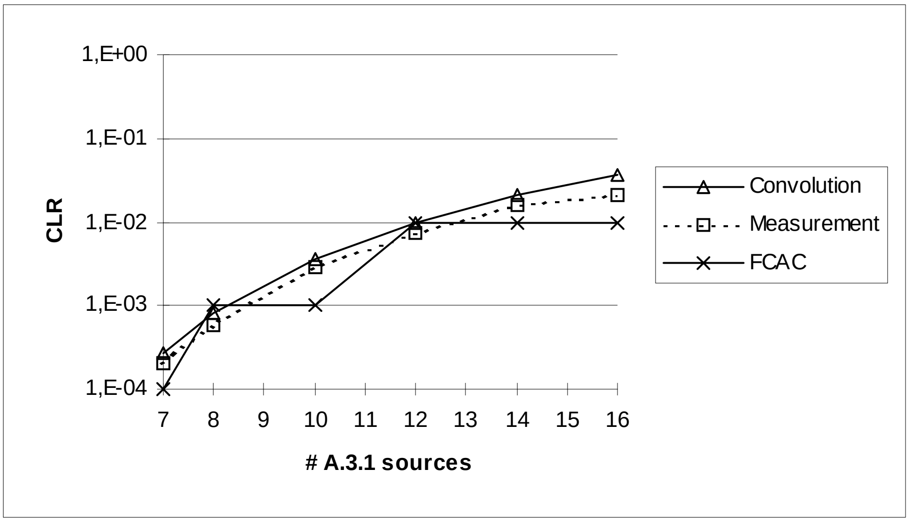


Fig. 6-8 Experiment B1 Cell loss predictions for a traffic mix of A.31 sources

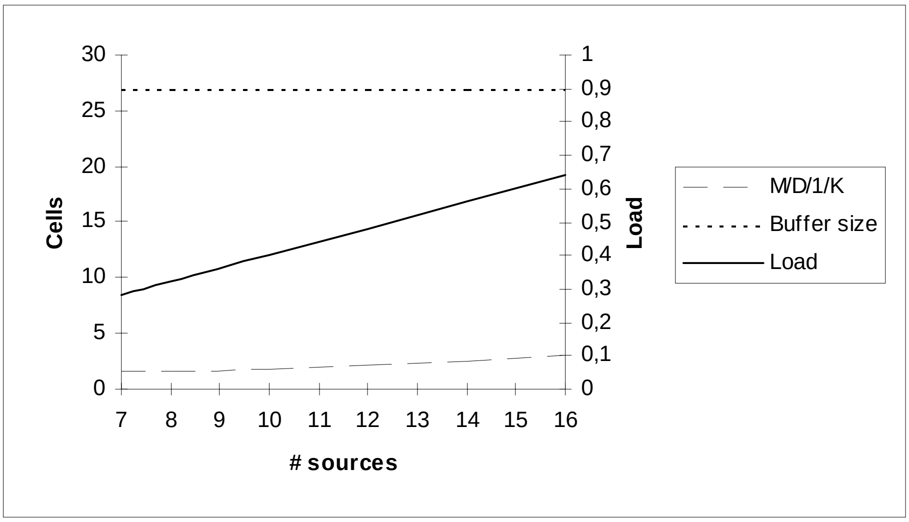


Fig. 6-9 Experiment B1 Load and Buffer size against the number of sources

Fig. 6-8 shows a good approximation in the CLR predictions obtained by convolution. The buffer size evaluated by the M/D/1/K queuing system (see Fig. 6-9) is smaller than the current buffer size. In this experiment the mean burst length is large, so the hypothesis of bufferless operation is achieved and the CLR predictions obtained by the convolution approach are an upper bound for the CLR measurements.

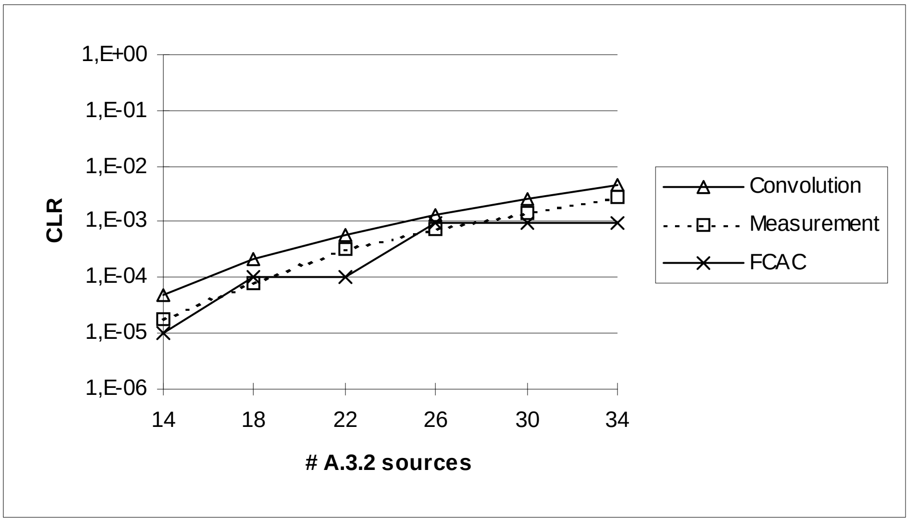


Fig. 6-10 Experiment B2 Cell loss predictions for a traffic mix of A.32 sources

The reduction of the load and mean burst length in experiment B2 decreases the CLR evaluated by convolution with reference to experiment B1. The result is more conservative in this situation.

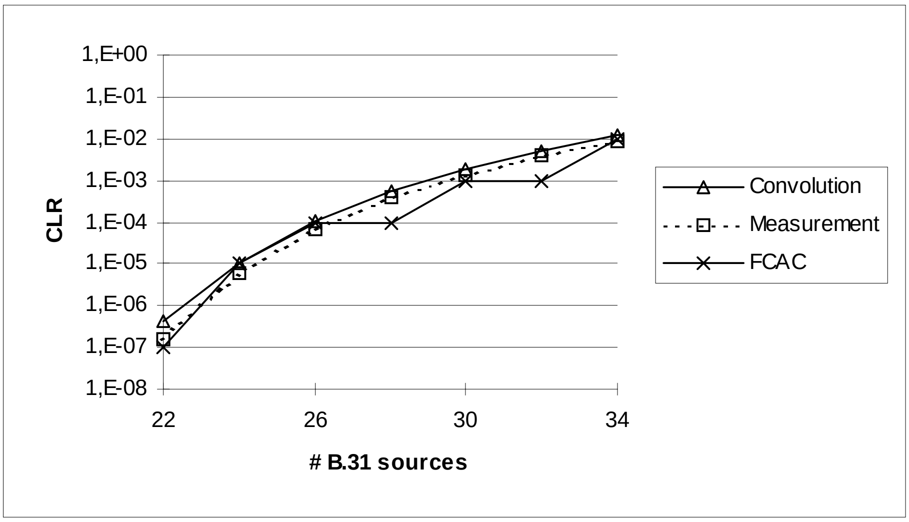


Fig. 6-11 Experiment B3 Cell loss predictions for a traffic mix of B.31 sources

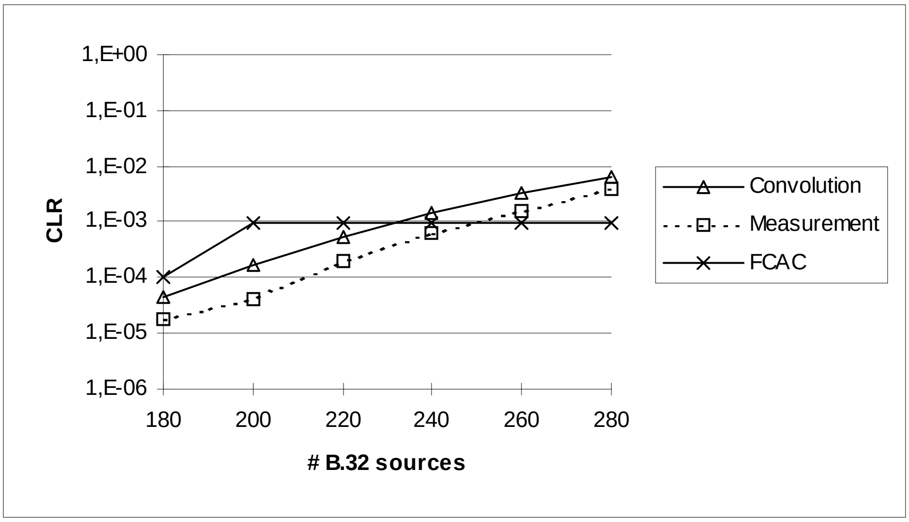


Fig. 6-12 Experiment B4 Cell loss predictions for a traffic mix of B.32 sources

ECA approximation is accurate in experiments B3 and B4. When the load of the link increases the convolution approximation is close to the measurements results.

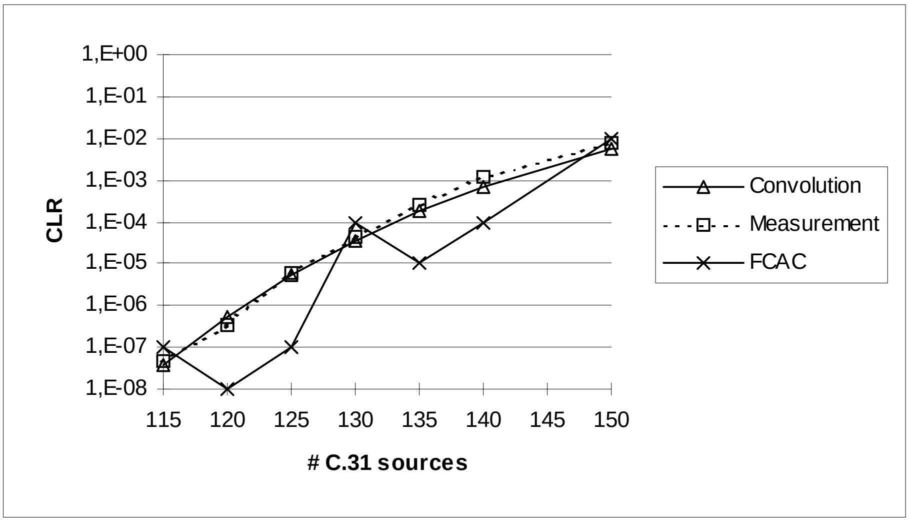


Fig. 6-13 Experiment B5 Cell loss predictions for a traffic mix of C.31 sources

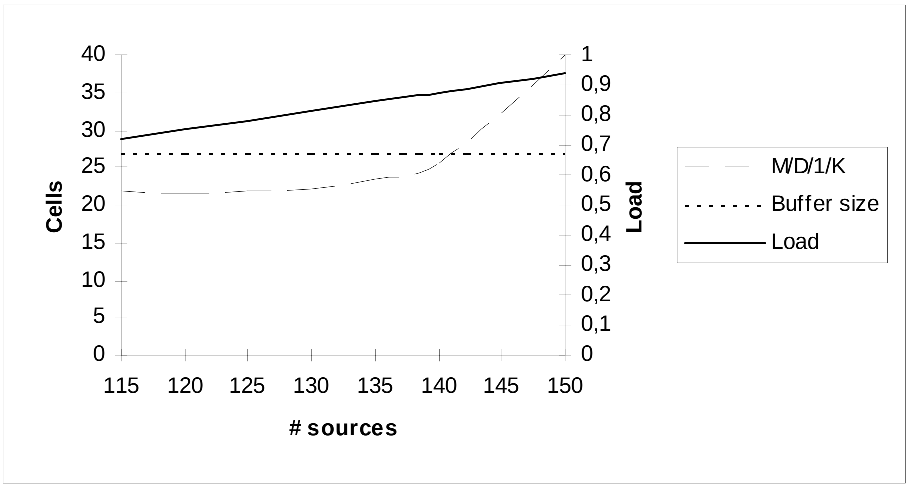


Fig. 6-14 Experiment B5 Load and buffer size against the number of sources.

