Dublin City University

School of Electronic Engineering

SIMULATION OF ATM MULTIPLEXER FOR BURSTY SOURCES

AUTHOR Chen Conger

SUPERVISOR Dr Thomas Curran

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To my wife Shumin Peng and my family.

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DECLARATION

I hereby declare that this dissertation is entirely of my own work and has not been submitted as an exercise to any other university.

Chen Conger

Chen conger

ABSTRACT

Asynchronous transfer mode (ATM) is a promising multiplexing and switching technique for implementing an integrated access as well as transport network and has been adopted by CCITT as a basis for the future broadband integrated services digital network (BISDN) The ATM technique allows digital communication of any type to share common transmission links and switching devices on a statistical multiplexing basis Information is transmitted in the form of constant length cells In an ATM network, the major parameters to cause ATM network performance deterioration are the cell loss and the cell delay at the buffer queue in the ATM multiplexer Therefore, the performance parameters of an ATM multiplexer are specifically focused on the cell loss probability, the cell delay, and the distribution of queueing length at buffer in this study

The performance of an ATM multiplexer is studied, whose input consists of the superposition of homogeneous bursty (ON/OFF) sources, i e , all the superposed sources are characterized by the bursty sources of the same parameter values The cell loss probability and the distribution of queuing length at buffer under different offered load and buffer size conditions are evaluated

An ATM multiplexer with three priority classes is simulated using the priority assignment control method of [15] Under the priority assignment period P and the priority assignment ratio W_n in this method have been defined, the relationship between the traffic balance of classes and buffer size of each is studied The cell loss probability and delay time of each class (same sources and different sources between classes) are evaluated The results are useful to design a economic and effective ATM multiplexer

Contents

Chapter 1 Introduction

The emergence of fibre optics technologies has promoted the rapid development of digital communication systems[1,2] Since about 1985, worldwide activities to evolve Integrated Services Digital Network (ISDN) into an Optical-fibre-based universal Broadband ISDN (BISDN) have resulted in the first baseline standard documents, such as CCITT [1] Recommendation I 121[3] The introduction of BISDN has stimulated the emergence of a wide variety of new services Certain services have been presented in T1[2]'s BISDN Baseline Document[4] and CCITT I 221 BISDN recommendations [5,6] These include

- Broadband video telephony
- Broadband video conference
- Video surveillance
- High-volume file transfer
- High-speed telefax
- Video mail
- Broadband videotext
- High definition television distribution

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[1] CCITT The International Telegraph and Telephone Consultative Committee

[2] The T1 committee is sponsored by the Exchange Carriers Standards Association (ECSA) and accredited by the American National Standards Institute (ANSI)

- Video retrieval
- High-speed unrestricted digital information transmission etc

The above set is not exhaustive and there could be other BISDN services which will be popular in the BISDN era From the point of view of the source characteristics, these sources can be multi-media (voice, data, and video), multi-point (broadcast, point-to-point, and multi-party), and multi-rate (from a few kb/s to several hundreds Mb/s). From the point of view of the traffic-flow characteristics, besides a broad range of bit rates from different sources, each source may have a time varying bit rate requirement over the duration of a connection. Different sources may exhibit a varying amount of burstiness In a word, these new services will have very diverse traffic flow characteristics (e.g., bit rate and burstiness) and quality of service (QOS) requirements. In this context, it is critical that the multiplexing and switching techniques in the BISDN used to support such a wide variety of new services, and the associated bandwidth management, be extremely flexible.

1.1 BISDN Based on ATM

Traditional time division techniques , which divide bandwidth into a number of fixed capacity channel(e.g., Synchronous transfer mode: STM) , simply do not have this required flexibility Packet-switched techniques, which allow variable-length packets, may not meet the stringent delay requirements specified for some services. Asychronous transfer mode (ATM) is a specific packet-oriented transfer method which uses an asynchronous time division multiplexing technique[7] The multiplexed information is organized into fixed-size packets, called cells. Each cell consists of two sections a cell header and a cell body. The cell header carries information such as the generic flow control(GFC)field, routing field(VPI/VCI), payload type(PT) field, cell loss priority(CLP) field, and header error control(HEC) field[8], The cell body carries the user data(For details, see Chapter 2). The ATM technique allows digital communications of any type to share common transmission channels and switching devices on a statistical multiplexing basis. The reason why ATM is suitable for such a multimedia traffic environment is that it offers a great flexibility in bandwidth allocation through the assignment of fixed length cells. Therefore, ATM is a promising multiplexing and switching technique for implementing an integrated access as well as the transport network and has been adopted by CCITT as the basis for the future BSIDN [9] The BISDN user-network interface is based on ATM over fibre optic facilities

1.2 Objectives and Summary of Thesis

The major benefit of the ATM technique is flexible and efficient allocation of communications bandwidth traffic statistics[10,11] But one of the major problem raised by the ATM technique is the definition of a valid and efficient bandwidth assignment strategy. This strategy requires a full understanding of the impact of various kinds of traffic on the quality of service (QOS) of an ATM network The purpose of this thesis is to address the performance of an ATM multiplexer using simulation methods. The inputs to the multiplexer consists of a number of voice sources, which are of a bursty nature. Only a single ATM output channel with a capacity of 150 Mb/s is considered. For multiple QOS, an ATM multiplexer with three priority classes is also simulated using the priority assignment control method of [63] Generally speaking, the QOS is evaluated by cell loss probability (CLP), the cell time delay, the buffer occupancy distribution, and the gap distribution of the consecutively lost cells. Therefore, The cell loss probability, the distribution of queue length of the buffer under different offered load and buffer size conditions, and the probability distribution of delay time are evaluated. In order to design a economic and effective ATM multiplexer the relationship between the traffic balance of classes and the buffer size of each class is also studied in this thesis

In Chapter 2, the state of development of switching, multiplexing, and transmission technologies is reviewed. This is done mainly for purpose of introducing an ATM multiplexer. The traffic process of a single bursty source is also discussed in this chapter.

The methods of generating random numbers are discussed in Chapter 3. Because, the random numbers of each probability distribution are derived from the uniform distribution, therefore emphasis is placed on the methods of generating random numbers with a uniform distribution. The chapter will also refer to the method of generating random numbers with a geometric distribution. In order to guarantee the quality of these random numbers, some methods of testing random numbers are also discussed in this Chapter.

In Chapter 4, The performance of an ATM multiplexer is studied, whose inputs consist of a number of homogeneous bursty (ON/OFF) sources, i.e., all the sources are characterized by bursty parameters of the same value. The cell loss probability and the distribution of queue length of the buffer under different offered load and buffer size conditions are evaluated.

In Chapter 5, an ATM multiplexer with three priority classes is simulated using the priority assignment control method of [?]. The relationship between the traffic balance of classes and the buffer size of each class is studied in order to design a economic and effective ATM multiplexer. The cell loss probability and delay time of each class are evaluated. Finally, conclusions are presented in Chapter 6.

Chapter 2 Overview of Broadband ATM Multiplexer for Bursty Sources

2.1 Switching Technologies

The step-by-step mechanical switch was invented by Strowger to replace manual switching in 1897[12] Mechanical means of providing interconnections were in turn replaced by electronic crossbar switches The earliest switching machines were Space Division Switches(SDS)[13,14], for which a telephone conversation relied exclusively upon a single path in the switching network to facilitate a conversation The switching mechanism employed a toggling method, either mechanically or electronically, among electrical conductors There are two basic switching schemes in digital communications They are circuit switching and store-and-forward switching, and also there are two variants of store-and-ward switching message switching and packet switching (See Fig 2 1) Circuit switching is a much older technology than packet switching, and this technology has been used worldwide over the years for telephony Modern telephone systems and networks are used to transmit data, but voice has continued to constitute the bulk of their traffic[15]

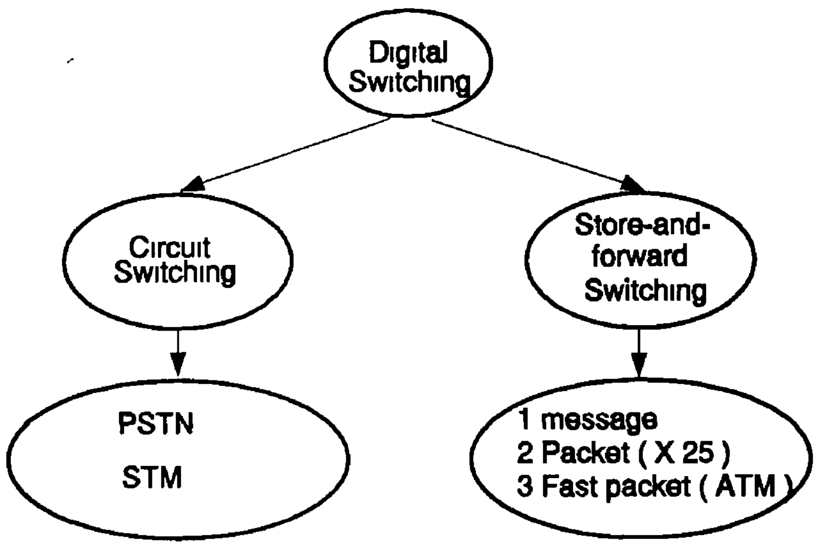

Fig 2 1 Switching techniques

2 1 1 Circuit switching

A circuit switched network is used to set up the traditional type of continuous connection between end points, namely, in circuit switching an electrical circuit is assigned and dedicated to each call and remains busy during the call Historically, circuit switching is an older technique that was designed originally for voice communication in the public switched telephone network (PSTN) With the advent of end-to-end digital communication, direct connections are now possible between digital devices operating at 56 or 64 kb/s over the

PSTN[16] Circuit switching for broadband digital communication will be expounded in section 2 2 2 1(STM)

2 1 2 Message switching

In message switching, the whole information or message(e g , a telegram, page of text, commercial letter) is transmitted through the network as one block of data To this end, it carries the address of the receiver which is read and interpreted at each node of the network En route, the messages are sorted, stored in memory for a certain time, according to the congestion state of the rest of the route, and then transported as a single message Transmission is typically unidirectional The transfer delay can be very long

2 1 3 Packet switching

The packet switching concept was first proposed in the early 1960s[17] by Paul Baran at the RAND Corporation whilst working on military communications systems, mainly to handle speech In packet switching, the information is broken up into pieces, called packet Packets move around the network, from switching centre to switching centre, on a store-and-forward basis The traditional packet switching model, based on X 25 interfaces, is implemented on all three of the lower layers of the OSI model(See [18]) By 1976[14] the CCITT system had adopted the widely accepted X 25 standard for the data user interface to a public packet switched network, allowing user equipment and software to be developed as commercial products to utilize packet switching services Packet switching is widely applied to data and computer-based communications, and it is much more efficient for the transmission of data than circuit switching

2 1 4 Fast packet switching

Fast packet switching is a telecommunication technology which combines features of statistical multiplexing and packet switching Unlike normal packet switching networks, which are designed for data traffic, existing fast packet switching devices operate directly at the original 24-channel, 1 544 Mb/s PCM standard (US, Japan, and Canada), and the 30-channel, 2 048 Mb/s PCM standard (European countries) To satisfy the needs of the multi-rate connections, fast packet switching is used[19] The linkage between Asynchronous Transfer Mode (ATM) and fast packet switching on which it is based will be outlined in section 2 2 2 2

2.2 Multiplexing Techniques

Multiplexing is the operation which consists of grouping several channels, each assigned to a particular communication, in such a way as to transmit them simultaneously on the same physical medium(cable, carrier frequency of a radio link, satellite, etc) without mixing or mutual interference At reception, a demultiplexing as perfect as possible allows these channels to be separated and restored to their original form[20,15] The purpose of the multiplexing is to reduce the number of transmission links needed to carry the sources over some distance between nodes or access points

Multiplexing techniques were introduced in early telegraphy to allow simultaneous transmission of multiple signals over the same line[21] The multiplexing techniques consist of three types, e g , Space division multiplexing, Frequency division multiplexing, and Time division multipleing But, two types of multiplexing techniques are in use today[3] (See Fig 2 2)

1) Frequency Division Multiplexing (FDM)

2) Time Division Multiplexing (TDM)

For TDM, bandwidth is shared by sources transmitting at different times For FDM, bandwidth is shared by sources transmitting at different frequencies or wavelengths For SDM, bandwidth is shared by sources transmitting over

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[3] Recent publications include Space Division Multiplexing among multiplexing techniques It is accomplished by bounding the metallic wires or coaxial pairs together to form a single cable It is a useful and economical technique when relatively few channels are to be transmitted in a short distance within a building or in an exchange area for local customer connections

separate physical channels

Of the foregoing multiplexing methods, TDM is of most relevance to the current study and will be discussed further

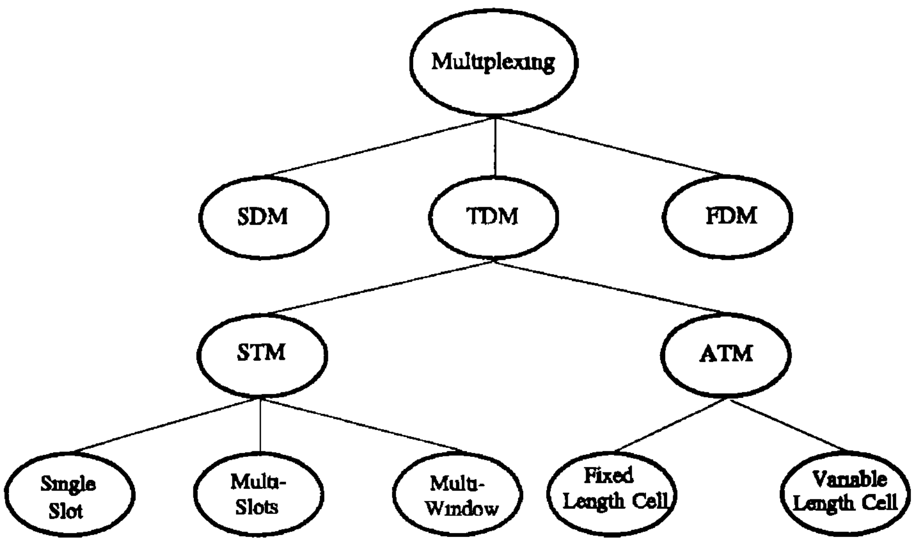


Fig 2 2 Multiplexing techniques

2 2 1 Frequency division multiplexing technique

FDM is the oldest and most common type of multiplexing In FDM, each signal is allocated a discrete portion of the frequency spectrum It allows several analog voice signals (or data channels) to share the same line without interfering with each other Once multiplexed, these signals could be amplified and transmitted at low cost over a long distance. However, the cost of multiplexing, which grows linearly with the number of signals multiplexed, makes these wider band transissions suitable only for long distance and high volume routes for which the increased multiplexing cost can be justified. Therefore, the development of FDM tends to consolidate the long distance network into fewer routes with higher capacity.

2.2.2 Time division multiplexing techniques

Time division multiplexing was used in which a rotary switch connected the various signals to the transmission line. A basic problem then (and now) was the synchronization of the transmitting and receiving commutators. A practical solution introduced by Baodot in 1874 involved the insertion of a fixed synchronizing signal once during each revolution of the switch[14].

There are two major kinds of multiplexing modes in TDM. The first mode assumes a common time reference among the sources, which is denoted as a frame reference *d.* and is generally referred to as the Synchronous Transfer Mode (STM). In STM, the communication bandwidth assigned for each source is termed a Circuit multiplexing(See Fig. 2.3). The second mode assumes no frame reference, with the bits of each source having no implicit ownership, unlike STM, for which each slot is assigned an owner. Hence it is referred to as the Asychronous Transfer Mode (ATM). In

ATM, all information is packed into fixed-size slots called cells or packets, thus the communication bandwidth assigned for each source is termed a packet multiplexing (See Fig.2.4). These two modes are discussed in Section 2.1.2.1 and Section 2.1.2.2 respectively.

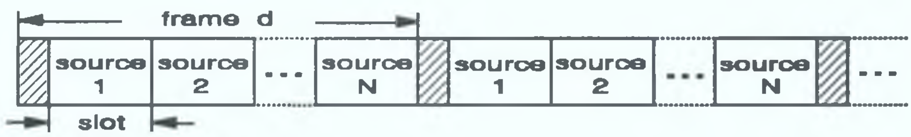


Fig. 2.3 STM principle

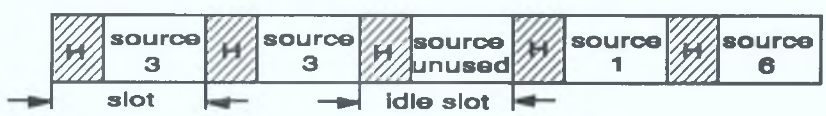


Fig. 2.4 ATM principle

2.2.2.1 The synchronous transfer mode

Three TDM formats of the synchronous transfer mode have been discussed in detailed by Hui in[14,22]. They are referred to as, single slot TDM, multi-window TDM, and multi-slot TDM

Suppose N communication sources S_1, S_2, , S_2, , S_N share a single high speed channel, and each source S_i transmits at a bit rate b_i ($b_i = b$ is a constant when all sources have the same bit rate)

<u>*Single Slot TDM*</u> Assuming that a common frame reference exists among the sources, which have the same rate b, each frame is divided into N slots, where the bits of each source are placed in a chosen slot within a frame, and the source uses the same slot within consecutive frames Therefore, the transmission channel can be shared by at most $N = C/b$ sources This form is only suited to a single-rate environment, namely where all sources S_i have the same rate b

<u>*Multi-Window TDM*</u> Bandwidth assignment becomes more complicated in multi-rate environments In a situation where there are K types of sources of bit-rates b_k, K windows are created within each frame for each type of source, in order for these to carry the multi-rate traffic In general, the multi-window TDM format represents all transmission formats which divide the capacity of a transmission channel to accommodate a fixed number of source types and a fixed number of services per source type As such, the network must choose, among the broad

range of source bit-rates, a set of standard bit-rates, and window sizes This format makes sharing of bandwidth inflexible among service types

Multi-Slots TDM A more flexible TDM format results from removing the restriction that a source may be assigned a maximum of one slot within a frame These leads to the allocation of more flexible time slots and a circuit assigned for a source may have many slots (see [13,15])

The above three formats have the same disadvantages in that the choice of frames is very difficult for a multi-service environment, and the format depends on the transmission channel capacity C, as the number of slots in a frame is given by $N=C/b$ (b is a bit rate of the particular service type among the K service types) In addition, STM systems must accommodate peak transfer rates, thus they cannot transport different quality of services (QOS) efficiently STM is usually most usefully employed by fixed-rate services Higher-layer multiplexing and switching would be required to make maximum use of an STM system

2 2 2 2 The asynchronous transfer mode

There two basic forms of asynchronous multiplexing, e g , fixed length cells and variable length cells

Fixed length cells The information to be transferred is packed into fixed_sized slots called "cells" or "packets" [23,24,25] (See Fig 2 5)

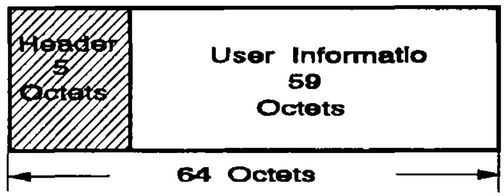


Fig 2 5 Fixed length cell

Each cell consists of two sections a cell header and a cell body The cell header carries information such as generic flow control(GFC)field, routing field(VPI/VCI), payload type(PT) field, cell loss priority(CLP) field, header error control(HEC) field, and reserved field(RES) The cell header structure vary with the interface, i e , User Network Interface(UNI) and Network Node Interface(NNI)(See Figs 2 6 - 2 7), for example, the GFC function is unique for the UNI, the NNI does not have this functionality, Other header functions are common for both interfaces But the maximum commonality between UNI and NNI is desired The cell structure at the UNI and NNI is described in this study The cell body carries the user information[8] (See Figs 2 6 - 2 7)