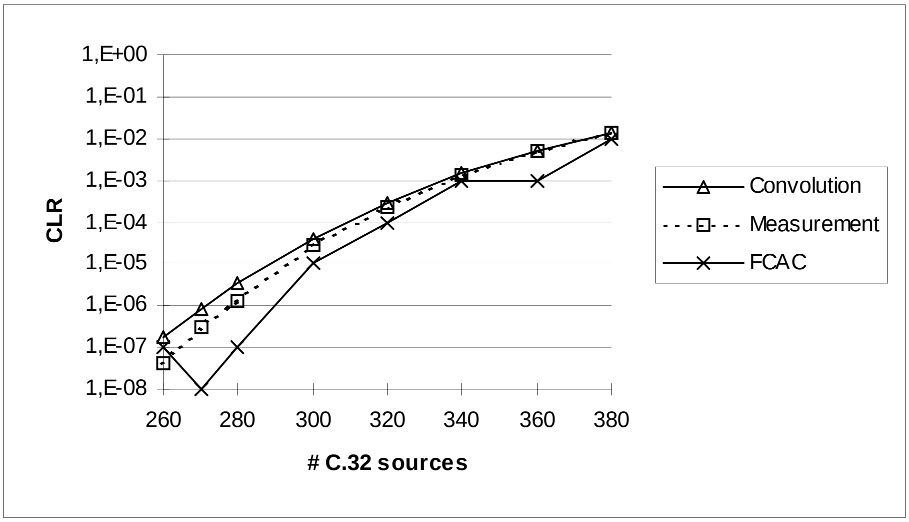


Fig. 6-15 Experiment B6 Cell loss predictions for a traffic mix of C.32 sources

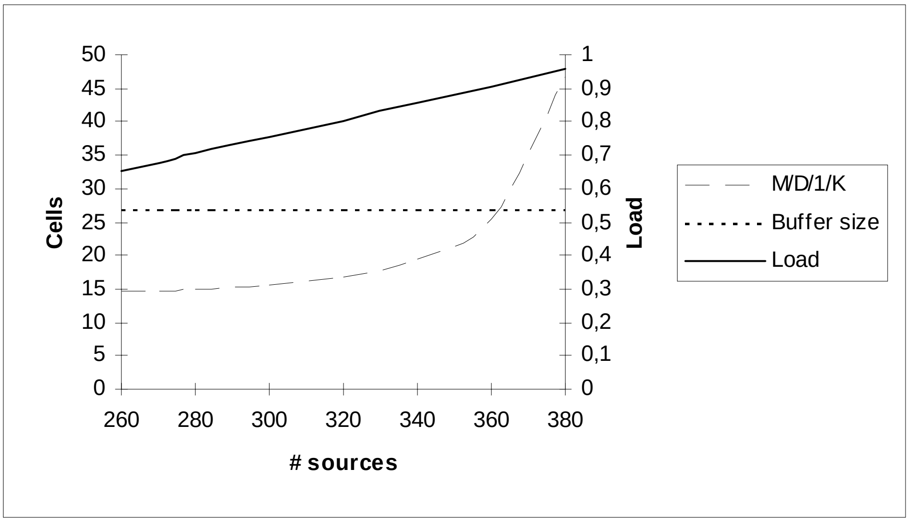


Fig. 6-16 Experiment B6 Load and buffer size against the number of sources

Fig. 6-14 and Fig. 6-16 show that the buffer size evaluated by the M/D/1/K queuing system exceeds the buffer size used for experiments in the test-bed. This occurs if the load of the link is high (almost 100% in both experiments). This means that the load of the link is close to the mean traffic of all sources. This is an extreme experiment with very high load, which is only possible with a large number of sources (115 to 160, and 260 to 380 for experiments B5 and B6 respectively). In this extreme situation the convolution approach becomes optimistic (more than 130 connections in experiment B5.

Heterogeneous scenarios comprising a mix of two types of sources. (test bed + ECA + FCAC)

This set of experiments enables a comparison of the cell loss results obtained from on-line measurements in the Exploit ATM test-bed in Basel [R2061/28] with the cell loss predictions given by both the ECA and FCAC approaches for traffic mixes with two different types of traffic sources multiplexed on a single ATM link.

Experiment	*Background Traffic*	*#*	*Foregound Traffic*	*#*	*load*	*Figure*
B7	A.31	4	B.31	6-24	0.31-0.76	Fig. 6-17
B8	A.31	2	B.31	14-30	0.43-0.83	Fig. 6-18
B9	A.31	4	B.32	20-160	0.21-0.56	Fig. 6-19
B10	A.31	2	B.32	74-230	0.26-0.65	Fig. 6-20
B11	A.31	4	C.31	24-100	0.31-0.78	Fig. 6-21
B12	A.31	2	C.31	70-120	0.51-0.83	Fig. 6-22

Table 6-7 Specification for heterogeneous traffic (measurements)

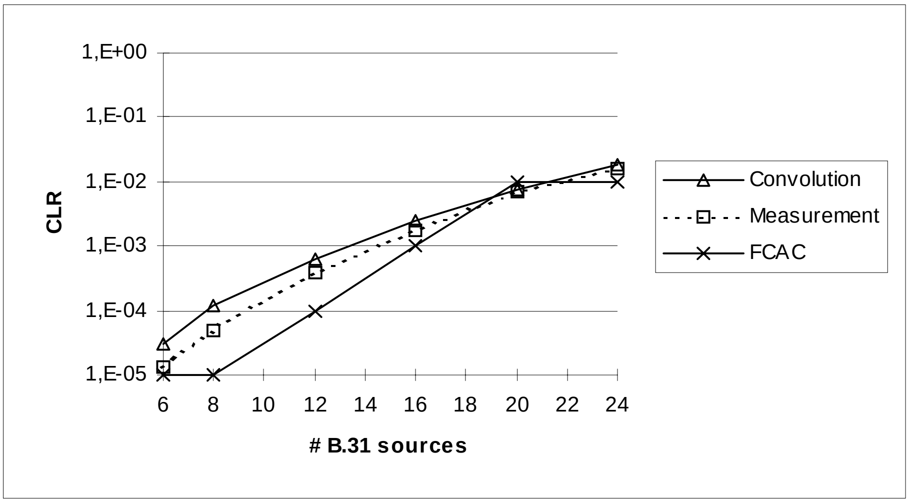


Fig. 6-17 Experiment B7 Scenario composed of 4 A.31 sources and a variable number of B.31 sources

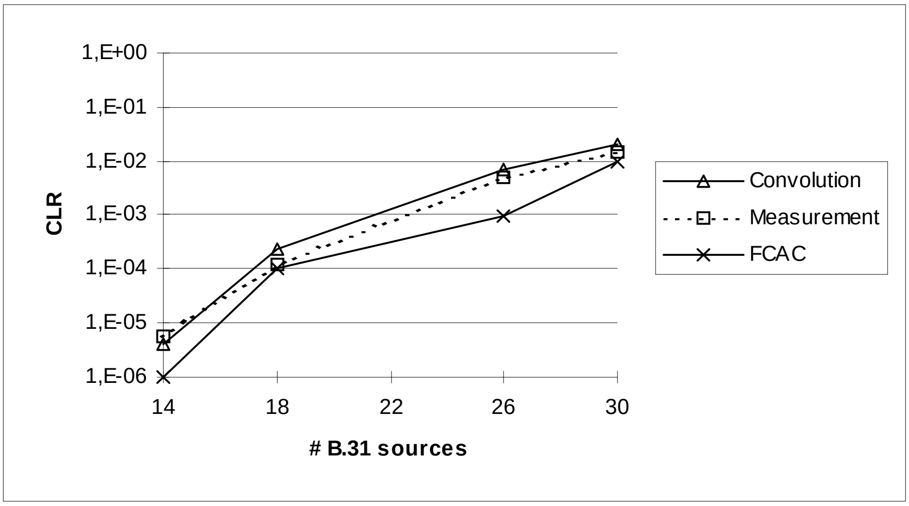


Fig. 6-18 Experiment B8 Scenario composed of 2 A.31 sources and a variable number of B.31 sources

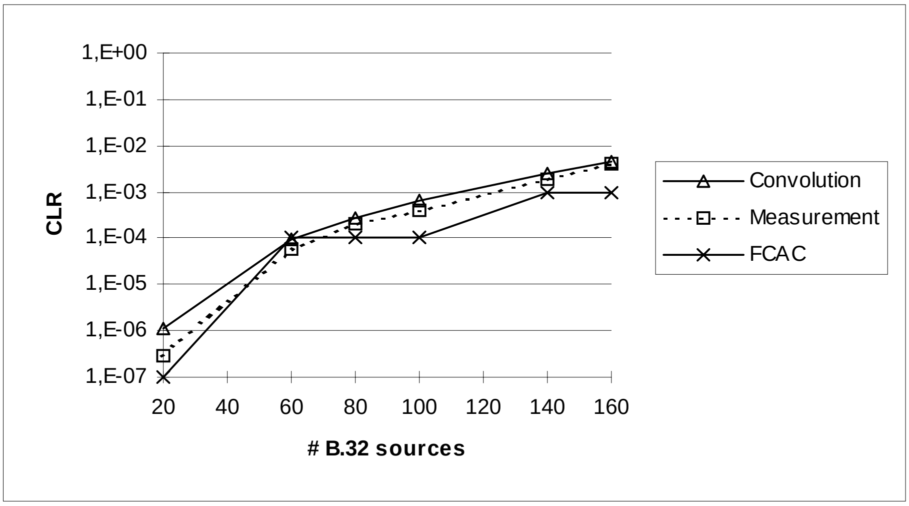


Fig. 6-19 Experiment B9 Cell loss predictions for a traffic scenario composed of 4 A.31 sources and a variable number of B.32 sources

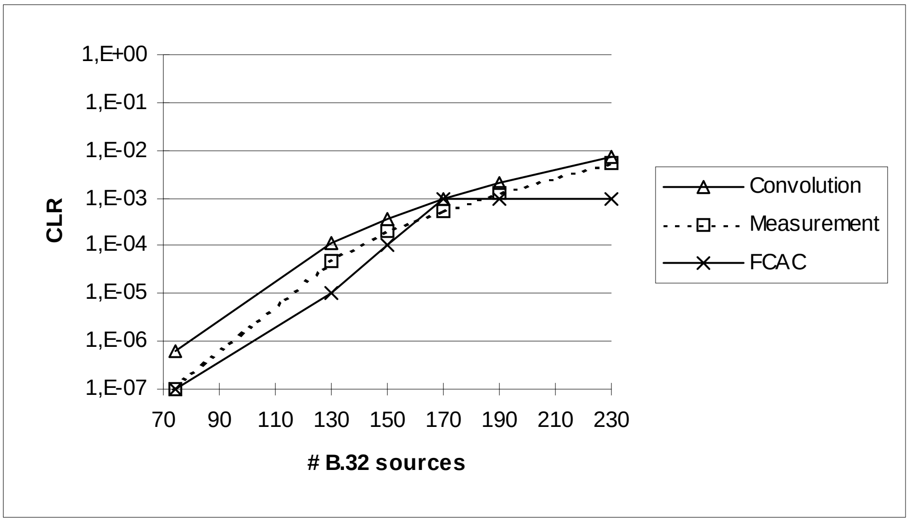


Fig. 6-20 Experiment B10 Scenario composed of 2 A.31 sources and a variable number of B.32 sources

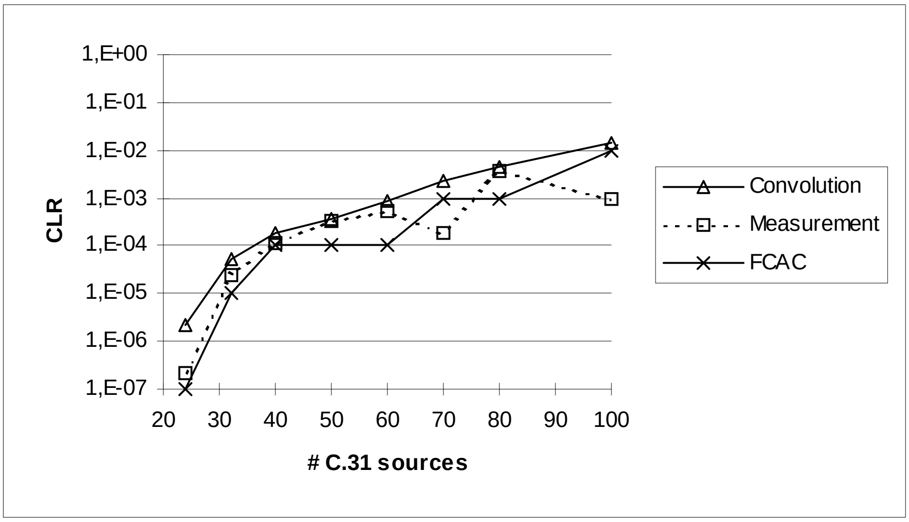


Fig. 6-21 Experiment B11 Scenario composed of 2 A.31 sources and a variable number of C.31 sources

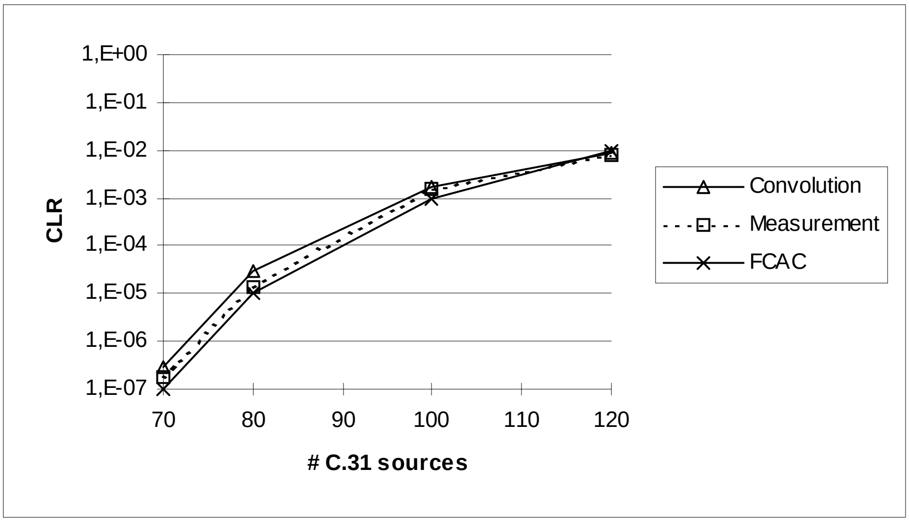


Fig. 6-22 Experiment B12 Scenario composed of 4 A.31 sources and a variable number of C.31 sources

General Comments

From figures Fig. 6-8 to Fig. 6-22 we can see that the cell loss results obtained by ECA are very similar to those obtained by measurements in the Exploit test-bed (ETB). The same can be said for FCAC cell loss predictions.

Considering that the buffer size used in the ATM test-bed experiments is very small (27 cells), it is not surprising that the cell loss predicted by the convolution approach is so close to the cell loss measured values. Convolution algorithms are based on a stationary bit rate distribution and, hence, the source's mean burst length value is ignored. This implies that the cell loss prediction given by ECA is very close to the cell loss obtained for a server system with no buffers. This also explains the slight difference between the measurements curve and the convolution curve as some of the generated cells can be stored in the server output buffer, and, therefore, a more optimistic cell loss is obtained (see measurements curve). The interference between different type of traffic is always determined by ECA.

It can also be observed from figures Fig. 6-8 to Fig. 6-22 that the cell loss prediction given by FCAC is close to the one obtained from measurements. The fact that the cell loss predictions given by FCAC can be more optimistic than predictions given by ECA, is due to the rounding-off errors, this being the closest negative power of ten.

Summarising, the FCAC predictions obtained for each of the fourteen homogeneous and heterogeneous traffic mixes shown previously, reveal: a) the FCAC approximates to the measurements curve; b) the degree of certainty in the prediction increases with the number of examples for the traffic scenario studied; and c) the FCAC prediction is never less optimistic and is generally more optimistic than that for the convolution approach.

6.4 EXPERIMENTS IN SCENARIOS COMPRISING A MIX OF FIVE TYPES OF SOURCES.

The experiments reported in the following were possible, thanks to RACE consortium EXPLOIT by making available the equipment for traffic generation and analysis at the ATM test-bed at Basel for a week of experiments (experiments C1 to C9).

The network configuration is the same as the one used in the experiments of section 6.3 (C = 155.52 Mbit/s and K = 27 cells). The sources used are VBR sources, modelled as On-Off sources with parameters: peak bit rate, mean “on” and “off” periods. The traffic mixes studied are composed of 6 different type of sources (heterogeneous traffic mixes). All the experiments have the same background traffic each composed of the sources described in Table 6-9.

	Traffic Class	*Remarks*
Background Traffic	NTUA-PC 1 NTUA-PC 2 NTUA-PC 3 NTUA-PC 4	One connection for each class Sum of average bit rate 58.27 Mbit/s
Foreground Traffic	ATM100 NTUA-PC 7	Variable Peak Rate & Mean-ON Variable Mean-OFF

Table 6-8 Description of the traffic mixes

Traffic Class	*Peak Rate (Mbit/s)*	*Mean On period (s)*	*Mean Off Period (s)*
NTUA-PC1	37	0.000836	0.00336
NTUA-PC2	37	0.00126	0.00293
NTUA-PC3	37	0.00167	0.006715
NTUA-PC4	37	0.00252	0.005878

Table 6-9 Traffic characteristics of On-Off sources for the background traffic

The foreground traffic changes from one experiment to another and the characteristics of the traffic sources used in each experiment are described in Table 6-10.

After this set of experiments some more simulation runs were performed in order to observe the increase of the cell loss ratio for the multiplexed connections, when one more ATM-100 connection (with the same traffic characteristics as specified for the corresponding experiment) is added, each time, to the background traffic which consists of the traffic mixes studied in experiments C1 to C7.

Limitations of time and the number of connections that can be analysed by the traffic analysers at the ATM test-bed made it impossible to make these measurements on-line and the cell rate simulator, LINKSIM (see [PIT93]) was used instead. Thus, the cell loss ratio shown in figures Fig. 6-23 to Fig. 6-24 for traffic mixes with one ATM100 source is obtained from on-line measurements, whilst the rest were derived from simulation.

These experiments will allow us to understand the influence of a new traffic connection (here the connection associated with the ATM-100 traffic generator) on the worst (maximum) cell loss to be expected for the connections.

Experiment	*NTUA-PC7*	*ATM-100*
C1	peak_rate=37 Mbit/s mean_on= 0.007975 s mean_off = 0.007975 s	peak_rate=8 Mbit/s mean_on= 0.02 s mean_off = 0.08 s
C2	peak_rate=37 Mbit/s mean_on= 0.007975 s mean_off = 0.007975 s	peak_rate=8 Mbit/s mean_on= 0.04 s mean_off = 0.08 s
C3	peak_rate=37 Mbit/s mean_on= 0.007975 s mean_off = 0.009568 s	peak_rate=8 Mbit/s mean_on= 0.02 s mean_off = 0.08 s
C4	peak_rate=37 Mbit/s mean_on= 0.007975 s mean_off = 0.009568 s	peak_rate=8 Mbit/s mean_on= 0.04 s mean_off = 0.08 s
C5	peak_rate=37 Mbit/s mean_on= 0.007975 s mean_off = 0.007975 s	peak_rate=8 Mbit/s mean_on= 0.02 s mean_off = 0.04 s
C6	peak_rate=37 Mbit/s mean_on= 0.007975 s mean_off = 0.007975 s	peak_rate=8 Mbit/s mean_on= 0.06 s mean_off = 0.12 s
C7	peak_rate=37 Mbit/s mean_on= 0.007975 s mean_off = 0.007975 s	peak_rate=8 Mbit/s mean_on= 0.01 s mean_off = 0.02 s

Table 6-10 Traffic characteristics of On-Off sources for the foreground traffic (experiments C1 to C7)

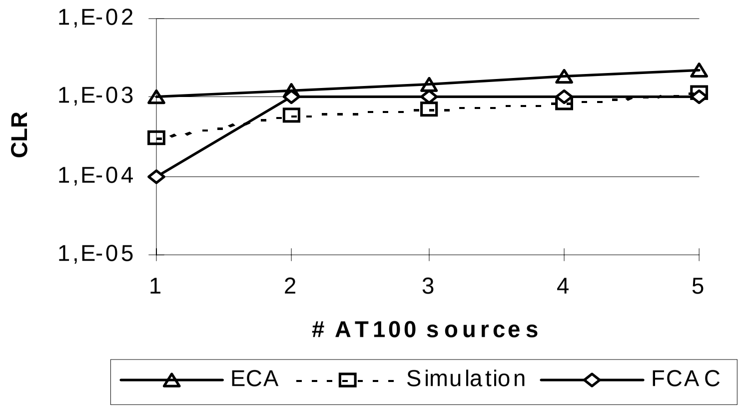
The FCAC cell loss prediction shown in figures Fig. 6-23 to Fig. 6-24 were obtained after tuning the rule base obtained using the Basel data with part of the data set obtained via simulation.