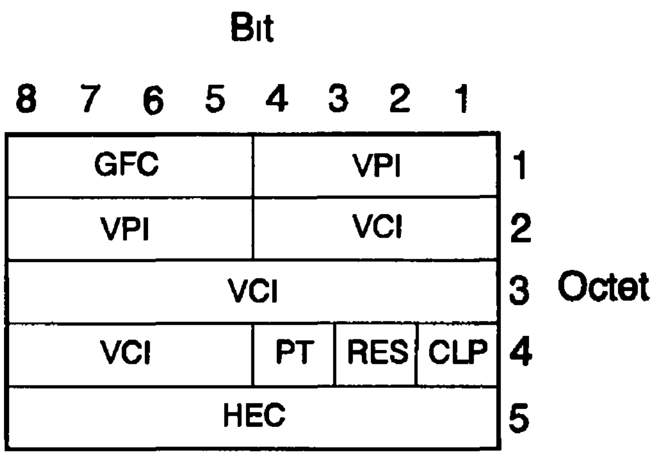


Fig 2 6 Cell header structure at UNI

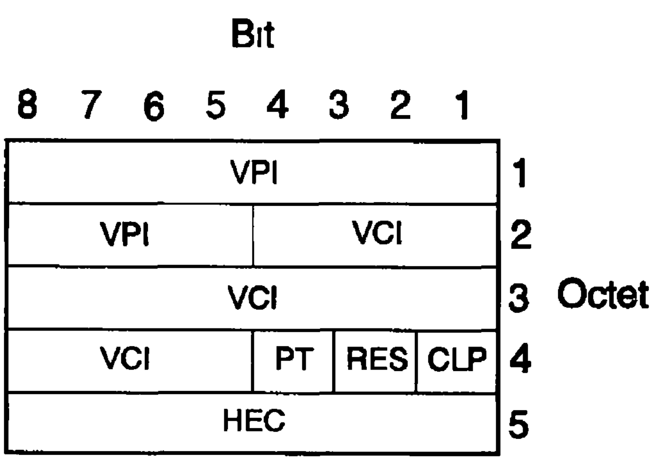


Fig 2 7 Header structure at NNI

There are two major factors in determining the proper cell length for ATM system First, the cell header uses up part of the communication bandwidth of the link, and it is inversely proportional to cell size l, consequently favouring long cells Second, a packetization delay is needed for the source to collect the l bits for a cell This delay is given by l/I_k (I_k is the bit rate of a single input source) Consequently, minimizing packetization delay requires choosing short cells A compromise has to be chosen between the packetization delay and transmission processing efficiency[26] The ATM cell is selected as a 5 octets header field and a 59 octets user information field in this study

Variable length cells Instead of fixed length slots, it is often convenient to use long (say 128 bytes or more) vaiable length cells (see Fig 2 8) Besides the label for ownership, the cell header should also contain the information for cell length to mark the end of the cell, as well as a flag to mark the beginning of the cell For variable length cells, the variable delay before a packet is successfully transmitted depends on the probability distribution of cell lengths Therefore, very long cells may jeopardize the timely delivery of other cells This complicates traffic engineering for controlling delay For this reason, variable length cells are often fragmented into smaller fixed length cells for transport

The asynchronous transfer mode (ATM) is based on fast packet switching (fixed length cells) to be a new transport (multiplexing and switching) technique [23,27], which attempts to eliminate the disadvantages of STM Where there is no existing frame reference, the usable capacity can be dynamically assigned on demand Thus it achieves flexible bandwidth sharing by allowing the sources to seize bandwidth when a sufficient number of bits are generated

In comparison to STM , ATM Multiplexing and switching are less dependent on considerations of bit rates for particular services ATM can flexibly support a wide variety of services and is particularly apt as the bit_rate of each source is time_varying or bursty

Due to ATM advantages mentioned above, ATM is a promising multiplexing and switching technique for implementing an integrated access as well as transport network and has been adopted by CCITT study Group XVIII (international) and T1S1 (North America) as a basis for the future BSIDN structure

2.3 Evolution of Broadband ISDN

Traditionally, the aforementioned services were carried via separate networks, as for example, voice on telephone network, data on computer networks or local area networks, video teleconferencing on private corporate network, and television on broadcast radio or cable networks These networks are largely designed for specific applications and are inflexible for other usages

The ideal situation is one in which a single network provides all these communication services, in order to achieve maximum economic value, based on shared usage The natural evolution of the public switched telephone network and the emergence of digital communication technology has led to the development of a concept known as the integrated service digital network (ISDN), which is built on the ability to provide integrated access to a myriad of services through a single interface[20,28] In 1984, the Plenary Assembly of the CCITT adopted the I series recommendations dealing with ISDN matters CCITT stated that " an ISDN is a network that provides end-to-end digital connectivity to support a wide range of services, including voice and non-voice services, to which users have access by a limited set of standard multipurpose user-network interfaces"[29] ISDN was not designed to support switching and transmission rates at the speeds necessary for the above-mentioned new service types, therefore, a new higher bandwidth method was needed About 1985, worldwide activities commenced to evolve ISDN into an optical-fibre-based universal broadband network Broadband ISDN (BISDN) extends these important concepts of ISDN[5], to provide a universal, long_term transport capability for all services[27,30] The Optical fibre transmission technology is the transport media of choice for network integration, because its vast and reliable transmission capability can accommodate all imaginable services Therefore, the BISDN access can be based on a single optical fibre for various services

There are two BISDN interfaces discussed indomestic and international standards One is the User Network Interface (UNI), and the other is the Network Node Interface(NNI) The UNI and NNI frame stuctures are based on optical interface rates and formats specified initially by CCITT[8] The BISDN architecture is showed by Fig 2 8

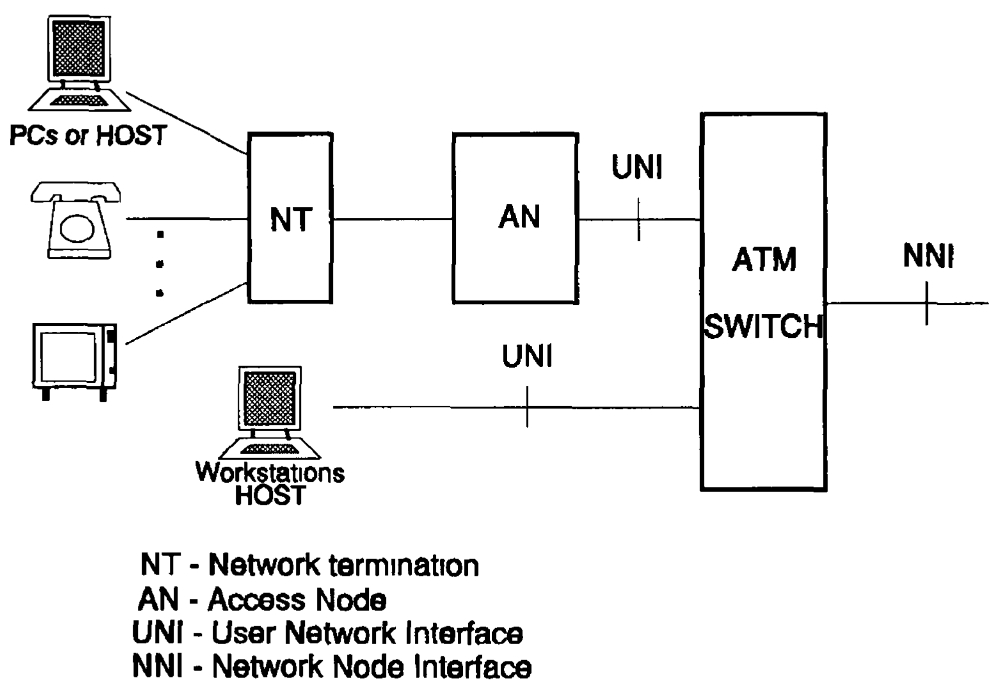


Fig 2 8 BISDN architecture

However, integration within the network can entail different meanings depending on which part of the network is being considered[3,31]

(1) *Integrated access* involves the sharing, among services from an end user, of a single interface to a single transmission link which connects the end-user to a larger network These links comprise the local access network (LAN) within the larger network A well integrated access network should provide flexible multiplexing of a maximum number of services (this issue will be discussed further in Chapter 4)

(2) *Integrated transport* involves the flexible sharing, among services from many users, of high-capacity transmission channels aside from the local access network Integrated transport avoids the segregation of different traffic types and media onto different transmission channels The integration of media on the same channel may facilitate easier interaction between media within the network

(3) *Integrated switching* consists of switching multi-rate, multi-media services within a single switching machine, with the emphasis on a single interconnection network An integrated switch

should remove the necessity of putting in place an interconnection network whenever a novel service of distinct traffic mode is introduced An integrated switching network must be flexible and attain the delay and bandwidth requirements of each service

The technological development of fiber optics, distributed processing, and HDTV (High Definition Television) have accelerated the trend toward the development of BISDN

2.4 Broadband Services

The introduction of BISDN should stimulate the emergence of a wide variety of new services In the business market segment, there is already a demonstrated need for high-speed Local Area Networks (LANs) and LAN interconnects within buildings and in campus enviroments The BISDN can extend these functions by interconnecting LANs from different locations into Metropolitan Area Networks (MANs)[6,32,33] and Wide Area Networks (WANs) The initial demand for broadband networks will likely be from business customers for high-speed date services The market for switched video services for both business and residential customers will emerge later So far, The Local Area Networks (LANs) make possible a distributed processing environment consisting of Personal

Computers (PCs), print servers, file servers, workstations, minicomputers, and gateways Many workstations are attached to LANs However, in the future, high-speed workstations may be connected directly to a public broadband network in order to communicate with other directly attached workstations as well as with workstations, PCs, and servers attached to LANs served by the broadband network Host computers could be connected directly together or to high-speed peripherals via a direct channel connection High-speed distributed host computing can be supported by a broadband network in conjunction with appropriate channel extension equipment[37] (See Fig 2 8)

2.5 ATM Multiplexer for Bursty Sources

In the simpest case, the ATM transport network may be a point-to-point link, consisting of an access multiplexer, link, and demultiplexer In a full evolved network context, the ATM transport network could span several multiplexing, switching, and demultiplexing stages as shown in Fig 2 9 In either case, each node may contribute to the end-to-end cell loss and delay due to buffering In order to meet end-to-end transport QOS requirement, a reference connection must be assumed and an allocation of QOS requirements to each node in the reference connection must be made For this reason, this study focuses on an ATM multiplexer The sources of the ATM multiplexer, the state modes of the sources, and the traffic processes will be firstly assumed in this section The performance of an ATM multiplexer without priority is studied It's input consists of superimposed of homogeneous bursty (ON/OFF) sources, i e , all the superimposed sources are characterized by the bursty sources of the same parameter values (See Chapter 4) An ATM multiplexer with three priority classes is simulated using the priority assignment control method in Chapter 5

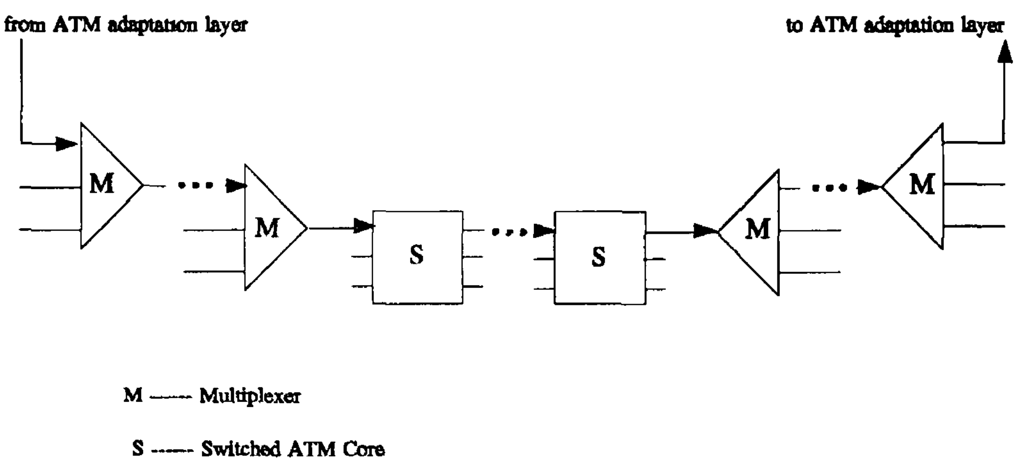


Fig 2 9 ATM transport layer reference connection

2 5 1 Sources of ATM multiplexer

It is assumed that N communication sources S_1, S_2, , S_i, S_N share a single high speed ATM channel (the bit rate C = 150 Mb/s in this study), and each source S_i

transmits at a bit rate $b_i(t)$.

Then, according to the characteristic of the traffic flow, these sources S_i may be divided into the following three forms:

A. The fixed bit rate, such as data information and full_motion video, etc.

B. The bursty (i.e., the bit rate of each source is time_varying), such as voice and still picture video, etc.

C. The bit rate is continuously varying, such as compressed full_motion video, etc.

It has already been noted in section 2.2.2.2 that when the above sources enter into the ATM multiplexer, all information is packetized into fixed_size cells (or packets), which are identified and switched by means of a label in the header.

2.5.2 State models of bursty sources

Many of the traffic sources that an ATM network supports display bursty characteristics. A bursty source emits cells periodically (at a peak rate) during a burst, which is of variable length. Silence periods are also of variable length. An example of this is conventional voice telephony ---- a telephone is communicating when in the active state, or not communicating when in the idle state This two state model is called "the alternating state process," which also suffices to describle other fix bit rate communications (see Fig 2 10a)

To save communication bandwidth, the telephone transmits only during these talk_spurt periods, thus, the traffic process becomes a three_state model (see Fig 2 10b)

In ATM network, the talk_spurt is organized into multiple fixed length cells, so that, a voice call is expressed as a four_state model(see Fig 2 10c)[12,34]

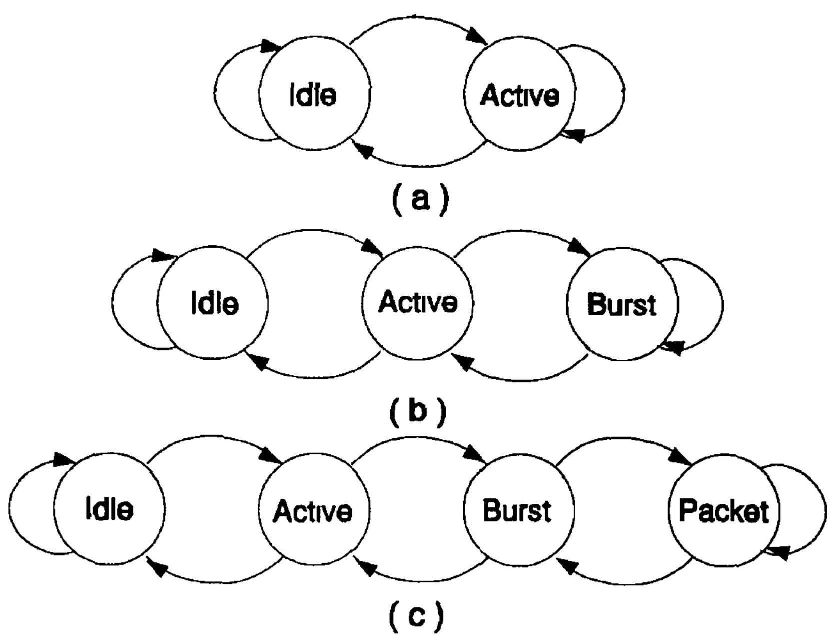


Fig 2 10 Multilevel traffic state models for telephony

2 5 3 Traffic processes of single bursty source

In an ATM multiplexer, all information such as data, voice, and video is conveyed using a fixed sized "cells" This means that the cell stream from a single bursty source is characterized by arrivals at fixed intervals of *T* during bursts with no arrivals during silences(see Fig 2 11) The successive burst and silence periods form an alternating renewal process, i e , all these time intervals are independent with each burst being of random length *XT*, and each silence period of random length *YS*(*YS* is a silence period) The number *X* of cells in a burst is uniformlly distributed on the positive integers For single bursty source, the alternating states of idleness and bursts appear after the call is set up[35,36,37]

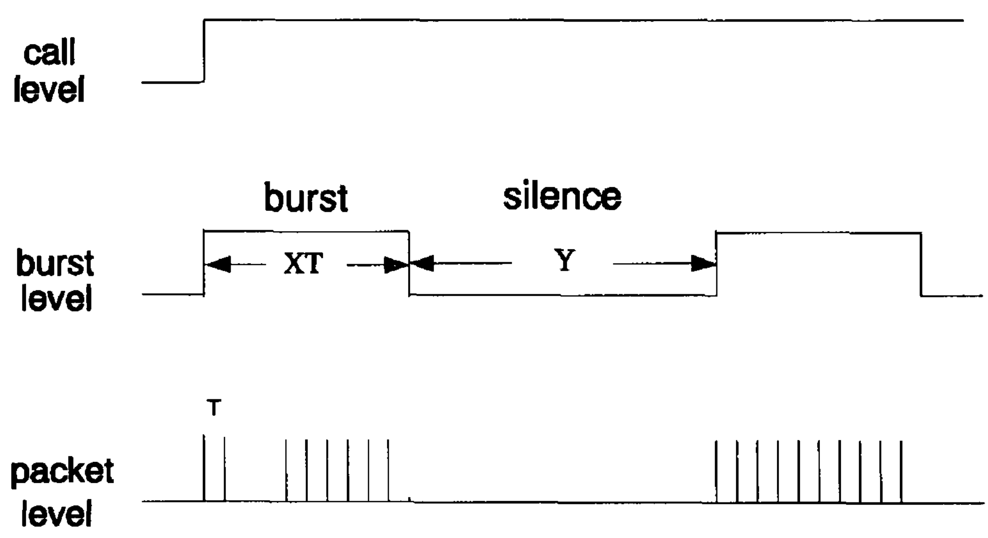


Fig 5 Multilevel traffic processes

For a voice source, it is necessary that the burst is digitized into a bit stream at I=64 kb/s, and segmented into cells of l=512 bits (i e , 64 octets) for transport over an ATM channel with a capacity of C=150 Mb/s Therefore, the voice packetization period is $T=l/I$=8 ms, entailing the source enters into the cell transmission state once every V cell time, where V equals the channel bit rate divided by the source bit rate In other words, a maximum of 2344 voice bursts can be accommodated at a time on a capacity of 150 Mb/s ATM channel[13,38]

$$V = \frac{C}{I} = \frac{150\,Mb/s}{64\,kb/s} \approx 2344$$

Chapter 3 Random Number Generators

The random numbers of various types of probability distributions are important tools in traffic simulation The traffic flows are simulated by the random numbers of the corresponding distribution These probability distributions can be derived from the uniform distribution, so that, emphasis is placed on the methodology for generating (0-1) uniform random numbers As the quality of the generated random numbers directly effects the simulated results, evaluation and test of the methods of generating these numbers is undertaken

3.1 Generating Random Numbers

There are three distinct types of method for generating random numbers[39,40],namely

(1) Manual methods

(2) Mechanical and/or electrical methods

(3) Digital computer methods

Undoubtedly, the last is most commonly used method of obtaining random numbers and involves the use of a pseudo_random number generator.

The suitability of the various types of random number generators may be judged against the following six criteria.

(1) The random numbers should be uniformly distributed.

(2) Each random number should be statistically independent of all other random numbers in the sequence.

(3) Sequences should be reproducible.

(4) Sequences should be not repeated over a given span or period.

(5) The random number generator should be computationally fast.

(6) The software describing the generator should be as concise as possible.

Pseudo_random number generators are normally considered to satisfy these criteria.