<table>
<tr><td colspan="2" rowspan="2"></td><td colspan="2">ATM100 (Mean-off = 0.08)</td></tr>
<tr><td>Mean-on = 0.02</td><td>Mean-on = 0.04</td></tr>
<tr><td rowspan="2">NTUA-PC7
(Mean-on = 0.007975)</td><td>Mean-off = 0.007975</td><td>Exp. C1</td><td>Exp. C2</td></tr>
<tr><td>Mean-off = 0.009568</td><td>Exp. C3</td><td>Exp. C4</td></tr>
</table>

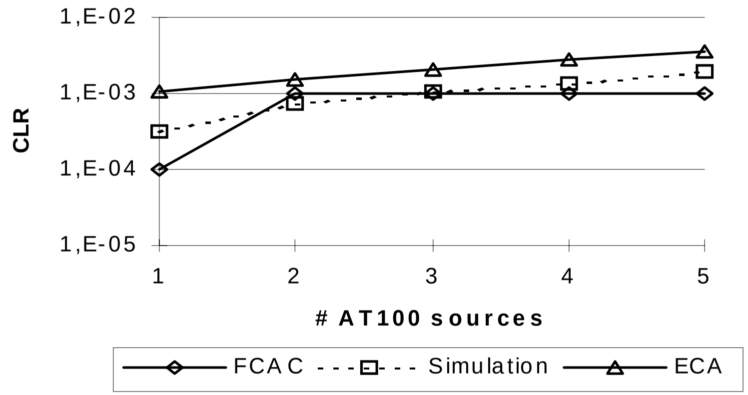
Table 6-11 Structure of experiments C1 to C4 varying the characteristics of both ATM100 and NTUA-PC7

ATM 100 ↑ mean-ON ⇒↑ load

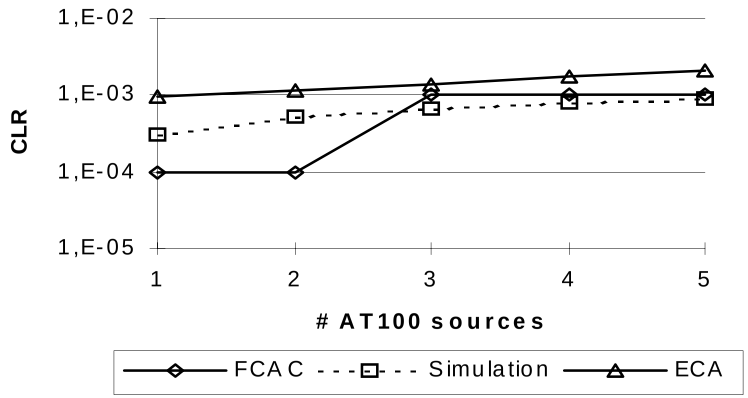
NTUA-PC7
↓ mean-OFF ⇒
↑ load



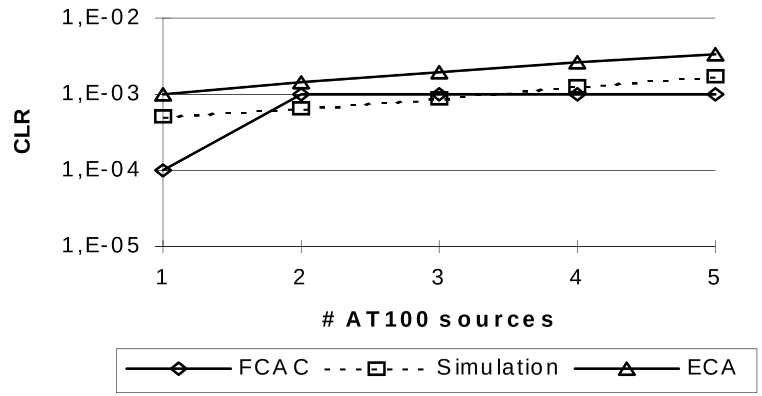
Cell loss Predictions for experiment C3.



Cell loss Predictions for experiment C4.



Cell loss Predictions for experiment C1.



Cell loss Predictions for experiment C2.

Fig. 6-23 Cell loss predictions for experiments C1 to C4

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Experiment	*Peak*	*Mean-on (s)*	*Mean-off (s)*
C7	8	0.01	0.02
C5	8	0.02	0.04
C2	8	0.04	0.08
C6	8	0.06	0.12

Table 6-12 Structure of experiments C2, C5, C6 and C7 ; the mean load is kept constant the mean on and off periods of the ATM-100 traffic source vary.

The results shown in Table 6-12 represent a comparison of experiments C2, C5, C6 and C7. For this set of experiments, the mean-on and off periods of the ATM100 traffic source varies whilst the mean on to mean off ratio is kept constant and equal to 2 for all experiments.

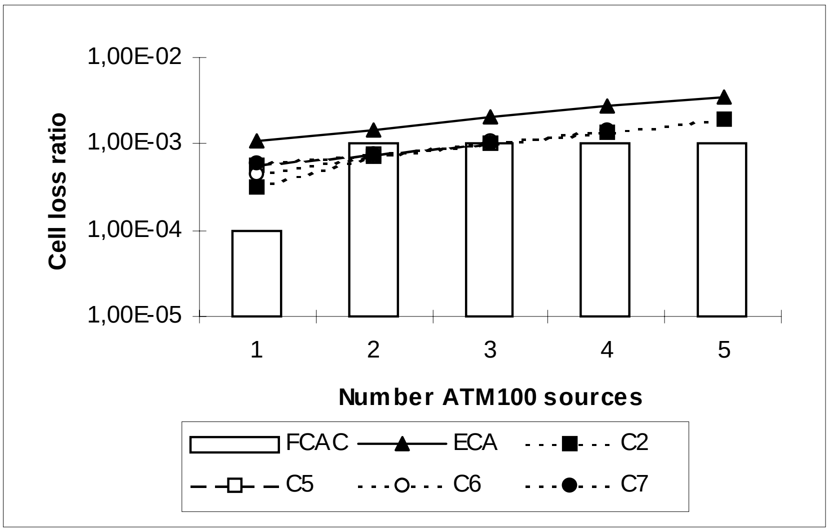


Fig. 6-24 Comparative Cell loss Predictions for experiments C2, C5, C6 and C7.

General comments

All experiments have been carried out by adding one more source to existing traffic and re-evaluating the cell loss ratio. When a fixed cell loss ratio (about 10^{-4}) is achieved the process is stopped. Experiments C1 to C4 show the cell loss ratio corresponding to variations of the mean burst length for the foreground traffic (ATM100 and NTUA-PC7), as described in Table 6-11. As expected, cell losses increase either because the mean-on period of the AT100 has increased or because the mean OFF period of the NTUA-PC7 has decreased (see also Fig. 6-23). The ECA prediction is conservative and follows the CLR behaviour obtained using the simulator. FCAC gives the same cell loss prediction for experiments C1, C2 and C4. In experiment C3, FCAC allows acceptance of one more ATM100 source for a cell loss requirement of 10^{-4}, which is as expected since experiment D3 is characterised by having the lowest load in the link of all four experiments in Table 6-11.

Finally, the effect of variations of the mean burst length, keeping the mean load constant, is analysed in experiments C2, C5, C6 and C7. The mean-on and mean-off periods of ATM100 traffic vary according to Table 6-12, maintaining the mean rate of the source constant. From Fig. 6-24, it can be observed that the duration of the mean-on and off periods have a negligible effect on the resulting cell loss ratio, for these traffic mixes. For traffic mixes with only one ATM100 source, the cell loss ratios differ slightly with variations in the duration of the bursts, as shown in Fig. 6-24.

The ECA prediction, being insensitive to variations of the size of the bursts, gives the same cell loss prediction for the four experiments. The prediction is conservative and this is due to the small number of sources and to the cell level contention carried out in the output buffer. Note that mean ON period for the aggregated traffic is relatively small. FCAC is sensitive to change in the size of the bursts but for the traffic scenario of experiments C2, C5 to C7, the influence of varying the mean burst length on the cell loss ratio is not sufficient to cause a different prediction pattern when increasing the number of ATM100 sources in the traffic mix.

6.4.2 Conclusions

The prediction results based on the convolution approach have been obtained using the ECA algorithm without considering optimisations. The cut-off mechanisms and other improvements biased to achieve a fast evaluation of the CLR can only be used for CAC. The presented experiments always evaluate the CLP for a given scenario. When ECA is used specifically for CAC, a only simple response (YES or NO) is needed and that enables such improvements. Anyway, for the experiments presented, the time required for the evaluation was negligible.

The crisp fuzzy set used for the output variable resulted in integer coefficients for the output cell loss ratios. A bell-shape fuzzy output set would give fractional coefficients for the cell loss ratios but would require an algorithm which is more complex and takes longer to compute than the algorithm used in the present study.

The experiments described in this section compare the cell loss ratios obtained by measurements on an ATM test-bed and using a cell rate simulator with cell loss ratios predicted using CAC approaches based on the convolution algorithm (ECA) and on a fuzzy-based decision support system (FCAC).

The cell loss predictions obtained by FCAC and ECA measurement are in agreement with the cell loss reference values obtained via on-line measurements and using simulations, although the cell loss prediction given by ECA is generally conservative and the FCAC prediction is generally optimistic. In addition, the fuzzy results have associated degrees of certainty which increase with the number of fuzzy rules available to support the particular results.

The ATM test-bed uses a small buffer size of 27 cells, while the convolution approach assumes a buffer-less system.

6.5 EXPERIMENTS OF BANDWIDTH ALLOCATION BASED ON ANALYSIS

6.5.1 Objectives

In the next stage of the work, experiments have been carried out in order to compare the behaviour of different bandwidth allocation approaches for a set of scenarios. Fluid-flow, Linear Approximation and Gaussian are contrasted to the Convolution Approach.

The Demanded Bandwidth is evaluated for the different methods studied and it is compared in the following figures. Those figures show the Demanded Bandwidth required by a pre-fixed set of sources by limiting their probability of congestion (PC).

Scenario	Experiment	Comparison	Figure
large buffers to buffer-less	D1	Stationary vs. Fluid Flow	Fig. 6-25
(high to low) utilisation	D2, D3, D4, D5	Gaussian vs. Convolution	Fig. 6-26 - Fig. 6-29
(two to multi) state sources	D6, D7	Gaussian vs. Convolution	Fig. 6-30 - Fig. 6-31
Homogenous vs. Heterogeneous	D8, D9	Gauss/Linear/ Convolution	Fig. 6-32 - Fig. 6-33

Table 6-13 Scenarios for experiments D1 to D9.

6.5.2 Other studied Methods

Convolution results are compared to Linear, Fluid-Flux and Gaussian approximations. These methods have been presented in Section 2.6.2. They have been evaluated in the following manner:

a) <u>Linear method</u>: each source has an associated effective bandwidth. This depends on the traffic characteristics of the source, the link capacity **C** and the CLR admitted. In this work, an effective bandwidth for each type of call is evaluate. First, homogenous traffic we assume and the maximum number of sources that the link can transport **Nj** is evaluated by an exact method. The effective bandwidth is C/Nj for the j-type sources. As similar procedure is applied to the remaining types.

b) Fluid-Flow model: the bit rate generated by a number of multiplexed connections is represented as a continuous flow of bits. The intensity of this flow is evaluated according to the state of a Markov chain. An approximation presented in [GUE91] is used. This approximation has good accuracy when either the number of connections is small or the actual total equivalent capacity is reasonably close to the overall mean rate.

c) Gaussian allocation approach: this method is based on the assumption that the distribution of the required bandwidth of the existing calls can be approximated by a normal distribution with the same mean and variance. That allows the use of standard approximations to estimate the tail of the bit rate distribution. Formula (6) has been used for this approach.

d) Convolution Approach. An exact evaluation of the instantaneous rate on the link is obtained. Utilising the buffer-less assumption, Convolution is used for Bandwidth Allocation. In section 3, a detailed study has been presented.

All methods calculate a demanded bandwidth in order to ensure a pre-set upper bound for cell loss ratio.

6.6 HOMOGENEOUS (TWO STATE) TRAFFIC EXPERIMENTS

For this experiments a set of 50 On-Off sources has been analysed. These sources have a mean burst period equal to 100 ms; and the maximum PC allowed is10^{-5}.

Experiment D1

The figure over shows the demanded bandwidth (y-axis) evaluated by both: Fluid-Flow and Convolution. The number of sources is 50; and each connection has a peak rate equal to 4 Mb/s The utilisation of each source varies from 10% to 80% (x-axis). The fluid-flow model is evaluated for different buffer sizes (b = 0, 1, 2 and 3 Mbits), b = 0 means in fact small buffers, no differences have been obtained for buffer size up to 128 cells. A new set of values are appended to the graph in order to compare the convolution approach with the results obtained by the Fluid-Flow model. This experiment corresponds to Fig. 4 in [GUE91].

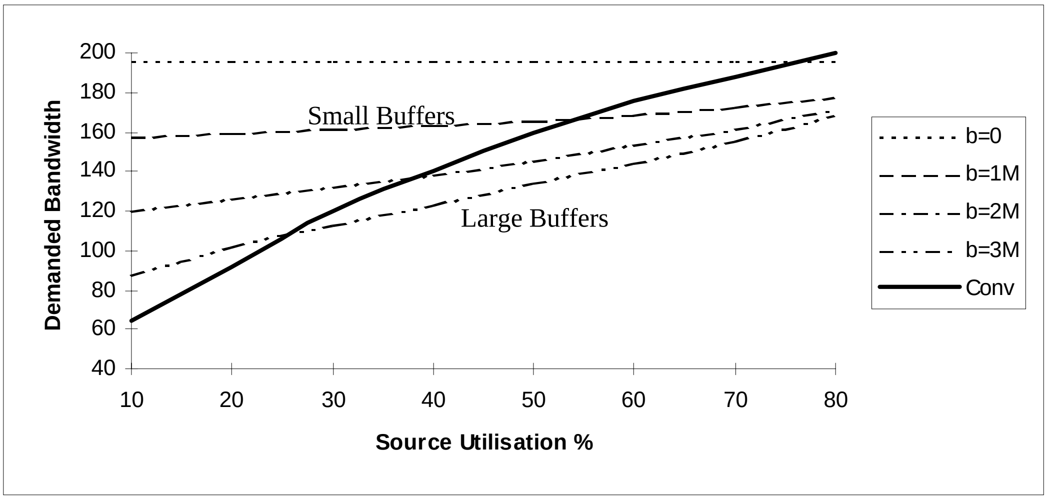


Fig. 6-25 Experiment D1 Fluid-Flow model versus Convolution approach.

For source utilisation higher than 25 %, the required capacity evaluated by convolution is greater than the required for the Fluid-Flow approximation with a buffer size of 3 Mbits. Furthermore, when the size of the buffer is 1 Mbit, the same effect occurs when the source utilisation is higher than 55 %. When using small buffers, the convolution approach always gives more accurate results. Note that small buffers are used in our study to limit maximum cell delay and jitter. Consequently, Fluid-Flow approximations will not be taken into account in the following experiments.

6.7 HOMOGENEOUS (TWO STATE) TRAFFIC EXPERIMENTS (GAUSSIAN VS. CONVOLUTION)

For experiment D2, which is the same scenario defined for experiment D1,a set of homogeneous sources with a of mean burst period 100ms, are analysed for the small buffer case. The demanded bandwidth is presented by varying the total number of sources for both the Gaussian approximation and Convolution approach.

The scenario for experiments D3, D4 and D5, involve: On-Off sources, the mean burst length is not considered (buffer-less model), source utilisation 80 %, 20 % and 10 %, the maximum Probability of Congestion is set to 10^{-5}.

Experiment D2

The next figure shows the demanded bandwidth (y-axis) evaluated by a Gaussian approximation and Convolution. The utilisation of each source varies from 10% to 80% (x-axis).

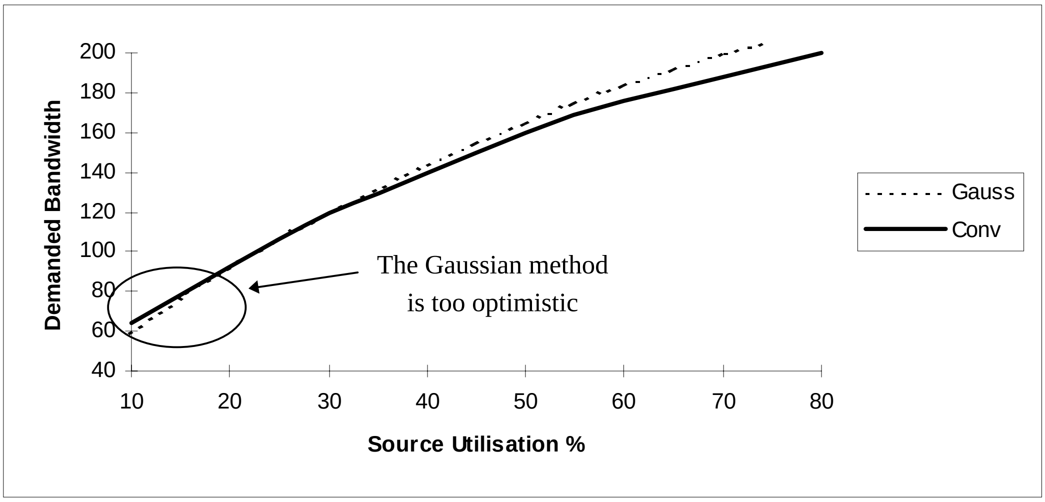


Fig. 6-26 Experiment D2 Gaussian versus Convolution

The required capacity evaluated by convolution is normally less than that required for the Gaussian assumption. For low utilisation, less than 20%, the Gaussian approximation gives a smaller demanded Bandwidth than the Gaussian approximation. But, the convolution approach gives always the most accurate results because of the Gaussian approximation is a simplification of Convolution, and the Gaussian approximation is a bit optimistic. Guerin et al. propose using Fluid-Flow approximations in this situation, but for small buffers the required bandwidth is perceptively higher than the Gaussian. Moreover, it is not clear how to define this situation [GUE91].

Experiment D3

The following set of experiments shows the demanded bandwidth evaluated by the Gaussian and the Convolution approaches for different number of homogeneous sources, three figures will be shown for 80, 20 and 10 % of source utilisation (low to high burstiness). The right axis displays the demanded bandwidth and the left axis displays the difference between both curves expressed a percentages.

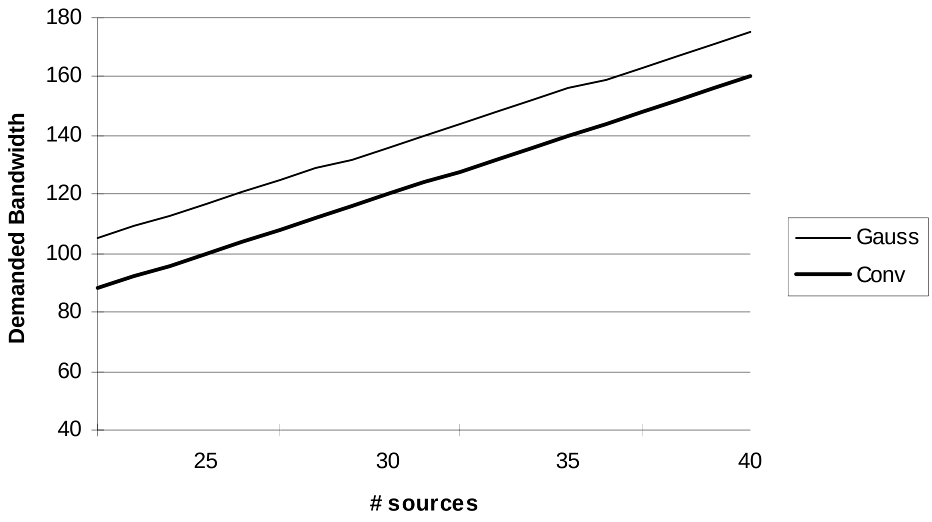


Fig. 6-27 Experiment D3 80% source utilisation

This figure shows that the convolution approach always evaluates a more accurate demanded Bandwidth than the Gaussian model. From 10 % to 40 % over-estimation is observed with the Gaussian model compared to Convolution. Therefore, the Gaussian approximation is too pessimistic in this traffic scenario.

Experiment D4

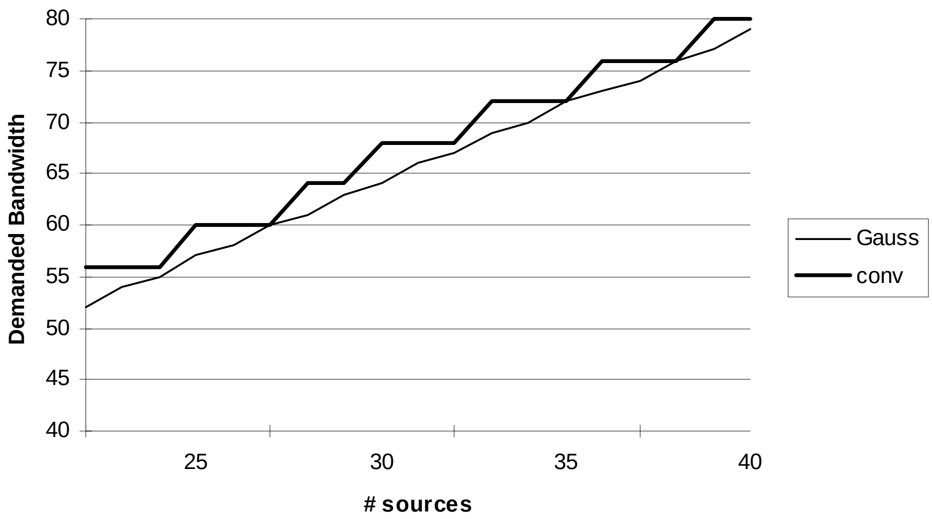


Fig. 6-28 Experiment D4 20% source utilisation

In this intermediate scenario, more accurate results have been obtained for an utilisation of 20 %. The differences between both methods are not significant.

Experiment D5

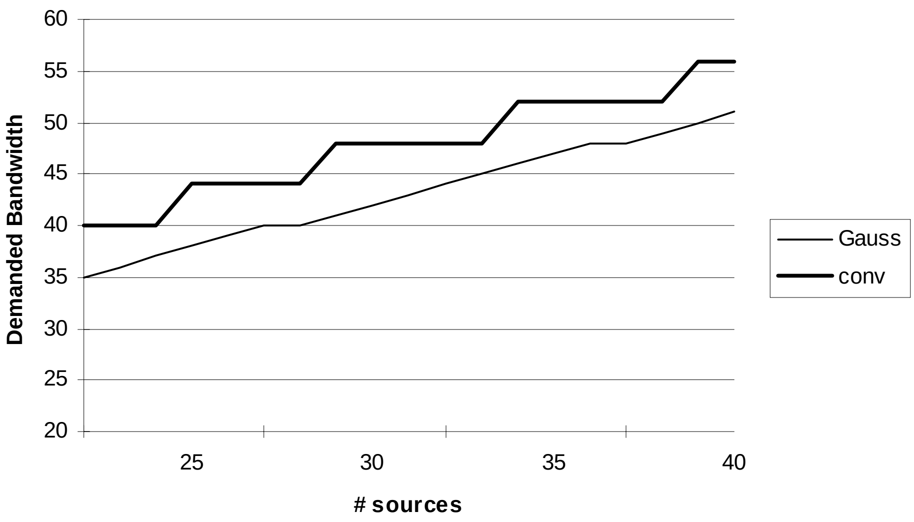


Fig. 6-29 Experiment D5, 10% source utilisation

In presence of bursty traffic it is interesting to note that the Gaussian evaluation is too optimistic (up to 20 % corresponding to 6 to 8 sources, and it remains about 10 % for more connections). This effect is unwanted as the actual cell loss may be greater than the evaluated cell loss ratio and consequently the established QOS requirements cannot be fulfilled.