Software which employs pseudo_random number generators is usually based upon the following congruence or residue methods

(a) The multiplicative congruence method

(b) The mixed (or linear) congruence method

Comparison of the mixed congruence method with the multiplicative congruence method indicates that its computation time is slower, but it can generate longer sequences that are not repeated over a given period The current study favours the mixed congruence method

The mixed congruence method is also called the linear congruence method, and uses a recursive expression to generate a sequence of numbers from a given initial value

This recursive expression[45,46,41] is of the form

$$x_{i+1} \equiv \lambda x_i + C_0 \quad (\mathrm{mod}\ M) \qquad i = 0, 1, 2, 3, \qquad (2)$$

which can be expressed in algebraic form as

$$x_{i+1} = \lambda x_i + C_0 - K_i M \qquad i = 0, 1, 2, 3, \qquad (3)$$

where K_i is the integer part of the quotient of $(\lambda x_i + C_0)/M$ So that, The equation 2 can be expressed in the following form

$$x_{i+1} = \lambda\, x_i + C_0 - integer\,(\frac{\lambda\, x_i + C_0}{M})\, M \qquad \textbf{(4)}$$

$$i = 0,\, 1,\, 2,\, 3,$$

The random numbers generated by equation 3 are in the range (0 - M) The maximum span therefore is M

To obtain a sequence of uniformly distributed pseudo_random numbers between 0 - 1, the above equation is divided by M[45,47]

$$\frac{x_{i+1}}{M} = \frac{\lambda\, x_i + C_0}{M} - integer\,(\frac{\lambda\, x_i + C_0}{M}) \qquad \textbf{(5)}$$

$$i = 0,\, 1,\, 2,\, 3,$$

Equation 4 can be expressed in the following form which is suitable for computation on a computer

$$\left.\begin{aligned} & x'_{i+1} = (\lambda x'_i + C) - integer\,(\lambda x'_i + C) \\ & x'_0 = \frac{x_0}{M} \quad , \quad C = \frac{C_0}{M} \\ & i = 1\ ,\ 2\ ,\ 3\ , \end{aligned}\right\} \qquad \textbf{(6)}$$

where x_i' is the new sequence between 0 - 1

The value of the modulus M is of the form 2^r (where r is a positive integer), which is the maximum number of random values in a sequence generated by the mixed congruence method The parameters λ , x_0 , C_0 and M determine the statistical quality of the random number generator, and by a suitable choice of values for x_0, λ, C_0 and M, sufficiently long sequences may be obtained

To achieve the full period or span of M pseudorandom numbers, the following are necessary conditions[45]

(a) The initial value x_0 must be any positive integer

(b) The multiplier constant λ should be of the form

$$\lambda = 4\alpha + 1 \tag{7}$$

where α may be any positive number

(c) The constant C_0 must be a positive odd number

(d) Since the vast majority of digital computers operate using binary numbers, the value of the modulus M is of the form 2^r In most cases, r is chosen to be the number of usable bits in the computer word In a computer having a word length of say 32 bits, r would have a value of 31, since one bit is used as the sign bit This means, M is chosen according to the capacity of the computer and the practical requirement(x_0=32413 0, C_0=15683 0, λ=44441 0, $M=2^{31}$ in this thesis)

3.2 Generating Geometric Random Numbers

In this thesis, the input sources of an ATM multiplexer are bursty A bursty source consists of cell arrivals occurring at fixed length intervals during bursts and no arrivals at all during silences The burst and the silence lengths are generated according to a geometric distribution as will be discussed in Chapter 4 Therefore, generating geometric random numbers to simulate every source input into the ATM multiplexer is of prime importance The random numbers forming geometric distribution are derived from the uniform distribution, and the method for generating uniform random numbers has already been discussed in section 3 1

Suppose that a sequence of Bernoulli trials is continued until the first success occurs Let ξ be the random variable which counts the number of trials before the trial at which the first success occurs ($0 \le \xi < \infty$) It assumes the value 0 if and only if the first trial yields a success, hence, with probability $P(k)$ [42]

$$P(k) = q^k p \tag{8}$$

where $p = 1 - q$

The expected value is given by

$$E[\xi] = \frac{q}{p} \tag{9}$$

The average length of the burst and silence are 300ms (220 cells) and 450ms (132352 slots) respectively, in this study Thus

$$q_B = \frac{220}{221} = 0\ 9954751 \tag{10}$$

$$q_S = \frac{132352}{132353} = 0\ 9999924 \tag{11}$$

Therefore, according to the above mentioned definition of geometric random variables ξ, a sequence of Bernoulli trials is continued until the first $x \geq q_B$ (q_S) (The random number x is taken from (0 - 1) uniform random number generator) ξ is that obtained through counting the number of random numbers before the random number $x \geq q_B$ (q_S)

3.3 Test of The Statistical Characteristics of Pseudorandom

To obtain a sequence of uniformly distributed pseudorandom numbers, the six criteria described previously should be fulfilled The requirements outlined in points 3 - 6 can be met by the methods discussed in the previous paragraph The first two criteria, i e , the uniformity and the independence of the random numbers, should be verified by using statistical tests The basic statistical tests required include

(a) Test of the parameters

(b) Test of the uniformity (or the distribution)

(c) Test of the independence

3 3 1 Test of parameters

The tests of the parameters include tests of the value of the mean, ith power sample moment, and variance

The density function [43,44] of the uniform random variable X on the interval (0 - 1) is

$$f(x) = \begin{cases} 1 & 0 \le x \le 1 \\ 0 & elsewhere \end{cases} \qquad (12)$$

and the corresponding distribution function is

$$F(x) = \begin{cases} 0 & x < 0 \\ x & 0 \le x \le 1 \\ 1 & x > 1 \end{cases} \tag{13}$$

The mean, 2nd power sample moment, and variance of the uniform random variable X on the interval (0 - 1) respectively are [47,48,52]

$$\left.\begin{aligned} E(\xi) &= \int_0^1 x\,dx = \frac{1}{2} \\ E(\xi^2) &= \int_0^1 x^2 dx = \frac{1}{3} \\ D(\xi) &= \int_0^1 \left(x - \frac{1}{2} \right)^2 dx = \frac{1}{12} \end{aligned}\right\} \tag{14}$$

The statistical estimator of the parameter of a random variable X can also be a random variable which depends upon a random sample $x_1, x_2, x_3, \qquad, x_N$ The three most common statistical estimators are the sample mean $\overline{X}$, the 2nd power sample moment $\overline{X^2}$ and the sample variance $\overline{S^2}$ as defined by

$$\left.\begin{aligned} \overline{X} &= \frac{1}{N}\sum_{i=1}^{N} x_i \\ \overline{X^2} &= \frac{1}{N}\sum_{i=1}^{N} x_i^2 \\ \overline{S^2} &= \frac{1}{N}\sum_{i=1}^{N} (x_i - \overline{x})^2 \end{aligned}\right\} \tag{14}$$

As such, $\overline{X}$ is a statistical estimator of $E(\xi)$, $\overline{X^2}$ of $E(\xi^2)$, and $\overline{S^2}$ of $D(\xi)$

The null hypothesises[47,49,50] can be tested as follows,

$$\left.\begin{aligned} E(\xi) &= \frac{1}{2} \\ E(\xi^2) &= \frac{1}{3} \\ D(\xi) &= \frac{1}{12} \end{aligned}\right\} \tag{15}$$

which allows us to prove that N random numbers are from the generator with a uniform distribution

Given the value of a random sample x_1 , x_2 , x_3 , , x_N from a (0 - 1) random number generator For the large sample case ($N \geq 30$), the expression [48,45] of the test statistic is

$$U = \frac{(\bar{r} - \mu_0)}{\left(\frac{\sigma}{\sqrt{N}}\right)} \qquad \textbf{(16)}$$

where $\bar{r}$ is the statistical value of one of the the equation 14, μ_0 is a constant value of the corresponding parameter in the equation 15, $\sigma/\sqrt{N}$ is a mean square deviation of the corresponding parameter in the equation 14 and N is the capacity (or size, N=2000 in this case) of the random samples

To calculate the test statistic U, it is necessary that σ^2/N is calculated, i e , $D(\bar{X})$, $D(\bar{X^2})$, and $D(\bar{S^2})$

For the large sample case ($N \rightarrow \infty$), according to the central limit theorem, if x_1 , x_2 , x_3 , , x_N have the same distribution, and $\sigma^2=D(\xi_i)$, $\alpha=E(\xi_i)$ ($i=1,2,3,$ N), U is a asymptotic nomal distribution $N(\alpha, \sigma/\sqrt{N})$ Thus

According to the definition of the variance[47], $D(\xi^2)$ is expressed by the following

$$\left.\begin{aligned} \bar{x} &= \frac{1}{N}\sum_{i=1}^{N} x_i \\ D(\bar{x}) &= \frac{\sigma^2}{N} = \frac{D(\xi)}{N} = \frac{1}{12N} \\ D(\overline{x^2}) &= \frac{D(\xi^2)}{N} \end{aligned}\right\} \tag{17}$$

$$D(\xi^2) = E(\xi^4) - [E(\xi^2)]^2 = \frac{1}{5} - \frac{1}{9} = \frac{4}{45}$$

thus

$$D(\overline{x^2}) = \frac{4}{45N} \tag{18}$$

By the same method, $D(\overline{S^2})$ is also solved

$$D(\overline{s^2}) = \frac{D[(\xi - \frac{1}{2})^2]}{N} \tag{19}$$

since

$$\begin{aligned} D[(\xi - \tfrac{1}{2})^2] &= E[(\xi - \tfrac{1}{2})^4] - [E(\xi - \tfrac{1}{2})^2]^2 \\ &= \int_0^1 (\xi - \tfrac{1}{2})^4 d\xi - [\int_0^1 (\xi - \tfrac{1}{2})^2]^2 \\ &= \frac{1}{80} - \frac{1}{144} = \frac{1}{180} \end{aligned}$$

thus

$$D(\overline{s^2}) = \frac{1}{180N} \tag{20}$$

Substituting equation 13 and equation 17 -- 20 into equation 16, the test statistics of the mean, 2nd power sample moment, and variance are:

$$\left.\begin{aligned} U_1 &= \sqrt{12N}(\overline{x} - \frac{1}{2}) \\ U_2 &= \frac{1}{2}\sqrt{45N}(\overline{x^2} - \frac{1}{3}) \\ U_3 &= \sqrt{180N}(\overline{S^2} - \frac{1}{12}) \end{aligned}\right\} \tag{21}$$

According to equation 21, the values of the test statistics U_1, U_2, U_3 were found to be 0.063324, -0.260758, -1.288283 respectively (See Table I). The critical values U_α of size α*[4] were then determined, U_α=1.645 . These results showed that $|U_1|$, $|U_2|$, $|U_3|$ < U_α . Therefore, the null hypothesis is accepted.

---

* The level of significance of the test α is usually chosen to be 0.05.

Table I

Lambda=44441 0 x_0=32413 0 C_0=15683 0

$\overline{X}$ =0 500409 $\overline{X^2}$=0 331595 $\overline{S^2}$=0 081186

$U1$=0 063324 $U2$=-0 260758 $U3$=-1 288283

x^2=5 73

J	P(J)	U(J)
1	3 224710e-03	0 144177
2	1 419969e-02	0 634712
3	2 927258e-03	0 130813
4	2 618280e-02	1 169759
5	-1 747989e-04	-0 007807
6	-3 061413e-02	-1 367050
7	2 185708e-02	0 975766
8	-2 260654e-02	-1 008971
9	-3 442464e-02	-1 536049
10	7 923329e-03	0 353455

3 3 2 Test of uniformity

The test of the uniformity of the pseudorandom numbers is also known as the goodness_of_fit test[48,49,46] The common chi_square goodness_of_fit test was used for the purposes of this study

The goodness of_fit test is based on the difference between observed and expected frequencies of the random numbers as follows

$$x^2 = \sum_{i=1}^{k} \frac{(O_i - Np_i)^2}{Np_i} \qquad \textbf{(22)}$$

where O_i (i=1,2,3, 10) and Np_i represent the observed and expected frequencies respectively, for the ith cell Hence N is the capacity of the random samples and p_i is the probability of the value of the random variable ξ in the ith cell

The test procedures employ the following forms
Each element of a given random sample x_1 , x_2 , x_3 , x_N falls into one of the k cells c_1 , c_2 , c_3 , c_k This test will determine at the α level of significance, whether or not it is reasonable to suppose the observed distribution of the N sample values is consistent with the null hypothesis

Step 1 Count the number O_i of observed elements in cell c_i, for $i = 1, 2, \ldots, 10$

Step 2 On the basis of the null hypothesis, calculate Np_i ($p_1=p_2=\ldots p_{10}=0.1$, for this project), the expected number of elements in cell c_i, for $i = 1, 2, \ldots, k$ ($k=10$)

Step 3 Calculate the chi_square statistic x^2 ($x^2=5.73$)

Step 4 Calculate the number of degrees of freedom m of the underlying chi_square distribution Set $m = k - 1 = 9$ in this study

Step 5 Find the critical value x_α^2 ($x_\alpha^2=16.92$) such that the probability of a chi_square random variable with m degrees of freedom will exceed x_α^2 is α (see APPENDIX B [8])

Step 6 Since $x^2 < x_\alpha^2$ the null hypothesis will accept that the random numbers generated by the random number generator are uniform (See Table II)

Table II

SUBINTERVAL	O_i	E_i	$(O_i - E_i)^2$
[0 0 - 0 1)	198	200	4
[0 1 - 0 2)	183	200	289
[0 2 - 0 3)	199	200	1
[0 3 - 0 4)	211	200	121
[0 4 - 0 5)	225	200	625
[0 5 - 0 6)	198	200	4
[0 6 - 0 7)	191	200	81
[0 7 - 0 8)	196	200	16
[0 8 - 0 9)	201	200	1
[0 9 - 1 0)	198	200	4
Total	2000	2000	1146

3 3 3 Test of independence

The main objective in testing the independence of pseudorandom numbers is to determine whether a linear relationship exists between the pseudorandom numbers A number of methods of testing independence are currently in use In this project, independence is examined using a test of the correlation coefficient

The correlation coefficient between two variables is a measure of their linear relationship and is denoted by r_{xy}[47,49,51], which is defined as

$$r_{xy} = \frac{s_{xy}}{s_x * s_y} \tag{23}$$

s_{xy} represents the covariance between x and y

s_x, s_y represent the standard deviation of x and y respectively

The equation above may be expressed by the following form

$$r_{xy} = \frac{\frac{1}{N}\sum_{i=1}^{N} x_i y_i - \frac{1}{N^2}\left(\sum_{i=1}^{N} x_i\right)\left(\sum_{i=1}^{N} y_i\right)}{\sqrt{\left[\frac{1}{N}\sum_{i=1}^{N} x_i^2 - \left(\frac{1}{N}\sum_{i=1}^{N} x_i\right)^2\right]}\sqrt{\left[\frac{1}{N}\sum_{i=1}^{N} y_i^2 - \left(\frac{1}{N}\sum_{i=1}^{N} y_i\right)^2\right]}} \tag{24}$$

To compute this equation for a data sample $\{(x1,y1), (x2,y2), \ldots, (x_N,y_N)\}$, rank the x_i in order, and likewise rank the y_i. As x_i and y_i are from a same population, the maximum distance between x_i and y_i is defined as j ($j=1,2,3, \ldots, j=(1-10)$). Therefore, $y_i = x_{i+j}$, $\bar{x} = \bar{y}$, and $s_x = s_y$ when these formulas are replaced into the equation 24, it can be written as follows

$$r_{x_i x_{i+j}} = \frac{\frac{1}{N-j}\sum_{i=1}^{N-j} x_i x_{i+j} - \bar{x}^2}{\bar{s}^2} \qquad (25)$$

So that, $r_{x_i x_{i+j}}$ is a number between -1 and +1. If $r_{x_i x_{i+j}}$ equals zero then x_i and x_{i+j} are not correlated. A positive correlation means that if a particular x_i happens to be larger than the mean $\bar{x}$, then x_{i+j} will also (on average) be larger than the mean $\bar{x}$. For a negative $r_{x_i x_{i+j}}$, a larger x_i will imply a smaller x_{i+j}, where $r_{x_i x_{i+j}}$ equals 1 (or -1) then x_i and x_{i+j} will have a perfect correlation.

The null hypothesis states, $r_{i\,i+j} = 0$. For a large sample size(N-j>50), according to the central limit theorem, the test statistic U has approximately a standard normal distribution Therefore, the expression[8] of the U is

$$U = r_{i\;i+j}\sqrt{N-j} \qquad (26)$$

Suppose the significance level is α = 0 05 ,the critical values of U_α are determined (see APPENDIX A) Since $|U| < U_\alpha$, the null hypothesis was accepted

Chapter 4 Simulation of ATM multiplexer for burst sources

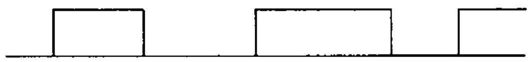
4.1 Introduction

The ATM technique allows digital communications of any type to share a common transmission channel and switching devices on a statistical multiplexing basis Information is transmitted in the form of fixed length cells Chapter 2 has outlined three kinds of sources in the ATM network, e g ,

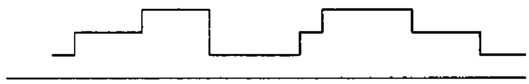
(a) *ON/OFF sources* emit cells periodically during activity periods, or "bursts" of variable length alternating with silences, also of variable length,

(b) *Piecewise constant bit rate sources* emit cells periodically at a frequency determined by their bit rate,

(c) *Continuous varying sources* The cell emission rate might vary continuously, in the sense that the interval between successive cells varies gradually or discontinuously, with the rate changing at random instants between different constant values[47] (See Fig 4 1)



(a) ON/OFF source



(b) Piecewise constant rate source

(c) Continuously varying rate source

Fig 4 1 Variable bit rate sources

The previously proposed analytical approaches[61] do not adequately cover the large variety of real situations and a complete analysis of the effects of the various source parameters on the performance of an ATM multiplexer is not available yet Therefore, the simulated modelling of an ATM multiplexer loaded by the superposition of homogeneous bursty sources is studied in this Chapter, i e , all the superimposed sources are characterized by the same parameter values The services of the different QOS requirements will be discussed in Chapter 5

Many of the traffic sources that an ATM network supports, display bursty characteristics Bursty sources such as voice sources consist of arrivals occurring cells periodically (at a peak rate) during talkspurts and no arrivals at all during silences In [61], the superposition of a multiplicity of bursty sources modeled by Markov Modulated Poisson Process (MMPP) It is assumed that the bursts (or silences) are characterized by the geometric and uniform distribution respectively in this chapter A comparison of results for the geometric distribution and uniform distribution is shown in Fig 4 2 for the mean offered load ρ = 0 9, 0 92, 0 94, 0 96, 0 98, 1 0 Thus Fig 4 2 it can be seen that there is no obviously difference for the cell loss values between the geometric distribution and uniform distribution To save simulation time(Because, the simulation time using geometric distribution is very long), the uniform distribution is employed in all simulation experiments that follow

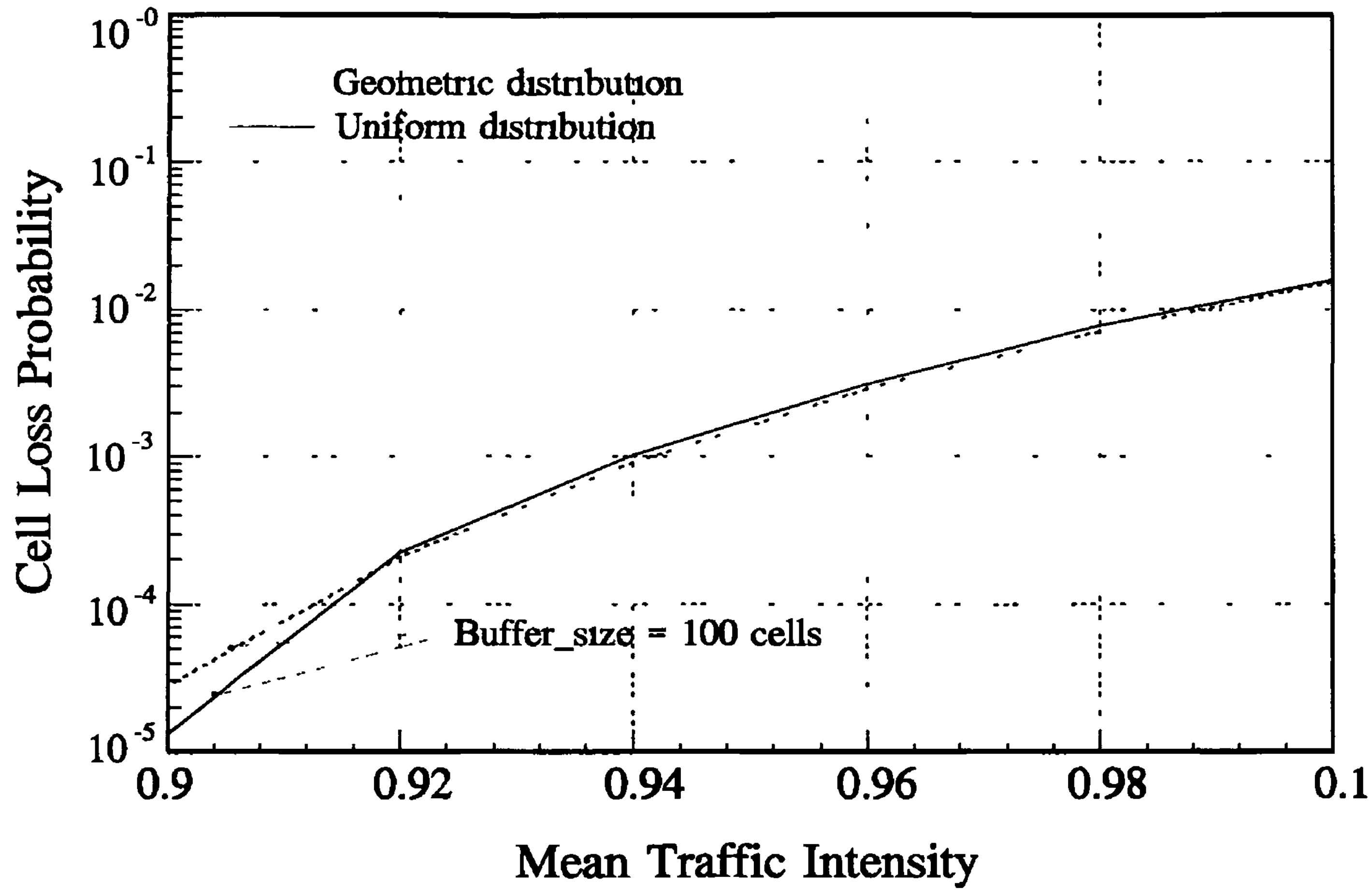


Fig 4 2 Cell loss probabilities versus mean offered load for the buffer size = 100 cells and for both geometric and uniform distribution

The multiplying of a large number of bursty sources on a high capacity ATM channel gives rise to the queuing system of single server and deterministic service time Queues of cells may arise because available capacity is "overallocated," using statistical multiplexing to gain efficiency in a superposition of bursty sources In general, the total bit rate of active sources is small enough to be handled by an ATM multiplexer, but, at times, a large number of sources will emit bursts simultaneously Even if the cell mean arrival rate is less than the output capacity, a queue will still occur due to the coincidence of cell arrivals from different sources If there is excessive use of the bandwidth, then this may cause cell loss at the buffer

To ensure network transparency with respect to offered services, the QOS parameters (the cell loss probability and the cell delay) will occur within specified limits, which are for further study and are dependent on connection type[6,48] The cell delay consists of the cell packetization delays, the propagation delays, and the delays of the output queue The cell packetization delays and propagation delays over transmission lines are fixed and unchangeable regardless of traffic In addition, the cell loss and misdelivery due to the ATM header field error during transmission are also independent and not considered further. So that, the key parameters that cause ATM network performance deterioration are the cell loss and the cell delay in the buffer queue to the output channel in the ATM multiplexer Therefore, the performance parameters of an ATM multiplexer are specifically focused on the cell loss probability and the cell delay in this study[49]

Delay control methods Two delay control methods have been developed expounded in [50] 1) The method of determining the upper limit of the queue length and delay can not exceed this upper limit, 2) To use larger queues and "clipped" cells, which exceed the delay bound, are counted in the lost cells[53] Method 1) is selected in this study (See section 4 3)

Loss rates Loss rates of the order of 10^{-10} have been quoted in other papers[51,52] As the operating speed of the UNIX workstation is limited, it is impossible to perform a simulation for a long enough time Thus, loss rates of the order of 10^{-6} are only obtained in this Chapter

Assuming a finite buffer size of five hundred cells (the maximum delay is 500 cells), the cell_buffer overflow probability is well under 10^{-6} at 90% mean offered load In order that the ATM multiplexer can be sized to satisfy such quality of service objectives, besides showing the cell loss probability for different K (buffer size) and ρ (mean offered load), the distribution of queuing length (or buffer occupancy) is also shown in section 4 5

An ATM multiplexer is composed of three parts, i e , input sources, multiplexing modules, and an output channel In this study it is modelled to be of the following basic form

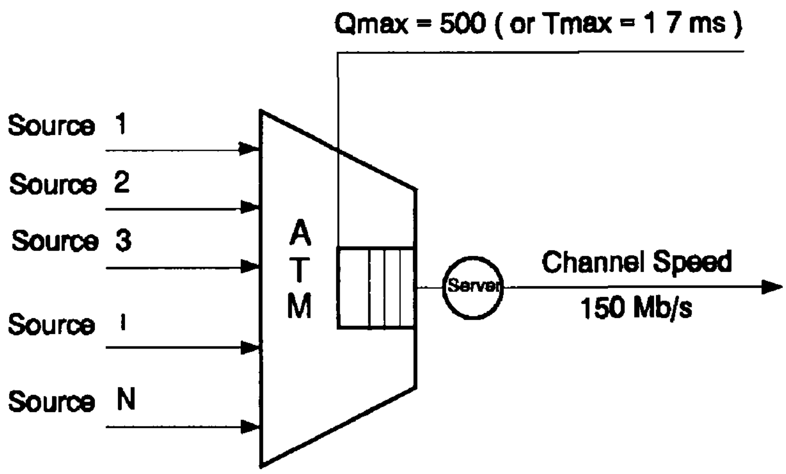


Fig 4 3 ATM multiplexer system

4.2 Simulation Traffic Source Models

The traffic processes of the bursty sources have already been discussed in Chapter 2 In summary, A channel is shared by N statistically identical and independent sources, alternating between bursts and silence periods In the following simulations these will be uniformly distributed with mean value λ(cells per burst) and μ (slots per silence period) Initially when the simulation is commented the burst state is generated at random for each source using a predetermined probability distribution ζ[53] It is assumed that ζ equals 0 4[14] in this Chapter

The parameters λ and μ can then be chosen according to the equation

$$\zeta=\frac{\lambda}{\lambda+\mu}$$

The cell arrival process from a single bursty source consists of arrivals occurring at fixed intervals of *T* ms during bursts and no arrivals at all during silences Therefore, the successive burst and silence periods form an alternating renewal process All these time intervals are independent, with each burst being of random length *XT* and each silence period of random length *YS* *T* is the time between two successive cells arrivals during a voice burst This is a small unit of information with a 4 octet header and 60 octet payload (a cell equals 512 bits) in this study(see Fig 4 4)[54]

The interarrival times of the cells are usually one packetization period *T*, but occasionally are expanded to encompass a packetization period plus a silence period ($Z = YS + T$) Thus, each cell interarrival time from one bursty source equals length T = 1 365 ms (The peak offered load of each source is 375 kb/sec, the mean offered load of each source is 150 kb/sec, and a cell length is 512 bits), with probability p = 219/220 and of length $YS + T$ with probability $(1 - p)$ = 1/220, as shown in Fig 4 4 In general the packetization period T depends on the cell length and the coding scheme used If we are to consider 64 kbs/sec ADPCM and a cell length of 512 bits then the value of T would be 8 ms The higher data rate which is used in this study would arise from high quality speech links

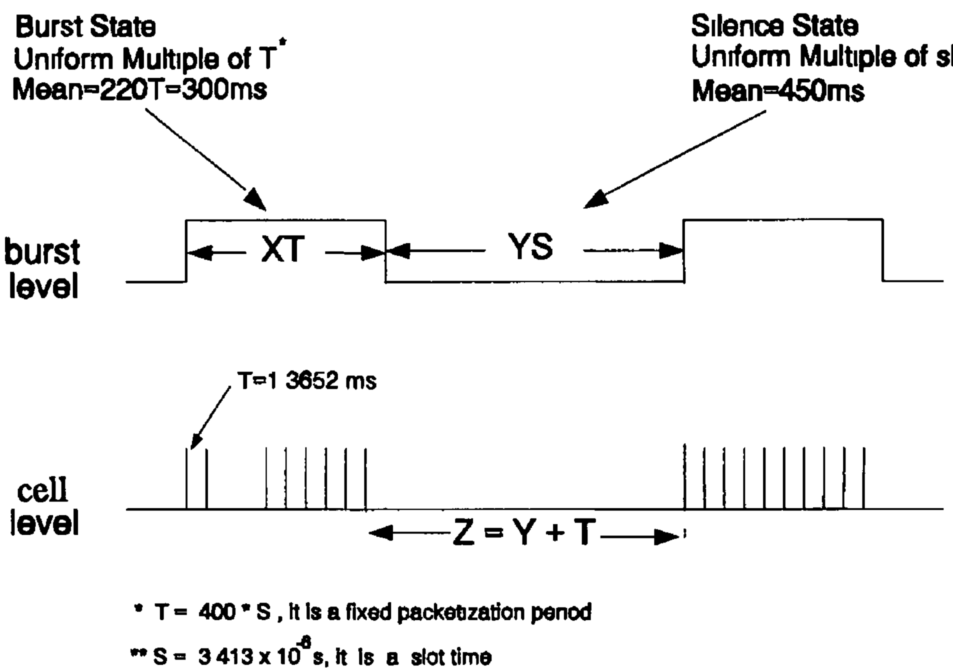


Fig 4 4 Packet arrival process for a burst source

4.3 Service Model and Flowcharts

The ATM multiplexer is assumed to be a standard single-server queue with limited buffer and a first-in-first-out (FIFO) service discipline The traffic entering the FIFO queue is a superposition of N independent and identically distributed cell streams A cell service time equals a fixed slot time of (512 bits)/(150 Mb/sec) ≈ 3 413 10^{-6} second In standard queuing parlance, this system can be simulated to become a discrete_time ΣG/D/1/K (general statistics/constant service time/single server/finite buffer) queuing model[44,55] The cells of each source arrive at the multiplexer with an average rate, I = 293 cells/sec (150 kb/sec) An arriving cell is transmitted immediately if the buffer is empty and the output channel is idle, otherwise this cell is queued in the ATM multiplexer buffer When the buffer content reaches the threshold Q_{max} (or delay time T_{max}), the newly arriving cells are discarded without generating retrials The server models a slotted channel with a fixed capacity of C = 150 Mb/sec, and a fixed cell size of D = 512 bits/cell The service rate is denoted by $U = C/D \approx$ 293000 cells/sec The flow diagrams illustrating this process are shown in Figs 4 5 - 4 6 The priority service discipline will be discussed further in chapter 5

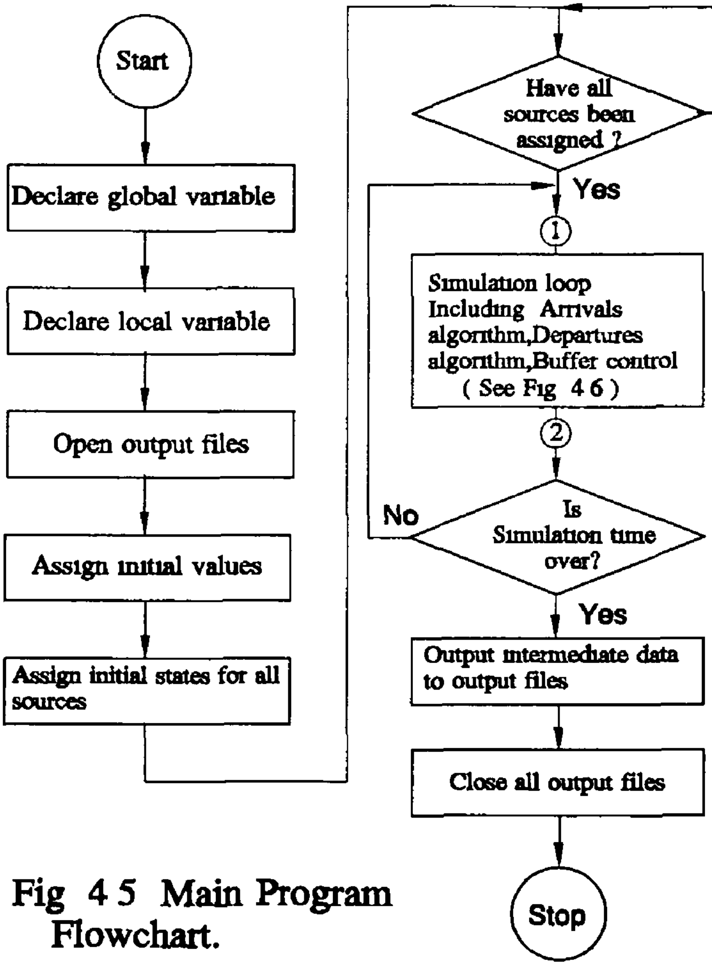


Fig 4 5 Main Program Flowchart.

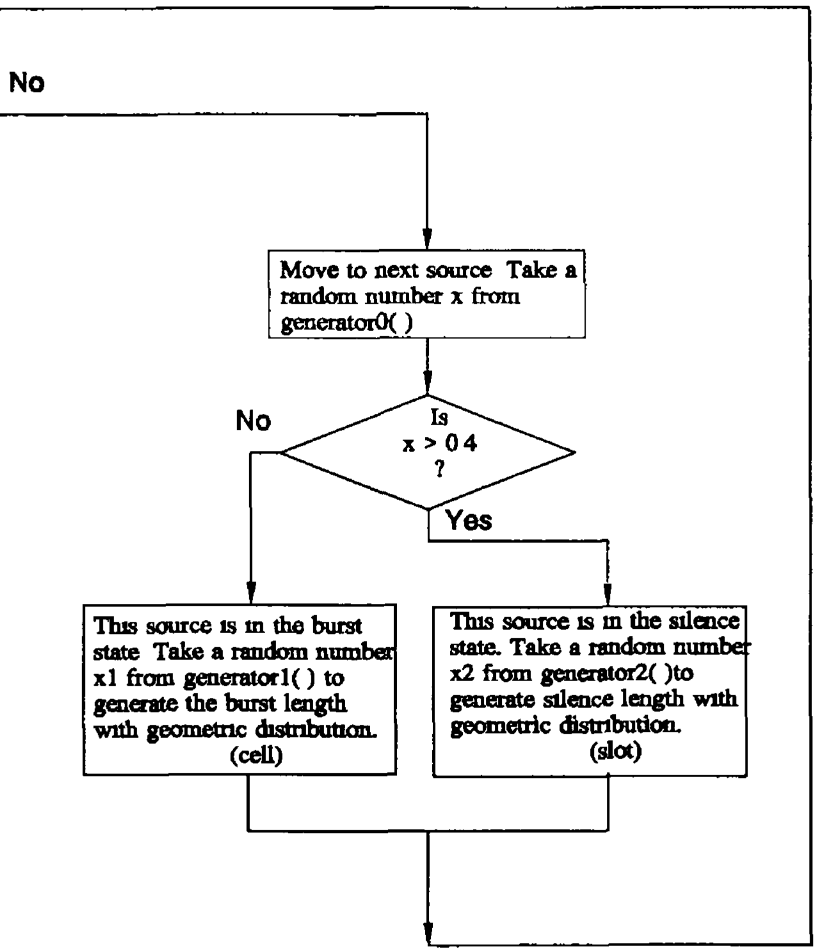
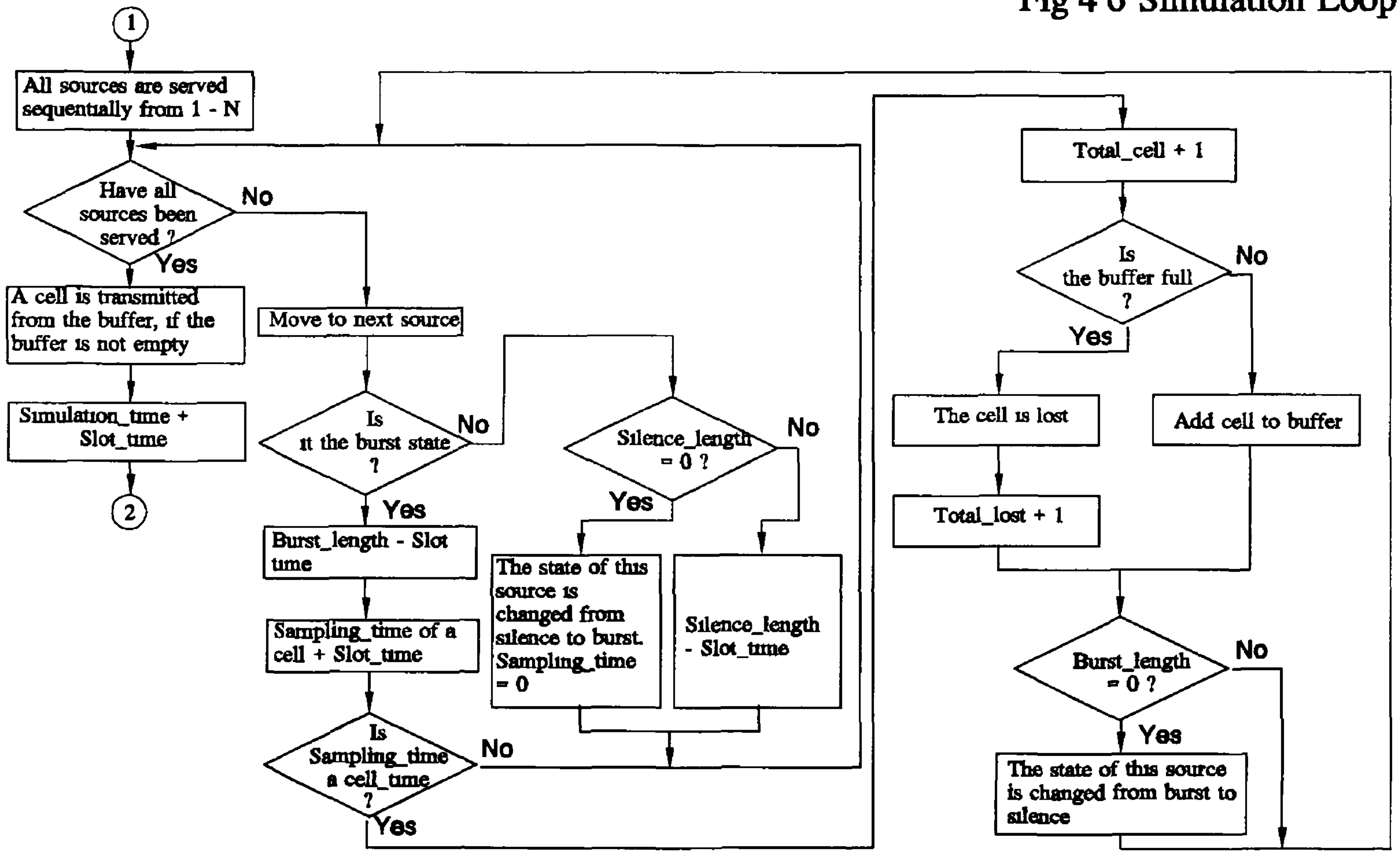

No
Move to next source Take a random number x from generator0( )
No
Is x > 0 4 ?
Yes
This source is in the burst state Take a random number x1 from generator1( ) to generate the burst length with geometric distribution. (cell)
This source is in the silence state. Take a random number x2 from generator2( )to generate silence length with geometric distribution. (slot)

Fig 4 6 Simulation Loop



4.4 Simulation Assumptions and Conditions

Several assumptions and conditions associated with the source parameters and the system parameters of an ATM multiplexer must be defined before it's simulation is attempted

The Source Parameters

1 The number of sources N is 900, 920, 940, 960, 980, and 1000, (that means the average offered load ζ is from 0 9 to 1 0)

2 All sources are independent and identically distributed

3 The average burst length of each source XT is 300 ms This average value is derived from a uniformly distributed burst length which is a multiple of T

4 The average silence length of each source YS is 450 ms Again a uniform distribution of an integer number of slots describes this random variable

5 The activity factor of the burst ζ is 0 4

6 The cell size is 512 bits

7 The average bit rate of each source A_0 is 150 kb/s (the peak bit rate A is equal to 375 kb/s, the mean offered load of N sources ρ equals NA_0)

The System Parameters

1 The buffer size K was chosen 100, 200, 300, 400, and 500

2 The service discipline is FIFO

3 There is only one output channel

4 The bit rate of output channel C is 150 Mb/sec and $M = C/A$ indicates the maximum number of sources that can be accommodated in the ATM multiplexer

For the values chosen this gives an absolute maximum number of channels N = 1000, program using the C language at workstation To obtain the performance measures (Cell loss probability and delay distribution etc), each simulation was run for 60 sec

4.5 Performance Results

Using the above values of the source parameters and the ATM system parameters, the simulation results are presented in Figs 4 7 - 4 15

The distribution of the queue length (or buffer occupancy), for Buffer size = 100, 200, 300, 400 and 500, and for different values of the mean offered load ρ (0 9, 0 92, 0 94, 0 96, 0 98, 1 0), is shown in Figs 4 7 - 4 11 For each buffer size, the probability distribution of queue length follows a bimodal behaviour, i e , the queue is alternatively almost full or almost empty[44] Such a phenomenon is inherent in the statistical multiplexing, and which is more pronounced when the value of ρ is higher Moreover, the curves of Figs 4 7 - 4 11 show that a buffer size of about 30 cells is the threshold beyond which the effect of the correlations cannot be neglected The region to the left of that threshold is called " sensitive region " The bimodal behaviour has been obtained using analytical and simulation methods for K = 50 and for different values of ρ (0 9, 0 5, and 0 2) in Ref [56]