6.8 HOMOGENEOUS (THREE STATE) TRAFFIC EXPERIMENTS (GAUSSIAN VS. CONVOLUTION)

In last experiments, it has been demonstrated, that the behaviour of the Gaussian versus the Convolution Approach varies depending on the utilisation of the sources. Low utilisation may cause optimistic results and vice versa. In order to study this problem new scenarios with three-state sources are analysed.

The source characterisation has been chosen in the GMDP model. The following table shows the characteristics of the used sources.

	State	*0*	*1*	*2*
A2	bit rate (Mbit/s)	0.4	2	
	probability	*0.625*	*0.375*	
A3	bit rate (Mbit/s)	0.2	0.4	2
	probability	*0.7*	*0.2*	*0.1*
B2	bit rate (Mbit/s)	2	10	
	probability	*0.625*	*0.375*	
B3	bit rate (Mbit/s)	0.3	1.375	10
	probability	*0.75*	*0.2*	*0.05*
C2	bit rate (Mbit/s)	6	30	
	probability	*0.625*	*0.375*	
C3	bit rate (Mbit/s)	3	6	30
	probability	*0.7*	*0.2*	*0.1*

Table 6-14 Source description

Results of the experiment D6

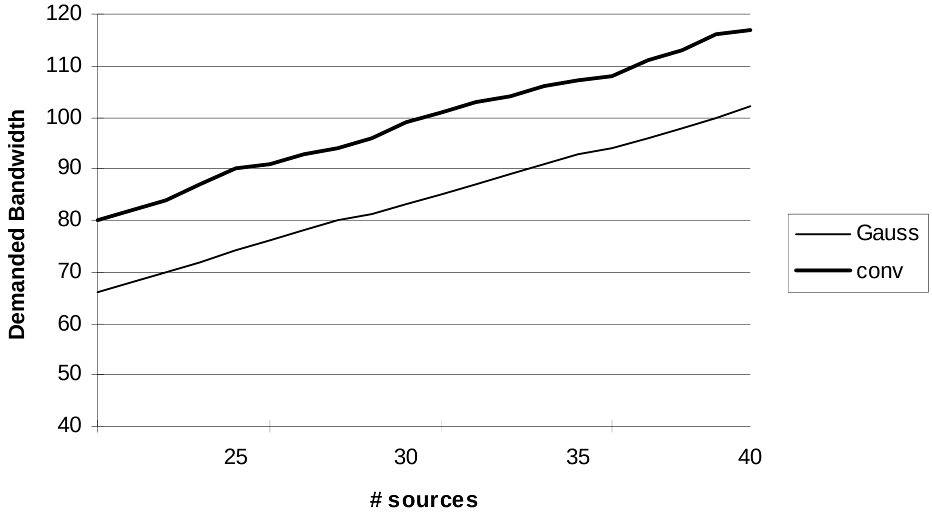


Fig. 6-30 Experiment D6 B3 traffic

As the above figure shows, for a small number of sources the difference between both approximations is about 25 %. For a greater number of sources this figure tends to 10%. Again the Gaussian approach is optimistic.

6.9 HETEROGENEOUS TRAFFIC EXPERIMENTS.

Several experiments with mixed traffic are now presented. The mixture corresponds to the GMDP table (table 2). On the y-axis the demanded bandwidth is shown and on the x-axis all the possible combinations of traffic are presented.

Results of the experiment D7

Mixture of two type of sources each having two states.

Type	*B2*	*C2*
max. number of calls	7	42

Table 6-15 Type of sources

Combinations of traffic correspond to 0 C1 sources mixed, with 0 B2 sources up to 42 C1 sources, mixed with 7 B2 sources, e.g. 0,0 0,1 0,2 .... 0, 7 . 1,0 1,1 1,2 .... 1 7 .... 42,0 42,1 42,2 .... 42,7.

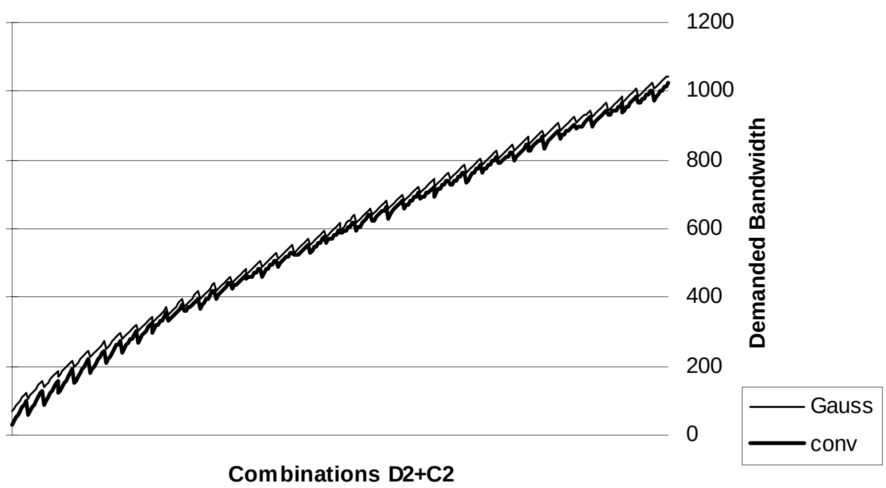


Fig. 6-31 Experiment D7 (two states sources)

In this scenario the Gaussian approximation is also accurate. Fig. 6-31 shows this clearly except for combinations of traffic that correspond to a small number of connections.

Mixture of three types of sources having all two states.

Type	*A2*	*B2*	*C2*
max number of calls	15	10	3

Table 6-16 Experiment 7 type of sources.

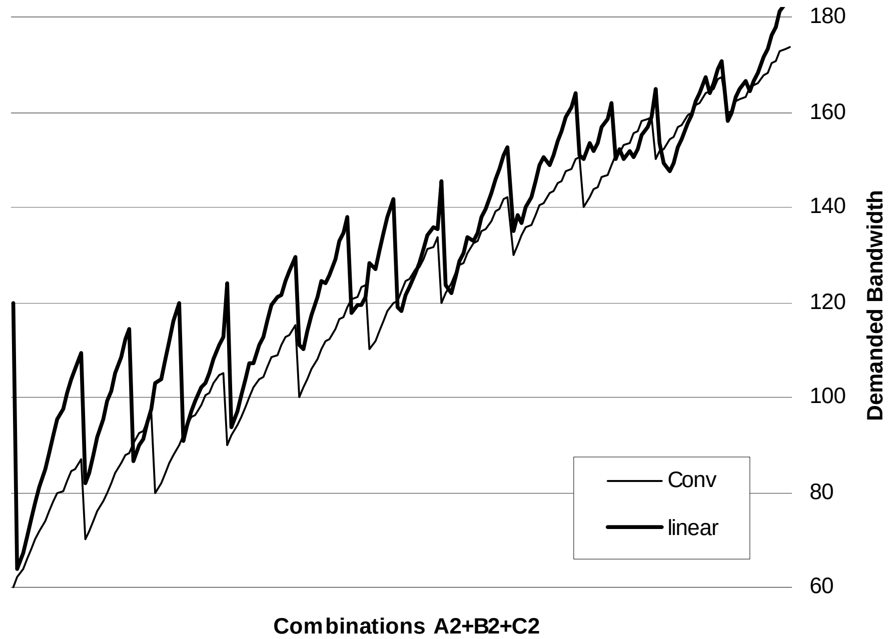


Fig. 6-32 Experiments D8

For simplicity, only combinations with one C3 connection and varying B2 and A2 types of traffic, as shown in Table 6-15, are plotted, since the remaining combinations show similar behaviour. The linear approximation has a changing behaviour that tends to be conservative. Differences up to 15% can be observed in the diagram.

Results of the experiment D9

In this scenario a mixture of three type of sources is considered, as the previous experiment. However, now all type of source now have three states, thus increasing the heterogeneity.

Type	*A3*	*B3*	*C3*
max number of calls	15	10	3

Table 6-17 Type of sources

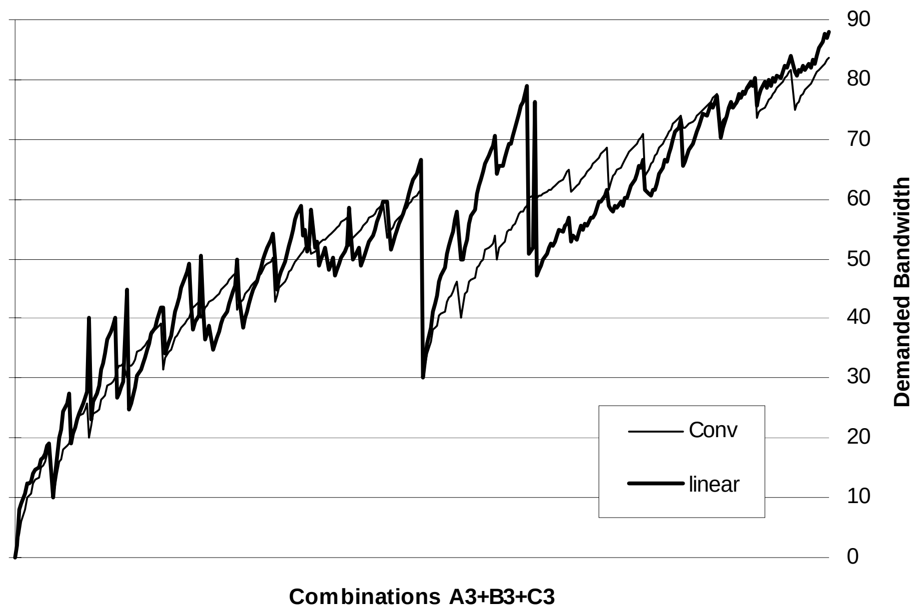


Fig. 6-33 Experiment D9 (0 and one C3 sources)

In the experiment D9 two different areas can be observed. First, Fig. 6-33 shows a changing behaviour in which it is not clear whether the linear approximation is conservative or optimistic.

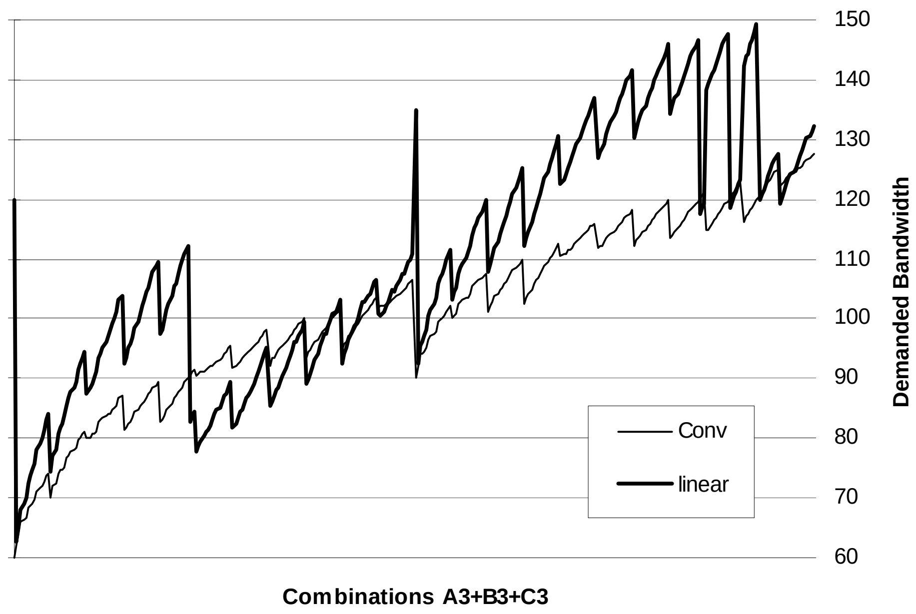


Fig. 6-34 Experiment D9 (2 and 3 C3 sources)

Finally, as Fig. 6-34 shows, the linear approach gives a generally optimistic evaluation of the demanded bandwidth (up to 20% less).