In Figs 4 12 - 4 13 , the cell loss probability is plotted against the buffer size, for different values of ρ In principle, a valuable reduction of the cell loss probability can only be achieved by reducing ρ or increasing K The former , with low values of ρ imply that the efficiency of the dynamic bandwidth assignment will be low With the latter, the obtained curves (See Fig 4 12) point out that an increase of the buffer size K is practically ineffective for the cell loss probability Such a phenomenon is more remarkable when increasing ρ Only if K is increased up to extremely high values can the cell loss values be evidently reduced, but it is also to be taken into account that a maximum delay T_{max} constraint implies an upper bound of K, for the hypothesized values of C As a consequence, the matching between the desired quality of service and the performance of the ATM multiplexer can be obtained only by limiting the value of ρ under the higher mean offered load ρ (See Fig 4 12) Oppositely, because the probability distribution of queue length is higher in the " sensitive region " when $0\ 9 \leq \rho \geq 1\ 0$ Therefore, by increasing the buffer size and reducing the mean offered load, the cell loss probability is evidently decreased in the region , especially when ρ is lower (See Fig 4 13)

Fig 4 14 - 4 15 shows the effect of buffer length and a range of different load values To satisfy the cell loss probability of less than 10^{-6}, ρ has to be less than approximately 0 9 and K greater than 200 cells

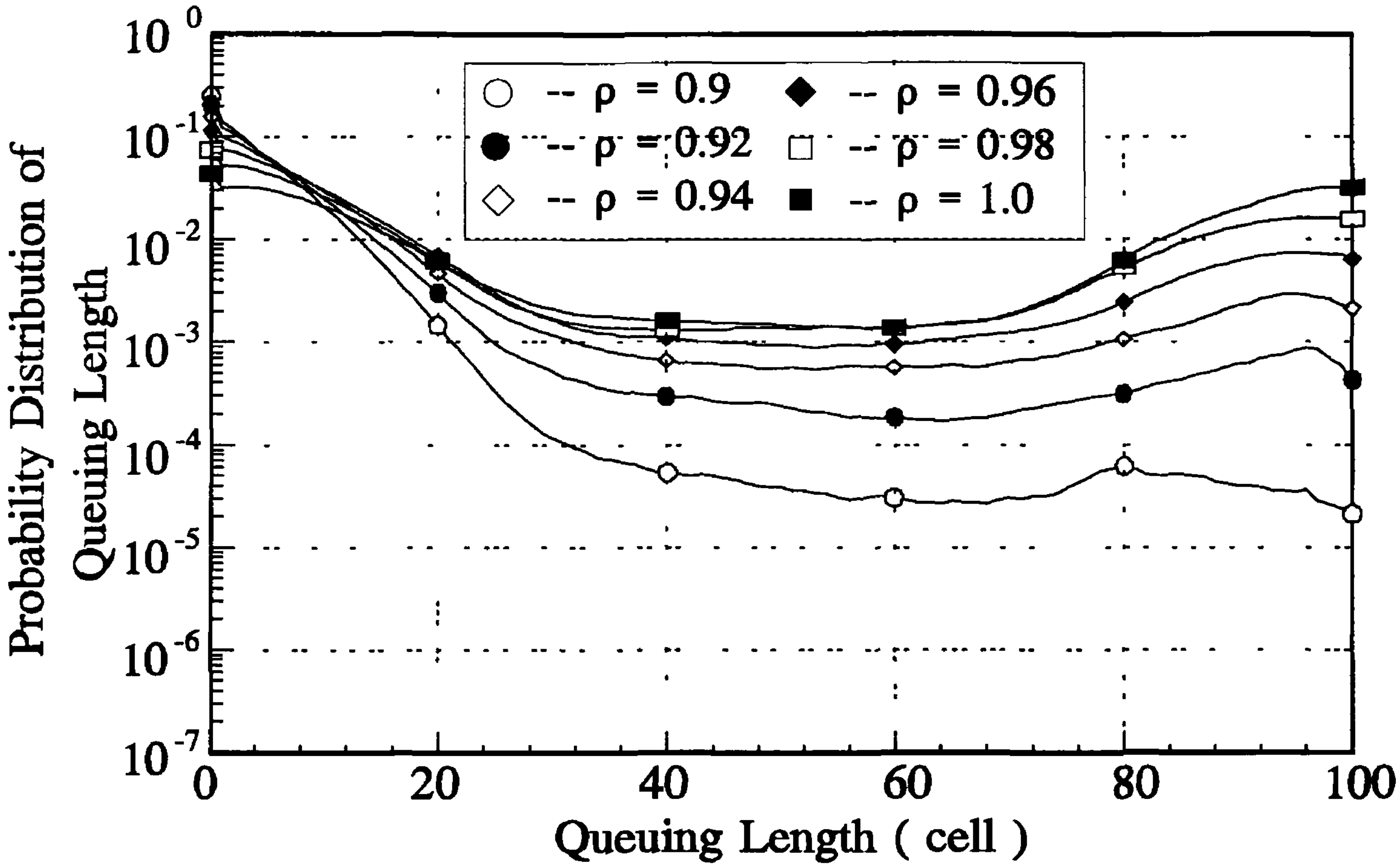


Fig 4 7 Probability distribution of queue length for various values of ρ and a buffer size equal to 100 cells

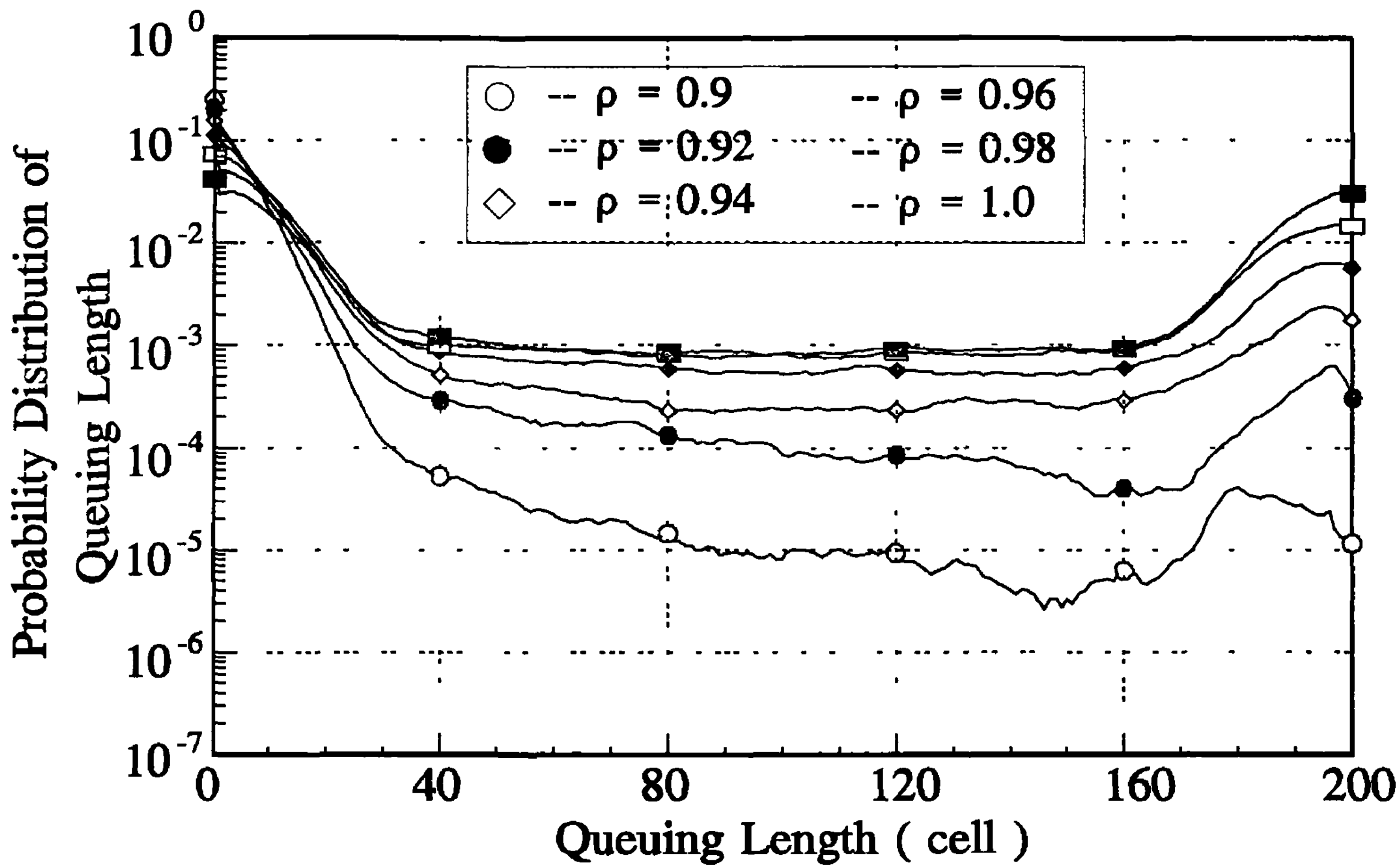


Fig 4 8 Probability distribution of queue length for various values of ρ and a buffer size equal to 200 cells

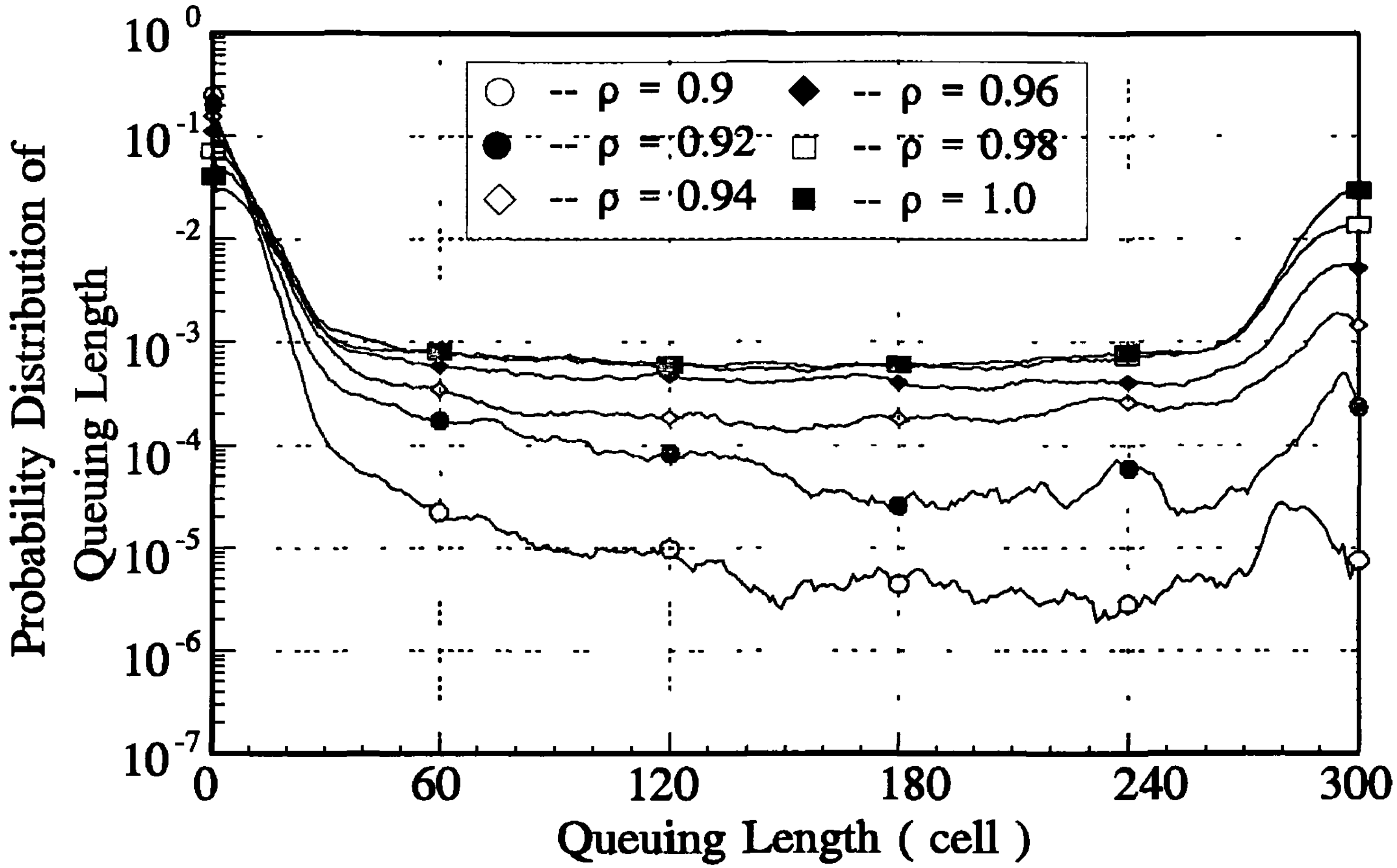


Fig 4 9 Probability distribution of queue length for various values of ρ and a buffer size equal to 300 cells

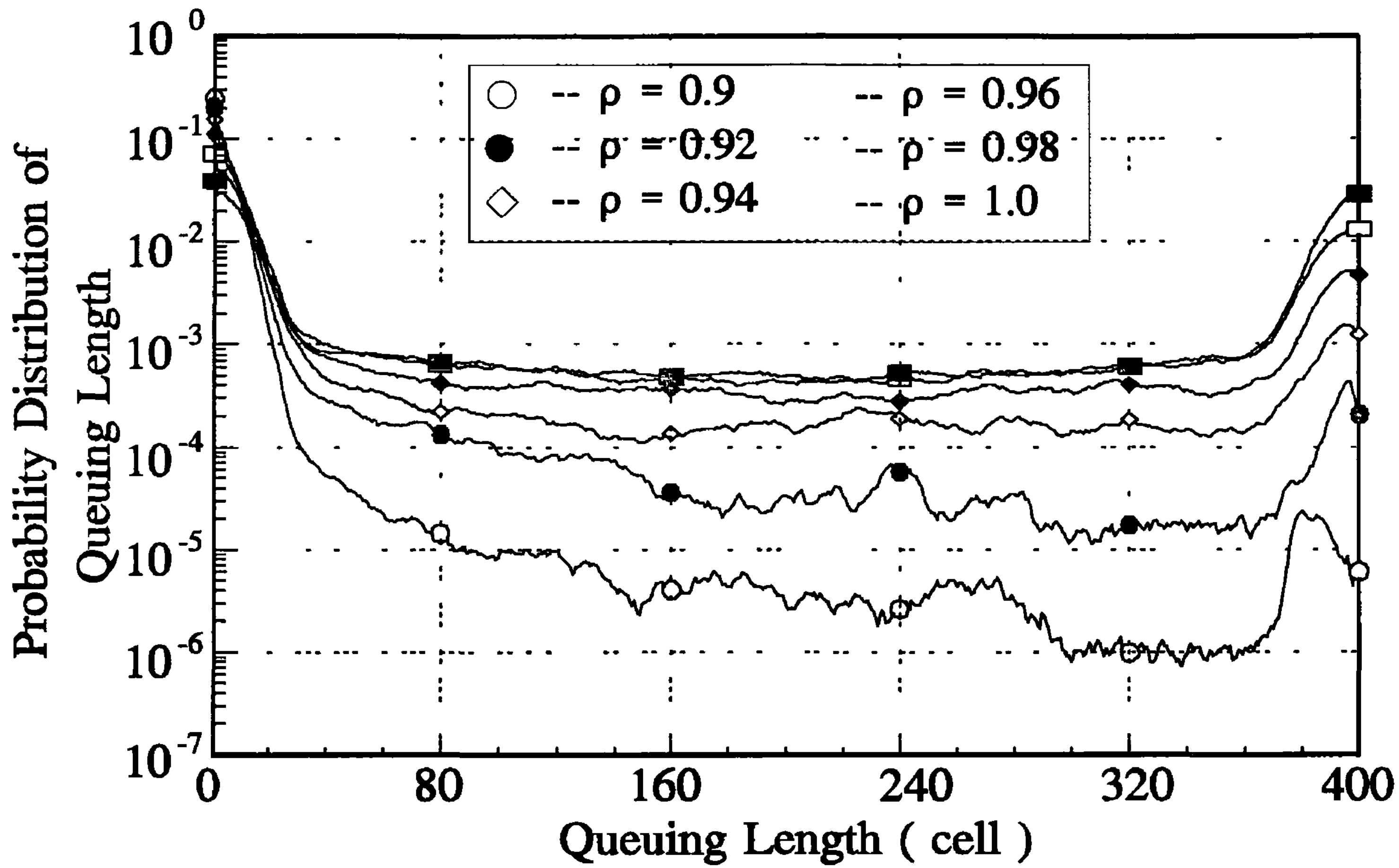


Fig 4 10 Probability distribution of queue length for various values of ρ and a buffer size equal to 400 cells

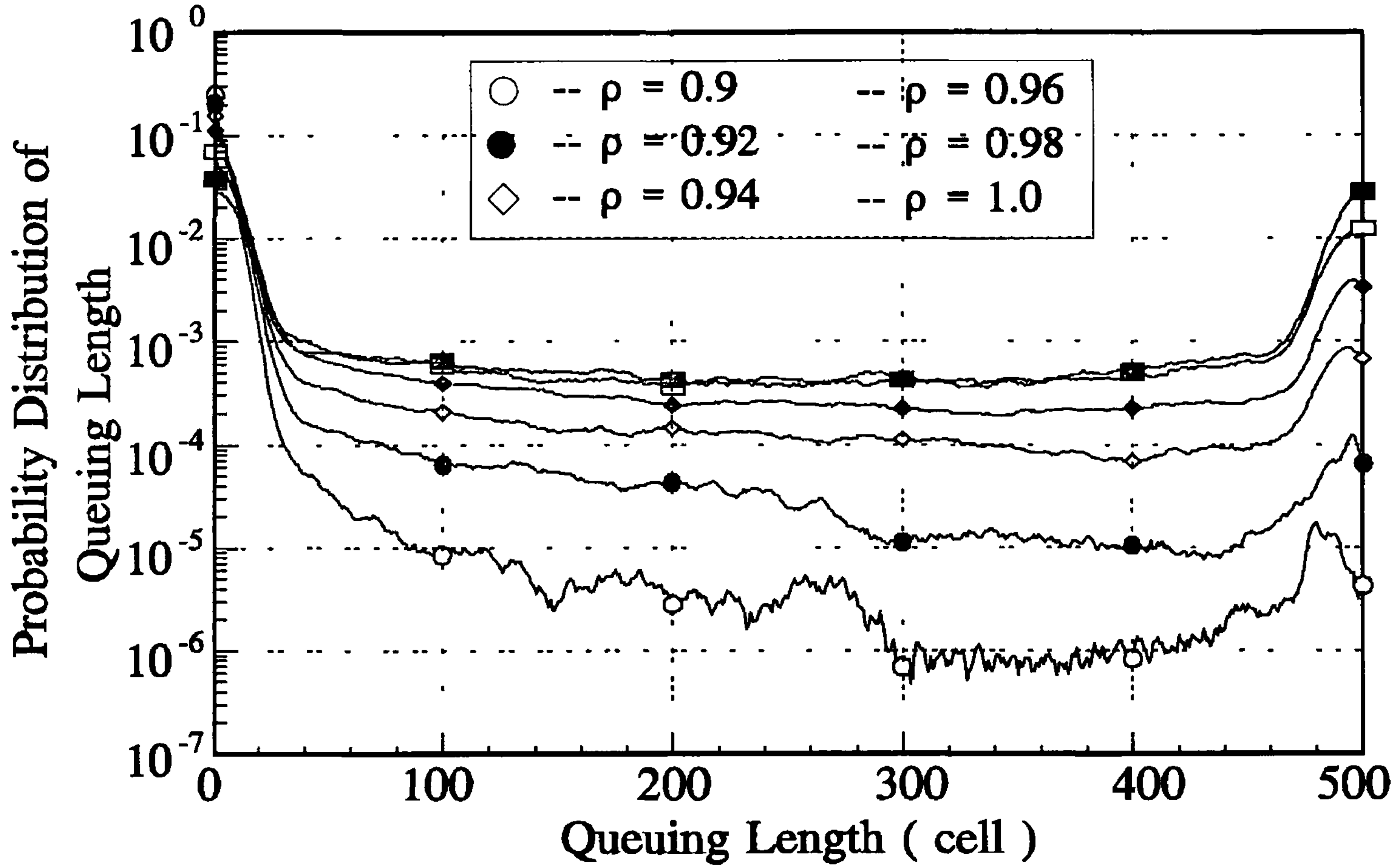


Fig 4 11 Probability distribution of queue length for various values of ρ and a buffer size equal to 500 cells

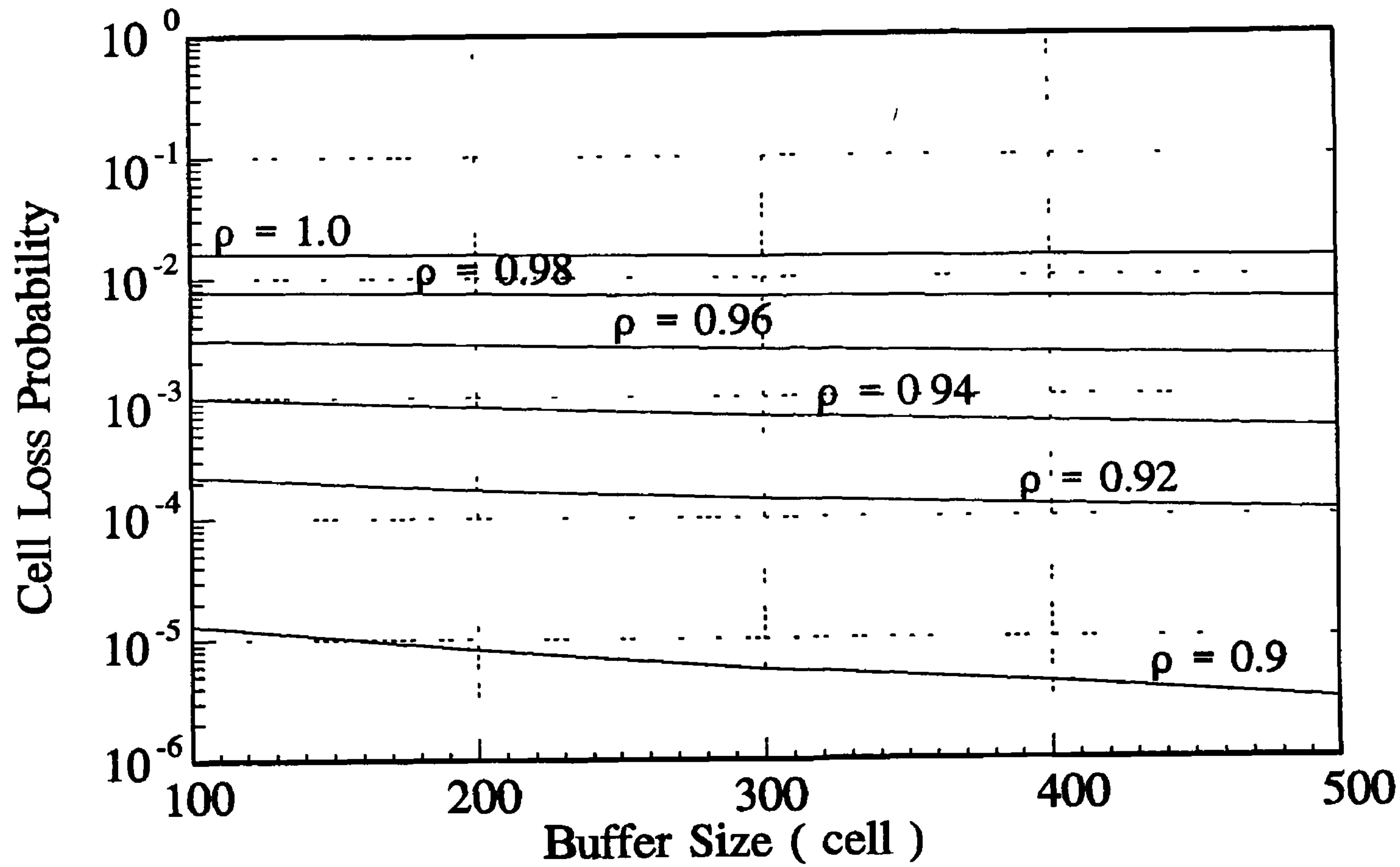


Fig 4 12 Cell loss Probability versus buffer size for various values ρ

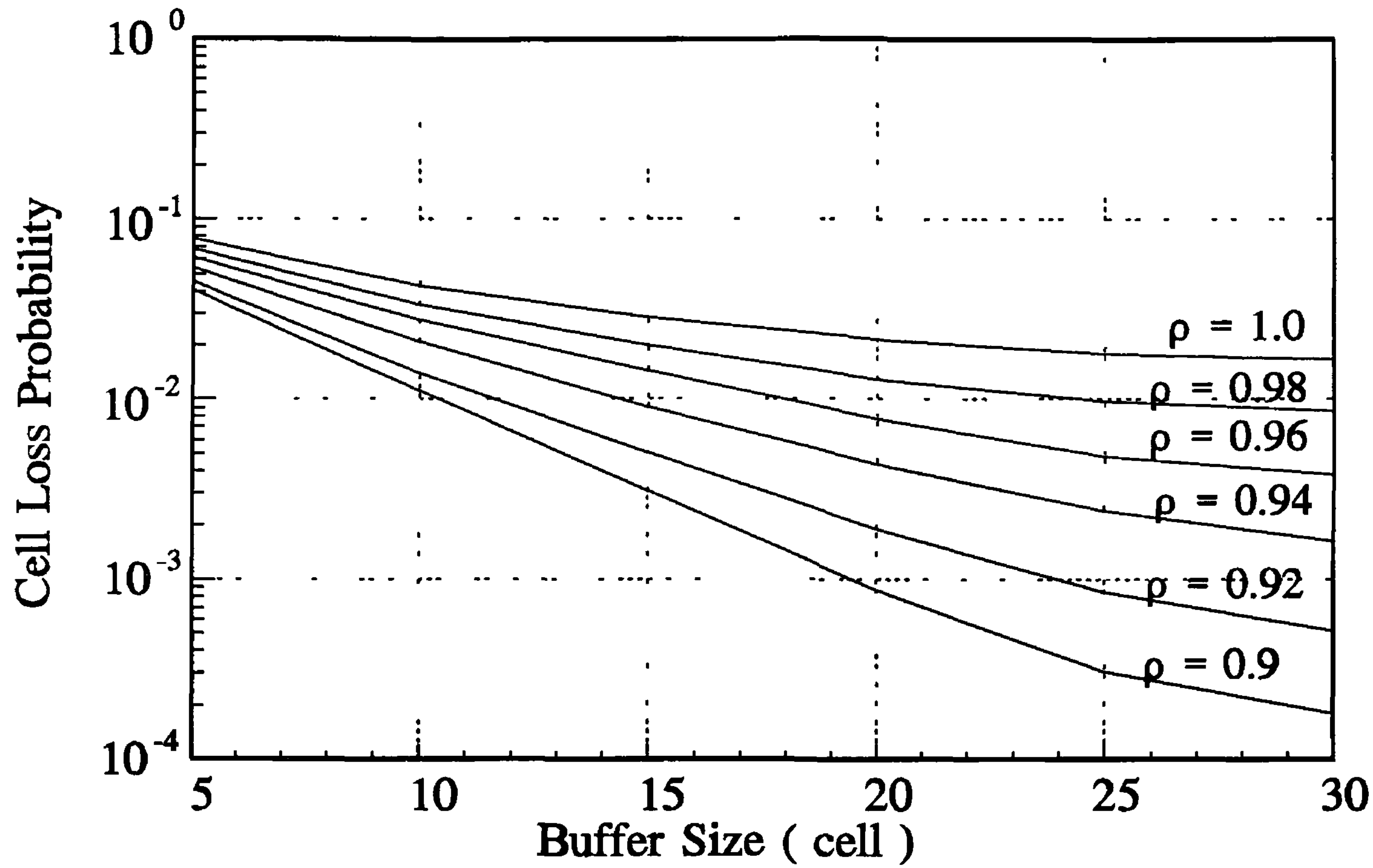


Fig 4 13 Cell loss Probability versus buffer size for various values ρ

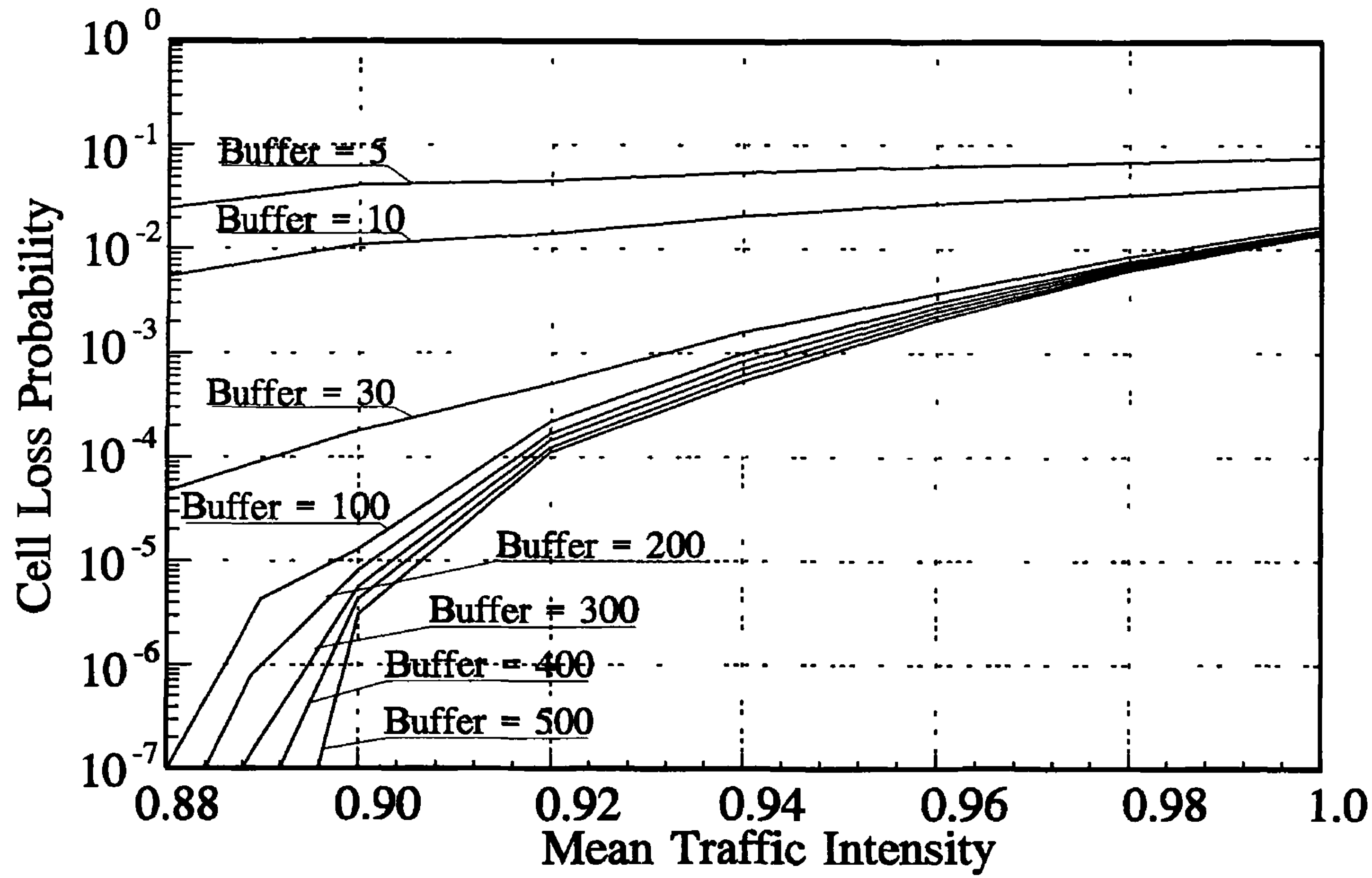


Fig 4 14 Cell loss Probability versus ρ for various values of the buffer size

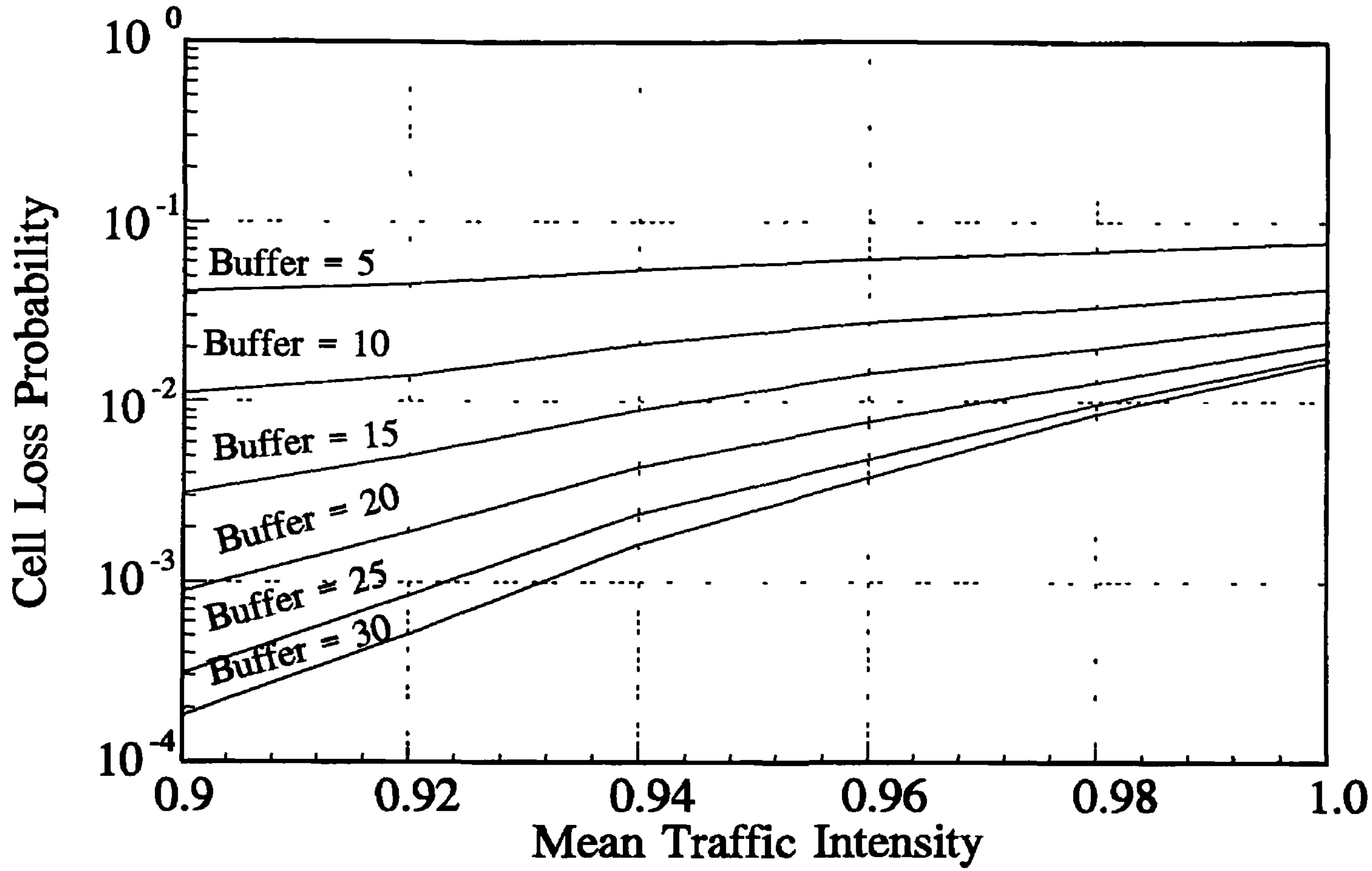


Fig 4 15 Cell loss Probability versus ρ for various values of the buffer size

Chapter 5 Simulation of ATM Multiplexer With Multiple QOS Classes

5.1 Introduction

It has been discussed in Chapter 4 that bursty sources of the same characteristics are statistically multiplexed to efficiently utilize the network resources in an ATM multiplexer In general, The ATM networks will provide a wide range of multimedia services with diverse traffic flow characteristics (e g , bit rate and burstiness) and quality of service (QOS) requirements Therefore, bandwidth management and traffic control are required to meet the QOS requirements of the various types of traffic[57,58,59,60] In studying the QOS, it has to be recognized that there will be various required values for cell delay time T_{max} (or $T_{average}$) and cell loss rate Γ in a multimedia environment Thus, two QOS control methods are possible to define a single QOS class and control the QOS of all the media information equitably, or to define multiple QOS classes and control each different classified medium individually In the first case, efficient use of the network resource is difficult since the most stringent QOS requirement should be selected as the network standard QOS In the second case, it will be shown in this chapter that the multiplexer gain can be improved by QOS control In addition, the users can select one of the classes appropriate to their requirements

In this Chapter, in order to design an economic and effective ATM multiplexer, the relationship between the traffic balance of classes and buffer size of each class is studied The cell loss probability and delay time of each class (same numerical distribution sources and different numerical distribution sources between the classes) are evaluated

5.2 The Architecture of an ATM Multiplexer with Priority Classes

The N sources are divided into three priority classes according to QOS requirements The sources of each class retain their bursty nature Traffic flow is a superposition of N_1, N_2, and N_3 ($N_1 + N_2 + N_3 = N$) statistically identical and independent sources, alternating between burst and silence periods These are assumed to be uniformly distributed with mean values λ_1(λ_2, λ_3) and μ_1(μ_2, μ_3), respectively Initially burst / silence state distribution is generated at random, from a predetermined probability distribution ρ_1 (ρ_2, ρ_3) The offered load to each class is denoted by A_1, A_2, and A_3, respectively The offered load on each input source of these three Classes therefore becomes

$A_1' = A_1/N_1$, $A_2' = A_2/N_2$, and $A_3' = A_3/N_3$, respectively Under this assumption, the buffer configuration is shown in Fig 5 1 The buffer length distribution of each class is mutually independent The buffer size of each class is selected according to the QOS requirements of each class In this study, only three FIFO buffers share a single server, in that there are only three QOS classes in this ATM multiplexer The cells of each class arrive independently from the N_1, N_2, and N_3 input sources and are distributed to each of the corresponding class buffers At each buffer, the cells are transmitted according to the FIFO discipline In one cell slot time of the output channel, only a single cell from among the three classes can be transmitted according to the priority assignment control (see Section 5 3) The server models a slotted output channel with a fixed capacity of C = 150 Mbits/s and a fixed cell size of D = 512 bits/cell The service rate is denoted by $u = C/D \approx 293000$ cells/s