6.10 CONCLUSIONS

Using a small buffer, the convolution approach gives always more accurate results than Fluid Flow approximations; small buffers being a premise in this study.

The required capacity evaluated by convolution is normally less than the one required for the Gaussian assumption. But, the Gaussian approximation is a bit optimistic for low utilisations. In the presence of bursty traffic, the Gaussian evaluation is also too optimistic. This effect is unwanted because the actual cell loss may be greater than the evaluated cell loss ratio and, consequently, the QOS requirements established for the user could not guaranteed.

Finally, the linear approximation varies from pessimistic to optimistic depending on the mixture of traffic. Therefore, the linear approximation is not a robust method.

7. Conclusions and Future Research

7.1 CONCLUSIONS

Probability of congestion / buffer-size and burst length aspects

In this work the utilisation of the Probability of Congestion (PC) as a bandwidth decision parameter has been presented. The validity of PC utilisation is compared with QOS parameters in buffer-less environments when only the CLR parameter is relevant.

It can be summarised that the convolution algorithm seems to be a good solution for CAC in ATM networks with relatively small buffers. If the source characteristics are known actual cell loss ratio can be very well estimated. Furthermore, this estimate is always conservative, allowing the retention of the network performance guarantees.

Several experiments have been carried out and investigated to explain the deviation between the proposed method and the simulation. Time parameters for burst length and different buffer sizes have been considered. Experiments to confine the limits of the burst length with respect to the buffer size conclude that a minimum buffer size is necessary to achieve adequate cell contention. Normally, small values for the buffer size are involved. Note that delay (due to propagation) is a no dismish limit for long distance and interactive communications, then small buffer must be used in order to minimise delay. On the other hand, by increasing the buffer size the CLR provided by analysis becomes too conservative. Then inaccurate results are obtained by analysis, which occurs when the mean burst length is similar to the buffer size.

The proposal: The enhanced convolution approach.

To overcome the drawbacks of the formula-based Convolution Approach, a new method of evaluation is analysed: the Enhanced Convolution Approach (ECA). Traffic is grouped in classes of identical parameters.

By using the multinomial distribution function instead of the formula-based convolution, a partial state corresponding to each class of traffic is obtained. Finally, the global state probabilities are evaluated by *multi-convolution* of the partial results. This method avoids accumulated calculations and saves storage requirements, specially in complex scenarios.

The ECA also computes the Individual Cell Loss Ratio (CLR_j) for each j-class of traffic. Based on the ECA algorithm, a expression for the CLR_j evaluation has been presented.

Cost and implementation issues

Sorting is the dominant factor for the formula-based convolution, whereas cost evaluation is the dominant factor for the enhanced convolution. Normally, the formula-based convolution needs to convolute a great number of elements, once a new connection is added a sorting and compacting process is carried out. On the other hand, the enhanced convolution approach needs more previous calculations; as more compacted vectors are obtained, as we expect via the multinomial distribution function, and these vectors correspond to each class, the amount of elements obtained is generally small.

With reference to the cut-off mechanisms presented, the major conclusion is the efficacy of the low probability cut-off. The first direct implication is the reduction in the storage requirements. Moreover, this reduction in the intermediate vectors implies an immediately rise in the evaluation time.

In presence of homogeneous traffic the applying only the ECA immediately increases the speed-up factor. However, the number of elements is the same for both methods as expected.

For heterogeneous traffic, sorting cost for formula-based convolution is again the predominant factor. This cost is specially large in complex scenarios such as heterogeneous and 3-state traffic. However, the dominant factor for ECA is the calculation cost. With a negligible reduction of accuracy, by using small probability cut-off, a great improvement in terms of both storage requirements and calculation cost is obtained.

When the number of connections for the different classes of traffic is not balanced, the ECA is faster than the formula-based convolution. This assumption is reasonable because in a link with heterogeneous traffic, background traffic could normally be constituted by a great number of voice (or similar) connections, and other consumer bandwidth connections (such as video sources) are relatively small in quantity.

We can conclude that by combining the ECA method with cut-off mechanisms, utilisation of ECA in real-time CAC environments as a single level scheme is always possible.

Bandwidth allocation experiments

The prediction results based on the convolution approach have been obtained using the ECA algorithm without considering optimisations. The cut-off mechanisms an other improvements oriented to achieve a fast evaluation of the CLR can only be used for CAC. The presented experiments always evaluate the CLP for a give scenario. When

ECA is used specifically for CAC a simple response (YES or NO) is needed that allows the utilisation of those improvements.

The cell loss predictions obtained by FCAC and ECA measurement are in agreement with the cell loss reference values obtained via on-line measurements and using simulations, although the cell loss prediction given by ECA is generally conservative and the FCAC prediction is generally optimistic. In addition, the fuzzy results have associated degrees of certainty which increase with the number of fuzzy rules available to support the particular results.

The ATM test-bed uses a small buffer size of 27 cells, while the convolution approach assumes a buffer-less system. Using small buffer, the convolution approach always gives more accurate results than Fluid Flow approximations; small buffers are a premise in our study.

The required capacity evaluated by convolution is normally less than that required for the Gaussian assumption. However, in the presence of bursty traffic the Gaussian evaluation is also too optimistic.. The linear approximation varies from pessimistic to optimistic depending on the mixture of traffic. Therefore, the linear approximation is not a robust method. The optimistic evaluation of cell loss is unwanted. If the cell loss is grater than the predicted cell loss the QOS requirements established for user are not satisfied

7.2 FUTURE RESEARCH

In this section future work towards of the complexity reduction and determination of the range of validity of the Enhanced Convolution Approach (ECA) is set out. Finally, the source modelling for more realistic environments is introduced.

Complexity Reduction

The reduction of the complexity in the evaluation of the ECA can be focussed in several directions. First, by optimising the obtention of the state-matrix corresponding to each class of traffic. The partial sorting cut-off mechanism, which is presented in section 5.2.1 can be the starting point.

Another improvement in ECA evaluation is obtained with Monte Carlo techniques. When the number of connections is very large, the number of possible states to be analysed leads to excessive calculation time and memeory requirements. However, the total information of the state of the system is sparse on the huge sub-matrix corresponding to each class of traffic. An investigation method based on the Monte Carlo can be applied; the sub-matrices are *multi-convoluted* at random in this case. When the convergence of the partial evaluation of the PC achieves enough accuracy the process is stopped.

By analysis of the ECA evaluation (see Fig. 4-3 Overview of the method.), a parallelisation of the ECA algorithm is clearly shown. First, the evaluation of each sub-matrix corresponding to each class of traffic can be obtained separately. Secondly, the *multi-convolution* can obtain partial results simultaneously; in the extreme case, given one element of one sub-matrix for each processor which investigates all the elements of the other sub-matrices.

The results obtained so far lead to reasonable expectation of achieving a real time CAC based on the ECA method. Moreover, the implementation of ECA as a first level in CAC would be posssible. In this situation a dynamic allocation of the bandwidth associated with a VP would be managed immediatly.

The Range of Validity

The range of validity could be determined in two directions: analysis or simulation. The analytical methods have difficulties to obtain temporary distributions in complex scenarios. What is not clear is whether analytical methods obtain results in complex scenarios, specifically for non On-Off sources.

To extend the validity of the method, more simulation with more complex scenarios are needed; increasing the heterogeneity may also be productive. A large number of connections with more sophisticated traffic and, more than three states emmiting in an ATM link have to be analysed.

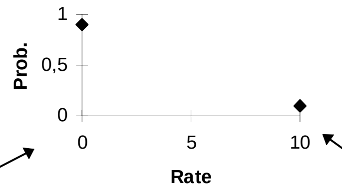
Source modelling

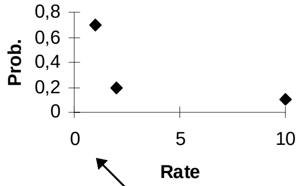
Source modelling for more realistic traffic is now an open issue. The ECA utilisation does not take account of temporal references (burst length), so that the source parametrisation is simplified. On the other hand, more reallistic traffic, such as SBR video sources, can be modelled as sources with more than two associated states. However, the drawback is that no police function is available when sources are not On-Off.

8. APPENDIX

8.1 APPENDIX A. EXAMPLE OF EVALUATION BASED ON FORMULA.

This Section shows a detailed evaluation of convolution based on formula (27). Three sources are multiplexed in a single link. The statistical distribution for the offered rate is evaluated





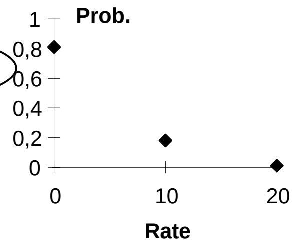
	Rate	Prob
a	0	0,9
b	10	0,1

	Rate	Prob
i	0	0,9
ii	10	0,1

	Rate	Prob
>	1	0,7
>>	2	0,2
>>>	10	0,1

	Rate	Prob.	Rate	Prob.	Rate	Prob.
a,i	0	0,9	0	0,9	0	0,81
b,i	0	0,9	10	0,1	10	0,09
a,ii	10	0,1	0	0,9	10	0,09
b,ii	10	0,1	10	0,1	20	0,01

Same rate = 10

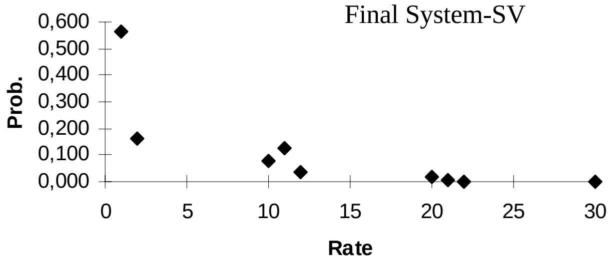


	Rate	Prob
/	0	0,81
//	10	0,18
///	20	0,01

Intermediate System-SV

		Rate	Prob.	Rate	Prob.	Rate	Prob.
/	>	0	0,81	1	0,7	1	0,567
/	>>	0	0,81	2	0,2	2	0,162
/	>>>	0	0,81	10	0,1	10	0,081
//	>	10	0,18	1	0,7	11	0,126
//	>>	10	0,18	2	0,2	12	0,036
//	>>>	10	0,18	10	0,1	20	0,018
///	>	20	0,01	1	0,7	21	0,007
///	>>	20	0,01	2	0,2	22	0,002
///	>>>	20	0,01	10	0,1	30	0,001

Final System-SV



The PC varies according to the capacity of the link. In this example, it is easy to see that the PC will be 0.001 if the capacity **C** is set to 25. This value corresponds to the probability of emitting at an instantaneous rate equal 30, since only for this rate the link is in congestion. If C is set to 15, four items on the System-SV are adding corresponding to the rates 20, 21, 22 and 30. The amount of PC now is 0.018 + 0.007 + 0.002 + 0.001 = 0.028. Note that PC would be the same for any C between 13 to 19, because there are no items in this segment.

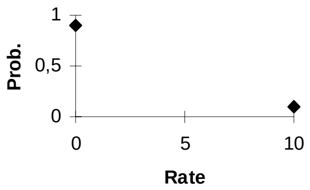
Using formulas (8) and (49) the CLR can be obtained. For a given capacity equal to 25, the point corresponding to rate equal to 30 is used in the following expression:

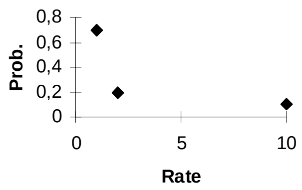
$$CLR(Y) = \frac{\sum_{L>C}(L-C)P(Y=L)}{E(Y)} = \frac{\sum_{L>C}(30-25)P(Y=25)}{E(Y)} = \frac{5 \bullet 0.001}{4.1} = 0.00122$$

The PC for this situation is 0.001, corresponding to P(Y = 30). Therefore, the CLR evaluation is based on the PC distribution.

8.2 APPENDIX B. EXAMPLE OF EVALUATION BASED ON ECA.

This Section shows a detailed evaluation of convolution based on the extended convolution approach. The same scenario showed in Appendix A is used. Three sources are multiplexed in a single link. The statistical distribution for the offered rate is evaluated:





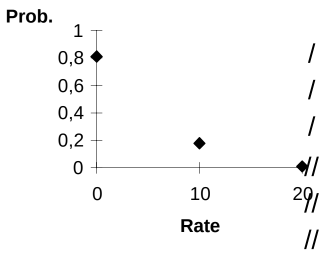
Rate	Prob
0	0,9
10	0,1

Multinomial

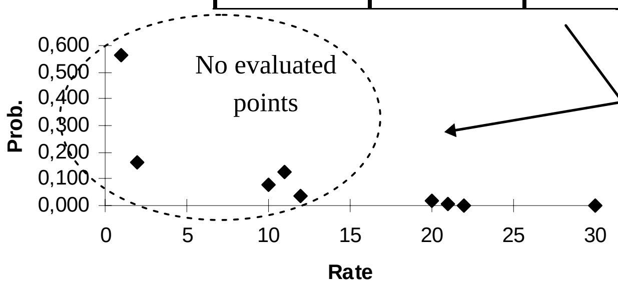
	Rate	Prob
>	1	0,7
>>	2	0,2
>>>	10	0,1

10	0	rate	#	part.	Prob.
0	2	0	1		0.9
1	1	10	2	0.9	+ 0.1
2	0	20	1		0.1

	Rate	Prob
/	0	0,81
//	10	0,18
///	20	0,01



		Rate	Prob.	Rate	Prob.	Rate	Prob.
/	>	0	0,81	1	0,7	1	-
/	>>	0	0,81	2	0,2	2	-
/	>>>	0	0,81	10	0,1	10	-
//	>	10	0,18	1	0,7	11	
//	>>	10	0,18	2	0,2	12	
//	>>>	10	0,18	10	0,1	20	
///	>	20	0,01	1	0,7	21	0,007
///	>>	20	0,01	2	0,2	22	0,002
///	>>>	20	0,01	10	0,1	30	0,001



Obviously, the same results are obtained now.

8.3 APPENDIX C. SOURCE CODE.

In this section the CAC algorithm is described. This a source code written in ANSI C language. The headers, initialisation routines, and well known algorithms are omitted. For simplicity, the code used to evaluate the calculation cost is also omitted.

```
/******************************** CAC_MAIN.C ********************************/
/****************************************************************************/
/*  Parameters
/*  file(.cfg):  contents the descriptions of the scenario (traffic classes)*/
/*  LinkCapacity: Capacity 'C' of the link in Mbit/s                        */
/*  maxPC:        Maximum probability of Congestion cut-off (0 = non used)  */
/*  MinVal:       Minimum value cut-off (this is not showed in this source) */
/*  sizeRed factRed tipeRed : Reduction parameters (see the corresponding   */
/*                            routine                                       */
/* #0sources #1sources ...  number of connections of each type              */
/****************************************************************************/

#define NARGC 8 /* according to the number of parameter used */

main(int argc, char *argv[])
{
    LoadConfiguration(argv[1]);
    LinkCapacity = atoi(argv[3]);
    max_pcong = (float) atof(argv[4]);
    min_val = atof(argv[5]);

    ELEMENTS_REDUCCIO = atoi(argv[6]);
    GRUP = atoi(argv[7]);
    REDUCTION_METOD = atoi(argv[8]);

    SourceTypeNumber=argc-NARGC;
    for (i=0;i < SourceTypeNumber; i++){
     cn[i] =  atoi(argv[i+NARGC]);
    load_factorial();

    ECAprocedure(cn); /* evaluation of PC and CLR using the ECA algorithm */

        /* print cost and time results */
    printf("\n elements%7d", maxEle);
    printf("\n Ksorts   %6.0f ", sorts / 1000);
    printf("\n Kcost    %6.2f  \n", (adds + (float) mults) / 1000);
    return 0;
}
```

```
/****************************** CAC_ECA.C ********************************/
/*  This code implements the ECA algorithm. This is the balanced version    */
/*  with memory storage. For simplicity some parts of the complete code     */
/*  has been omitted. The rutines to obtain individual cell losing are also */
/*  omitted.                                                                */
/***************************************************************************/

/***************************************************************************/
/*    This routine implements the reduction of states.                      */
/*    The reduction_type may be : mean, middle and the maximun rate         */
/*    Example for reduction vars                                            */
/*    REDUCTION_METOD= MEAN_RATE                                            */
/*    ELEMENTS_REDUCCIO=100;                                                */
/*    GRUP=3;                                                               */
/***************************************************************************/
static void     reduction(int reduction_type, int tipus)
{
    int             aux_r[MAXPOS];
    prob_type       aux_p[MAXPOS];
    static int      i, j;
    unsigned int    aux_index = 0;
     /* loading aux structures to reduce */
    memcpy(aux_r, SubMatrixRate[tipus], sizeof(int)*rows[tipus]);
    memcpy(aux_p, SubMatrixProbabilty[tipus], sizeof(prob_type)*rows[tipus]);
    i = 0;
    switch (reduction_type) {
    case MEAN_RATE:
    /* The mean of the involved rates is presented. For maximum and the middle
    element few modifications are need, SubMatrixRate and SubMatrixProbabilty
    store rate and probability for the tipus type of source */
        while (i < rows[tipus]) {
            SubMatrixRate[tipus][aux_index] = aux_r[i];
            SubMatrixProbabilty[tipus][aux_index] = aux_p[i];
            i++;
             /* explores all states to be reduced */
            for (j = 1; j < GRUP && i < rows[tipus]; j++) {
                SubMatrixProbabilty[tipus][aux_index] += aux_p[i];
                SubMatrixRate[tipus][aux_index] += aux_r[i];
                i++;
            }
            SubMatrixRate[tipus][aux_index] /= (int) j; /* mean rate */
            aux_index++;
        }
        break;
rows[tipus] = aux_index;
}
```

```
/*************************************************************************/
/*  This is the source code that evaluates the MDF for each element of      */
/*  the submatrix corresponding to a class of traffic.                      */
/*************************************************************************/
static prob_type calc_p(int num, int source, int v0, int v1, int v2)
/*************************************************************************/
{
    static prob_type ret_val;
      /* this expression evaluate the multinimial distribution function (MDF).
         Note that factorial and the corresponing inverse value have been
         evaluated and stored previously */
    ret_val = factArr[num] * factArrInv[v0] * factArrInv[v1] * factArrInv[v2];
    ret_val *= prExp[source][0][v0];
    /* the SourceStatesNumber variable
    if (SourceStatesNumber[source] >= 2) {
        ret_val *= prExp[source][1][v1];
        if (SourceStatesNumber[source] == 3)
            ret_val *= prExp[source][2][v2];
    }
    return ret_val;
}
/*************************************************************************/
/*  This function generates a submatrix SMX for on type of traffic.         */
/*************************************************************************/
void            upd_smx(int num, int tipus)
/*************************************************************************/
{
   rate_type       st_rate = (rate_type) 0;
    int             var0, var1, var2;
/* the variables cnst2ini and cnst1ini must be initialisaded depending on the
number of states of the traffic class ‘tipus’ */
    init_loop_var(num, tipus);
    indexx = 0;
    for (var2 = cnst2ini; var2 >= 0; var2--) {
        for (var1 = cnst1ini - var2; var1 >= 0; var1--) {
            var0 = num - (var1 + var2);
            st_rate = vp[tipus][0].rate * var0;
            st_rate += vp[tipus][1].rate * var1;
            st_rate += vp[tipus][2].rate * var2;
            work_r[indexx] = st_rate;
            work_p[indexx] = (prob_type) calc_p(num, tipus, var0, var1, var2);
            if (min_val == (double) 0 || (double) work_p[indexx] < min_val)
                indexx++;
        }
    }
}
```

```
/***************************************************************************/
/*                  Probability of congestion (after all updated SMX)       */
/***************************************************************************/
/*  This is the main procedure that investigate all the states of the       */
/*  system. The probability of congestion is evaluated in this procedure    */
/***************************************************************************/
(void)            multi_convolution_evaluation()
{
    int             rate;
    int             i;
    int             il[MXNUMTYPES];

    total_pcong = (prob_type) 0;
    for (il[4] = 0; il[4] < rows[4]; il[4]++) {
        for (il[3] = 0; il[3] < rows[3]; il[3]++) {
            for (il[2] = 0; il[2] < rows[2]; il[2]++) {
                for (il[1] = 0; il[1] < rows[1]; il[1]++) {
                    for (il[0] = 0; il[0] < rows[0]; il[0]++) {
                        /* first the associated rate to a possible state is
                        evaluated */
                        rate = SubMatrixRate[0][il[0]];
                        for (i = 1; i < SourceTypeNumber; i++)
                            rate += SubMatrixRate[i][il[i]];
                        /* if this rate is greater than the Capacity of the
                        link 'C', the analysed state in in congestion */
                        if (rate > LinkCapacity) {
                            partial_pcong = SubMatrixProbabilty[0][il[0]];
                            for (i = 1; i < SourceTypeNumber; i++)
                                 partial_pcong*=SubMatrixProbabilty[i][il[i]];
                            total_pcong += partial_pcong;
                           CLR[MXNUMTYPES]+=(rate-LinkCapacity)*partial_pcong;
                            for (i = 0; i < SourceTypeNumber; i++)
                                CLR[i] += (rate - LinkCapacity) * partial_pcong
                                * ((float) SubMatrixRate[i][il[i]] / (float)
                                rate);
                        }
                         if (total_pcong > max_pcong)
                            return;
                    }
                }
            }
        }
    }
}
```

```
/****************************************************************************/
/* evaluation for all connections, the balenced algorithm is used           */
/****************************************************************************/
void      ECAprocedure(int ConnectionNumber[])
{
int             i;

    /* inicialisation rutines. */
    test_mj(ConnectionNumber);   /* maximun size test */
    initSmx(ConnectionNumber);   /* the sub-matrices are innitialised */
     /* the sub-matrices are built */
    for (i = 0; i < SourceTypeNumber; i++)
        if (ConnectionNumber[i] != 0)
            prepSmx(ConnectionNumber[i], i);
    /* main procedure */
    (void) ssm_convolution_evaluation();

    /* print results */
    printf("\n\r Probability of Congestion : %14.12Lf ", total_pcong);
    printf("\n,  CLR = %14.12Le \n", (CLR[MXNUMTYPES] / 10.0) / fMean());
}
```

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