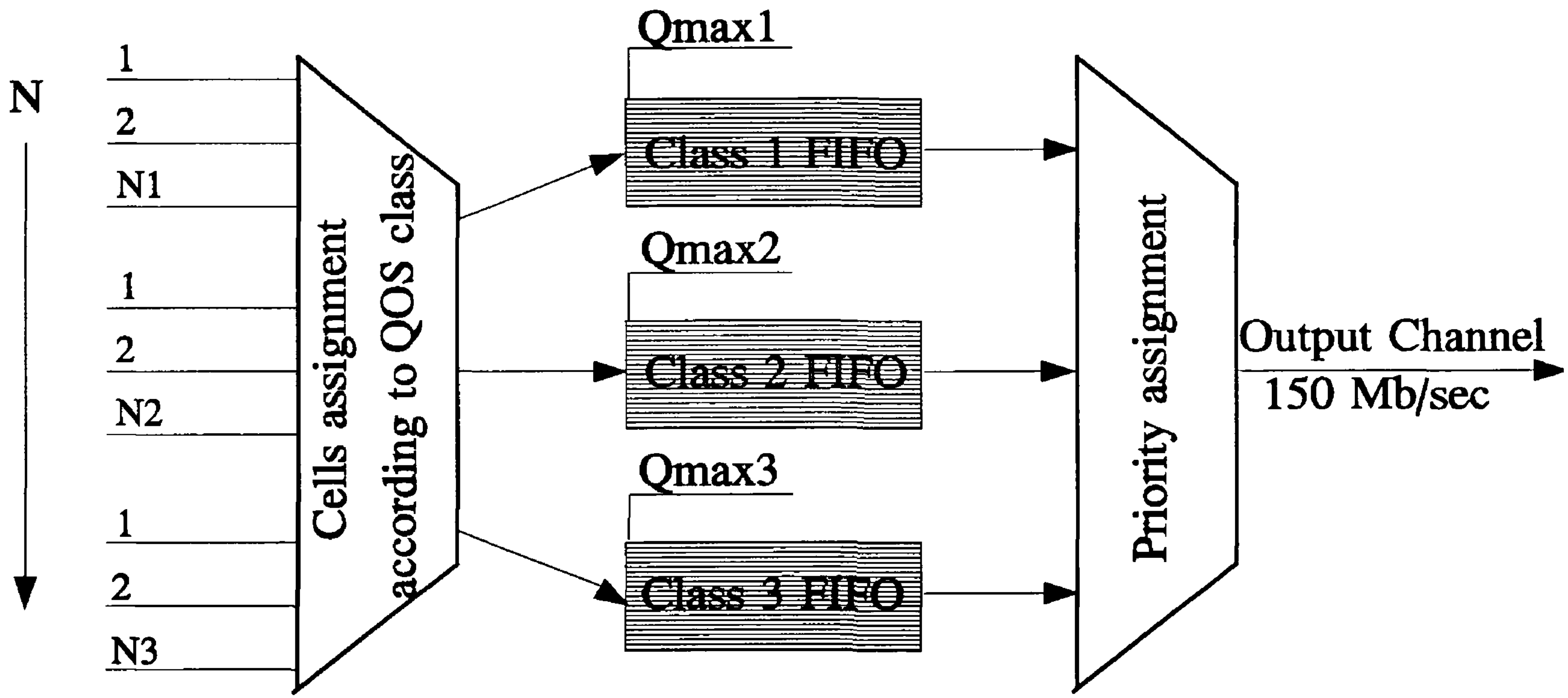


Fig 5 1 The architecture of an ATM multiplexer with priority classes

5.3 Priority Assignment Control Method

In a multimedia environment, the multiplexer gain can be improved by defining multiple QOS classes and using a QOS control which manages the required QOS of each class individually Priority assignment control is presented as a QOS control method in an ATM multiplexer This method has been discussed in[1] It has been confirmed in [1] by simulation that this control method is more effective when the interclass quality difference is bigger and the burstiness of the traffic is stronger in [1] Moreover, the control scheme to enlarge the actual admissible offered load region by adaptively changing the priority assignment ratio has also been presented In this control method, the transmission bandwidth (output cell slots) through which cells are transmitted with the highest priority is assigned to each class in a priority assignment period P Each class can be given various QOS values by changing the priority assignment ratio W_n, which determines the minimum output bandwidth of each class

1) Priority assignment ratio (W_1 W_2 W_n)
   This parameter is the most predominant parameter in controlling the QOS of each class and should be determined according to the required QOS of each class and the traffic balance of the classes The minimum output bandwidth of each

class is guaranteed by the priority assignment ratio

2) Priority assignment period P (cell slots) The priority slots of each class will be distributed in the period P To efficiently assign the priority slot to each class, the value of P should be a multiple of sum of the priority assignment ratio of each class ($W_1 + W_2 +$ $+ W_n$) The priority slots of each class should be distributed as periodically as possible in the period P in order to reduce the output burstiness of each class for the next switching node

3) Priority scheduling in each cell slot When more than two QOS classes are multiplexed, if there are no waiting cells of the class that has the highest priority at the slot, a cell of one of the other classes can be transmitted This allows improvements in bandwidth efficiency compared to a segregation of bandwidth among the separate queues The priority scheduling is shown in Fig 5 2

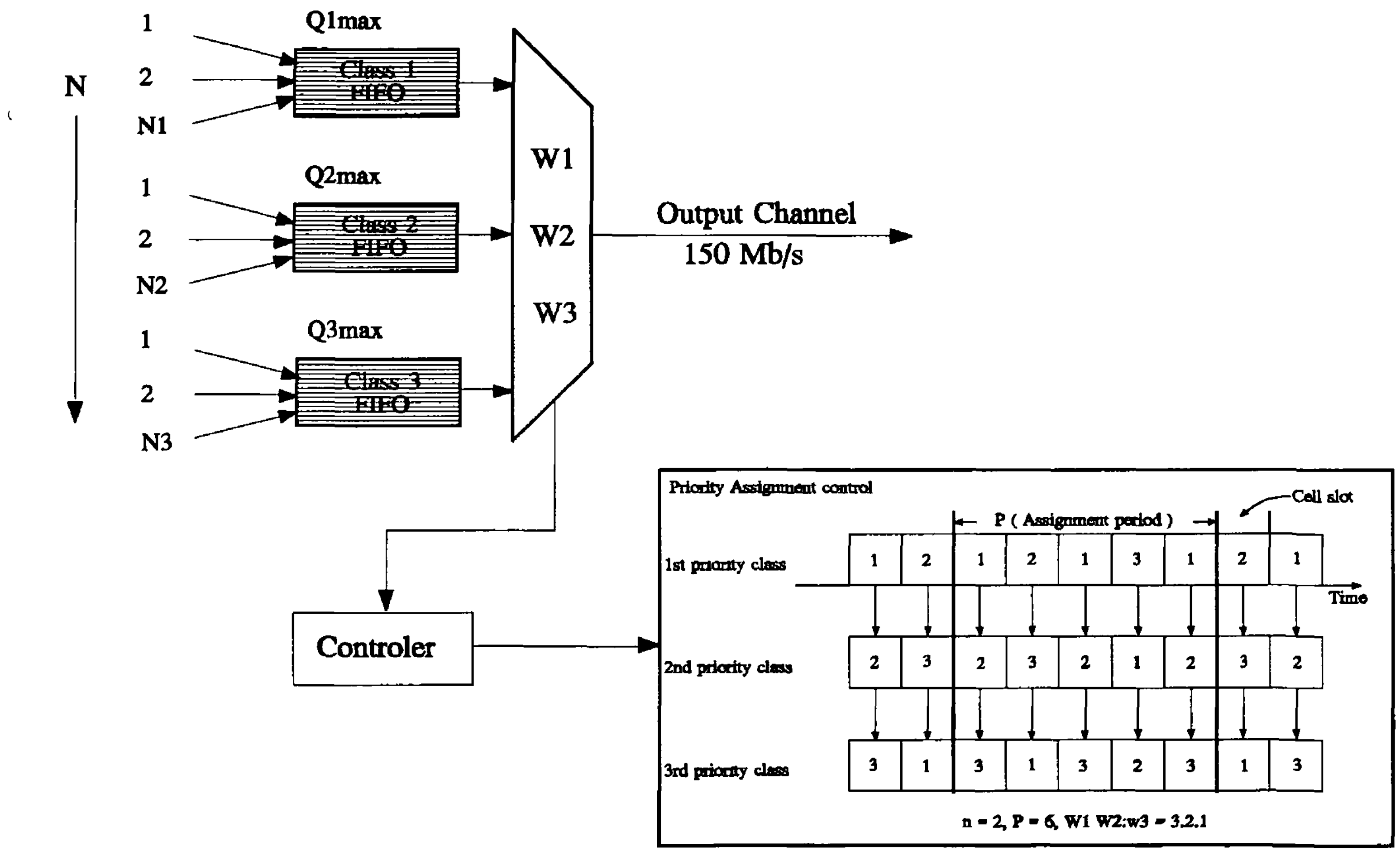


Fig 5 2 Priority assignment control method

In this control method, the maximum delay time of each class is not equal to the upper limit value of queue length It not only relates to the upper limit of queue length, but also relates to the priority assignment ratio In other words, the maximum delay time of each class is guaranteed by the priority assignment ratio and the upper limit of the queue length It is assumed that corresponding to the priority assignment ratio W_1 W_2 W_n, the upper limit of the queue length of each class Q_1, Q_2, Q_n will be determined so that the maximum delay time in each class D_1, D_2, D_n (cell slot) will meet the required value as follows[1]

$$Q_1 = D_1 * W_1 / (W_1 + W_2 + \quad + W_n)$$

$$Q_2 = D_2 * W_2 / (W_1 + W_2 + \quad + W_n)$$

$$Q_n = D_n * W_n / (W_1 + W_2 + \quad + W_n)$$

When the queue length exceeds the upper limit value, arriving cells are discarded If the priority assignment ratio is changed, the upper limit value of the queue length must also be changed

The flow diagrams illustrating this priority assignment control method are shown in Figs 5 3 - 5 4

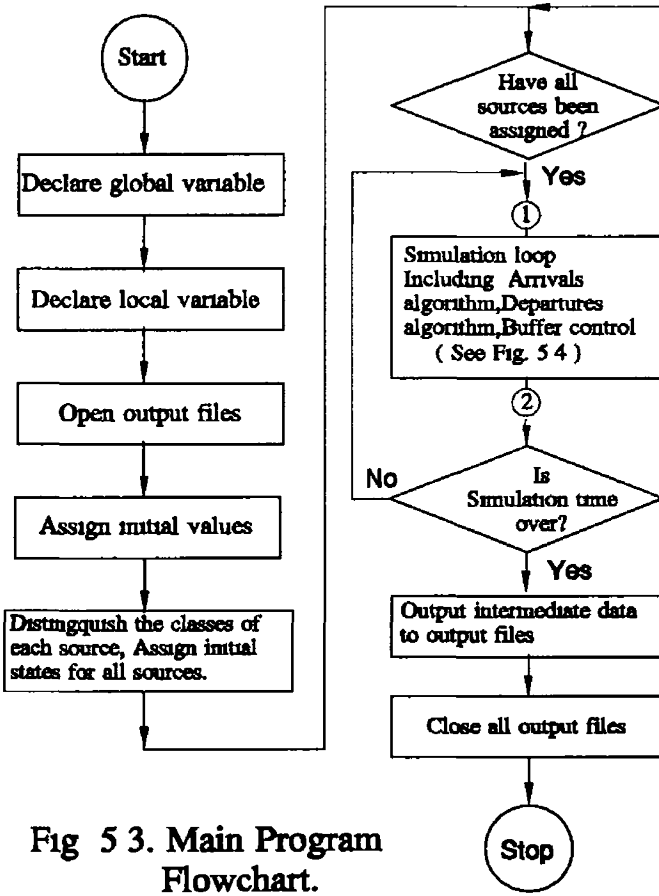
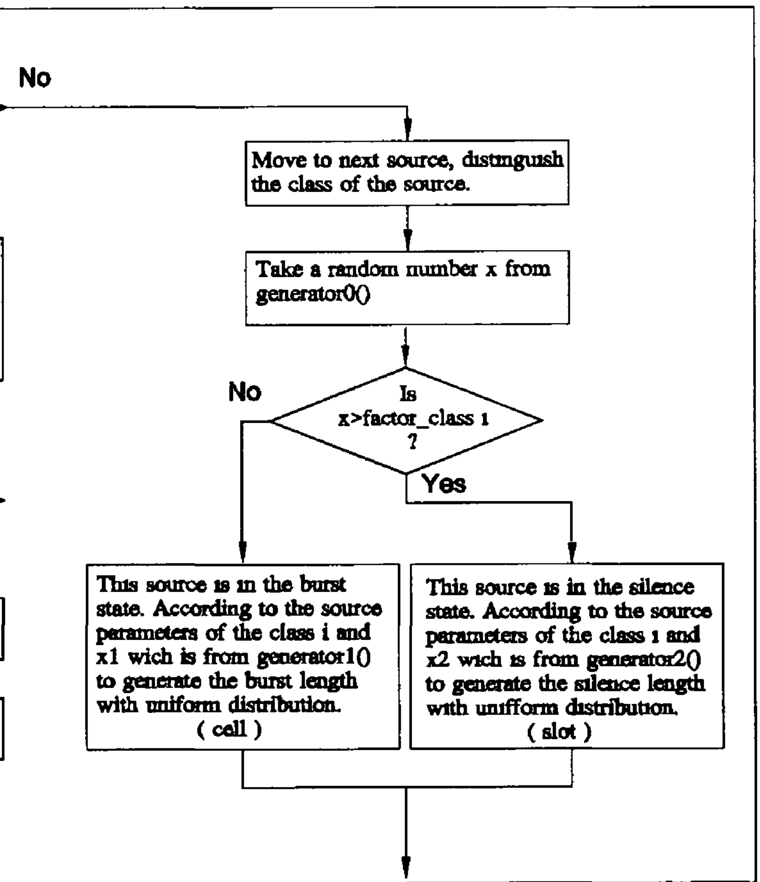
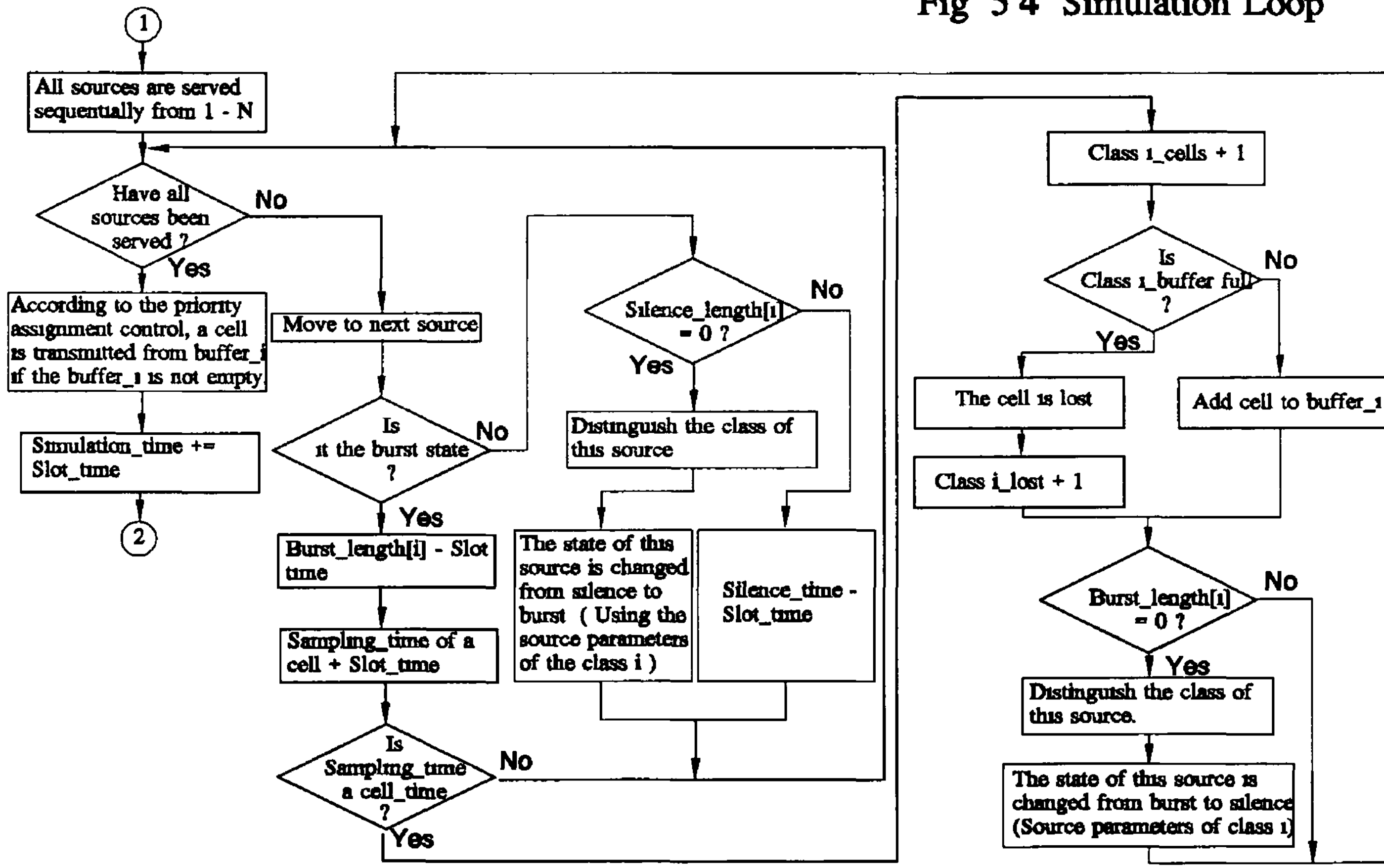


Fig 5 3. Main Program Flowchart.



No
Move to next source, distinguish the class of the source.
Take a random number x from generator0()
Is x>factor_class i ?
No
Yes
This source is in the burst state. According to the source parameters of the class i and x1 wich is from generator1() to generate the burst length with uniform distribution.
(cell)
This source is in the silence state. According to the source parameters of the class i and x2 wich is from generator2() to generate the silence length with unifform distribution.
(slot)

Fig 5 4 Simulation Loop



5.4 Performance Results

The results of the performance parameters simulated are specifically focused on the cell loss probability and cell delay time at the buffer of each class in this study All of the simulations reported in the Chapter are based on the assumptions of a fixed cell size of D = 512 bits/cell , the average bit rate of each source A_0 = 150 kb/s, and an output capacity of C = 150 Mb/s In addition, The priority assignment period P is equal to 6 cell slots and the priority assignment ratio of each class is $W_1 = 3$, $W_2 = 2$, and $W_3 = 1$ The simulations were run for 100 seconds Two different input categories are considered in this priority multiplexing scheme

Type 1 input category

The reference sources of each class have the same parameter values, and the average offered loads of each class A_1, A_2, and A_3 have the same values, i e , $A_1 = A_2 = A_3$ These values are

a The peak bit rate of each source F_p = 375 kb/s

b Average bit rate of each source F_a = 150 kb/s

c The burst activity factor $\rho = F_a / F_p = 0\ 4$

d Average burst length L_b = 300 ms

e Average sılence length L_s = 450 ms

f The buffer sızes of each class have the same values, ı e , 10 cells (or 15 cells)

Usıng these parameter values sımulatıon, of the ATM multıplexer wıth prıorıty classes has been carrıed out The sımulatıon results are presented ın Fıg 5 5 - 5 6

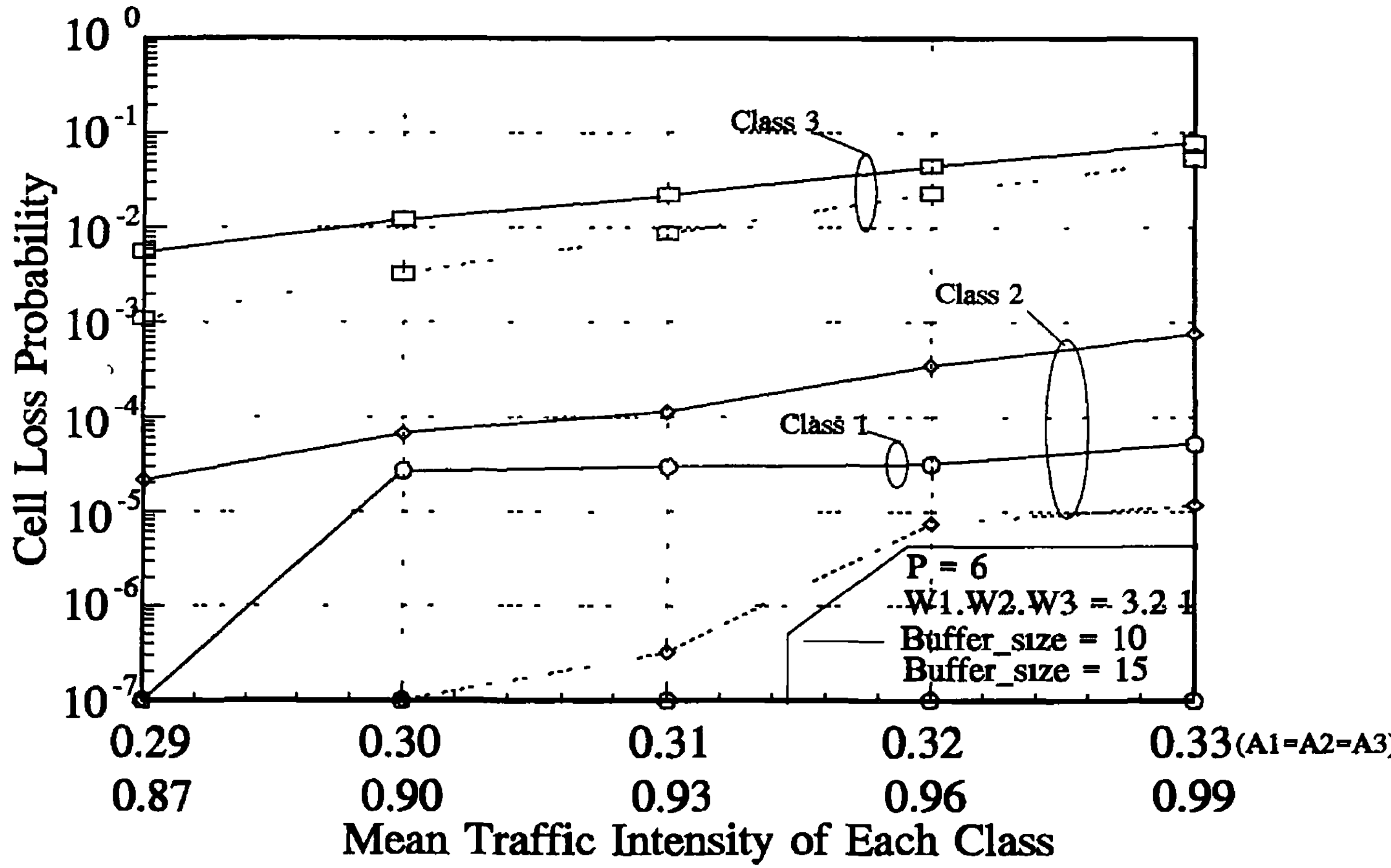


Fig 5 5 Cell loss probabilities of each class versus A ($A = A_1 + A_2 + A_3$) where $A_1 = A_2 = A_3$

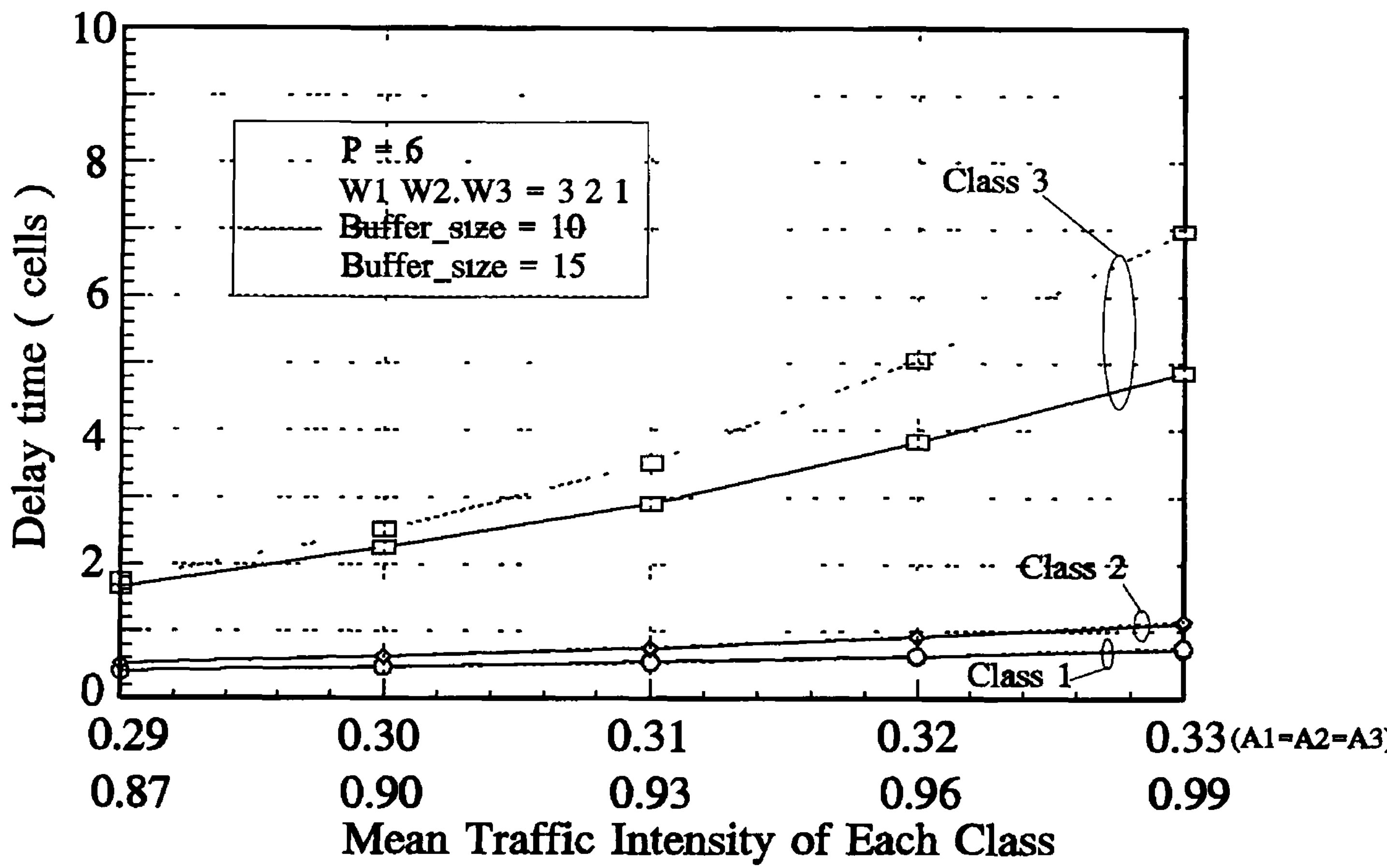


Fig 5 6 Delay times at each buffer versus A $(A = A_1 + A_2 + A_3)$ where $A_1 = A_2 = A_3$

The cell loss probability versus the buffer size (10 cells and 15 cells), and with different values of $A = A_1 + A_2 + A_3$ ($A_1 = A_2 = A_3$) is shown in Fig 5 5 This figure shows that when the buffer size of each class changes between 10 and 15 cells, then the cell loss probability of each class changes accordingly With higher priority classes, the change becomes more obvious Usually, the cell loss probability of each class steadily became higher with the increase of the average offered load of these three classes, but, after the total average offered load A reaches about 0 9, the cell loss probability Γ_1 remains approximately constant(the change of the cell loss probability is very little) The point of 0 9 is called "the critical point", and the right of the critical point is called "the steady region" In the region, Only by increasing K_1 can the cell loss probability Γ_1 be evidently decreased This is because class 1 has the highest priority of being served on the output channel When A is less than 0 9(A_1 < 0 3), the bandwidth allocation of class 1 is not enough the After 0 9, despite the fact that the increasing A_1, A_2 and A_3, Γ_1 will remain approximately constant at a cell loss probability of 3 119600e-05 for a buffer size of is 10 cells (it is zero when the buffer size is reduced to 15 cells) Of course, if A_1 is too big, the cell loss probability Γ_1 will be increased apparently

As is shown in Fig 5 6, the average delay time varies significantly with the buffer size at the lower priority class, while the variations of the average delay times for the different buffer sizes at higher priority classes are not apparent These phenomena can be explained by the fact that the probability distribution of longer queue lengths is generally higher for the lowest priority class, especially under higher average offered loads For the highest priority classes the probability distribution of smaller queuing length is usually higher, particularly under lower average offered load (See Fig 5 7)

It has been shown in Chapter 4 that the probability distribution of queue length follows a bimodal behaviour in the statistical multiplexing without priority class Such a phenomenon also happens when the priority classes are introduced as shown in Fig 5 7

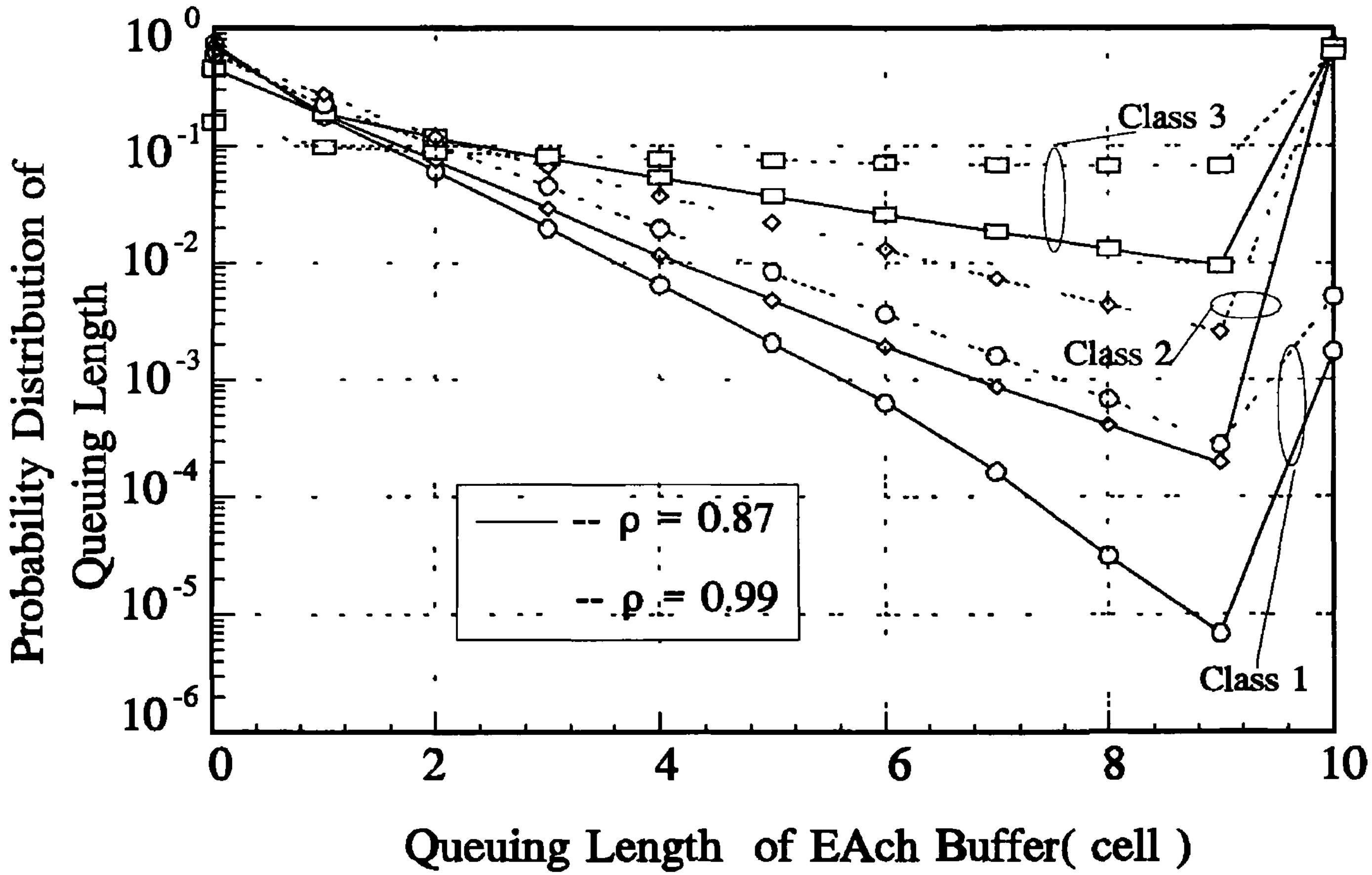


Fig 5 7 Probability distribution of queuing length for A =0 9, 0 99 ($A = A_1 + A_2 + A_3$, and $A_1 = A_2 = A_3$, the buffer size $K1 = K2 = K3 = 10$ cells

Type 2 input category:

The reference sources of each class are the same as in Type 1, but the average offered load (A_1, A_2, and A_3) of one of these three classes is a constant.

It is assumed that A_1 = 0.4 (or A_2 = 0.4, A_3 = 0.4), and then the average offered loads of other two classes are increased gradually. The results are shown in Fig. 5.8 - 5.13. The increasing average offered loads of any two classes of these three classes will lead to an increase in the total average offered load A. Generally speaking, the cell loss probabilities of each class is increased, and the delay times of each class is also increased(See Fig. 5.8 - 5.13). In Fig. 5.8, the average offered load of class 1 is a constant, i.e., A_1 = 0.4. The cell loss probability of class 1 (Γ_1) steadily became higher with the increase of the average offered load of the other two classes, A_2 and A_3. But, after the total average offered load A reaches about 0.92 (i.e., "the critical point" is 0.92.), Γ_1 will remain approximately constant at a cell loss probability of 8.332914e-05 for a buffer size of is 15 cells (it is about 1.269284e-03 when the buffer size is reduced to 10 cells). However, The value of the constant depends to a large extent on The priority assignment period P, the priority assignment ratio of each class, and the value of A_1 when the total average offered load A reaches "the critical point" (See Fig 5 10, 5 12)

In Fig 5 9, 5 11, 5 13, it is also shown that the variations of the average delay times for the different buffer sizes at higher priority classes(or at lowest priority class when A_3 is smaller) are not apparent

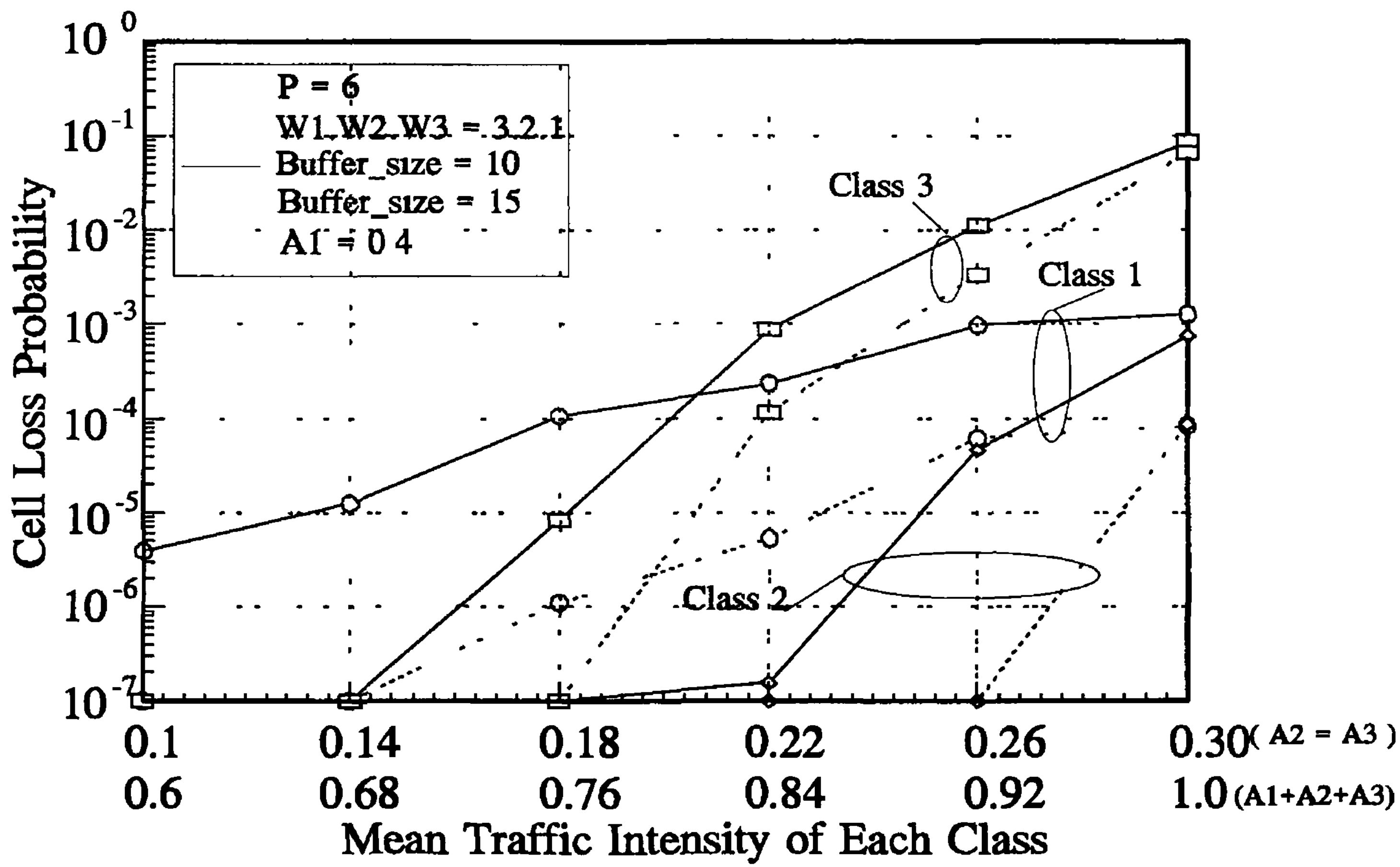


Fig 5 8 Cell loss probabilities of each class versus A ($A = A_1 + A_2 + A_3$) where A_1 is a constant

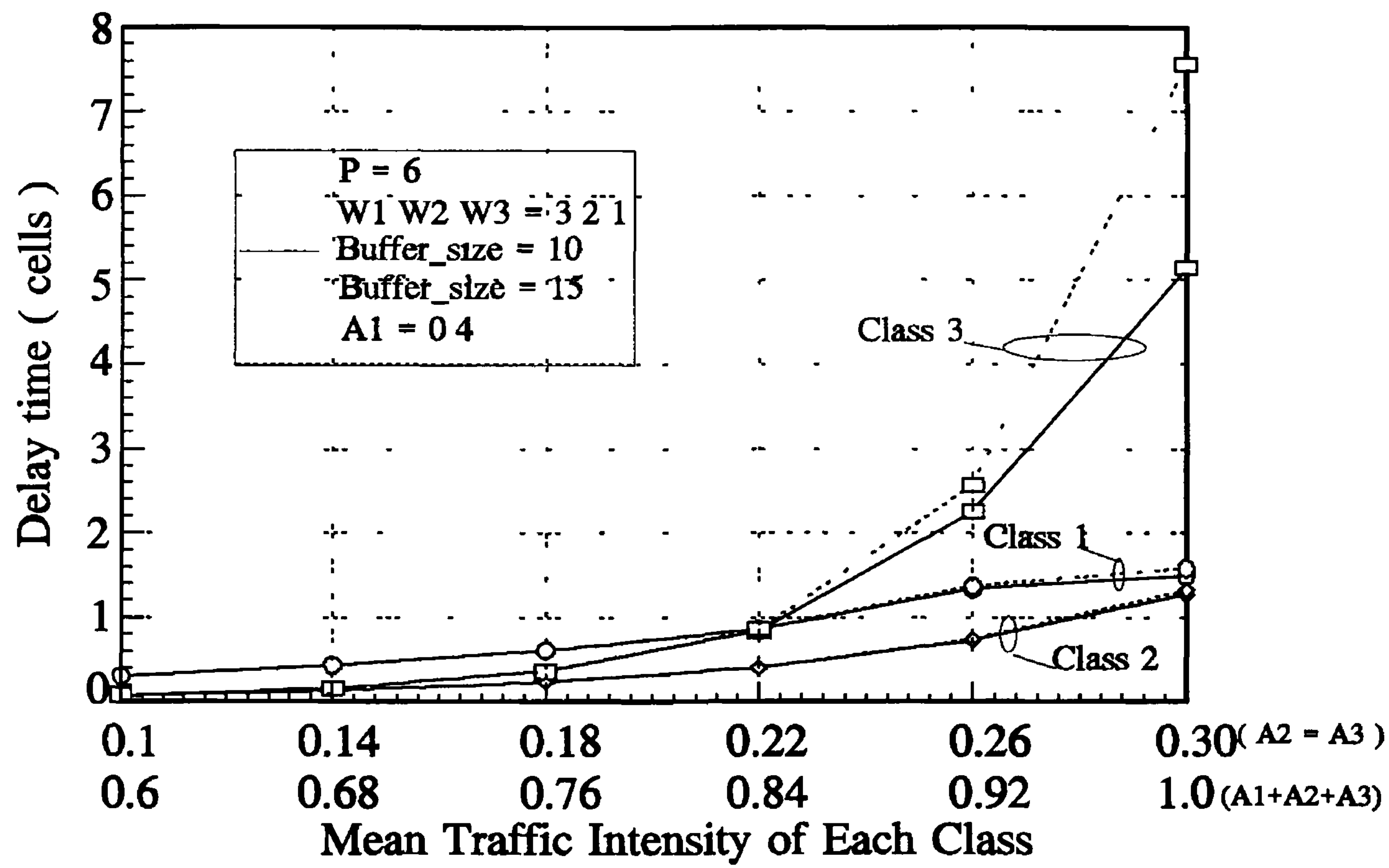


Fig 5 9 Delay times at each buffer versus A ($A = A_1 + A_2 + A_3$) where A_1 is a constant

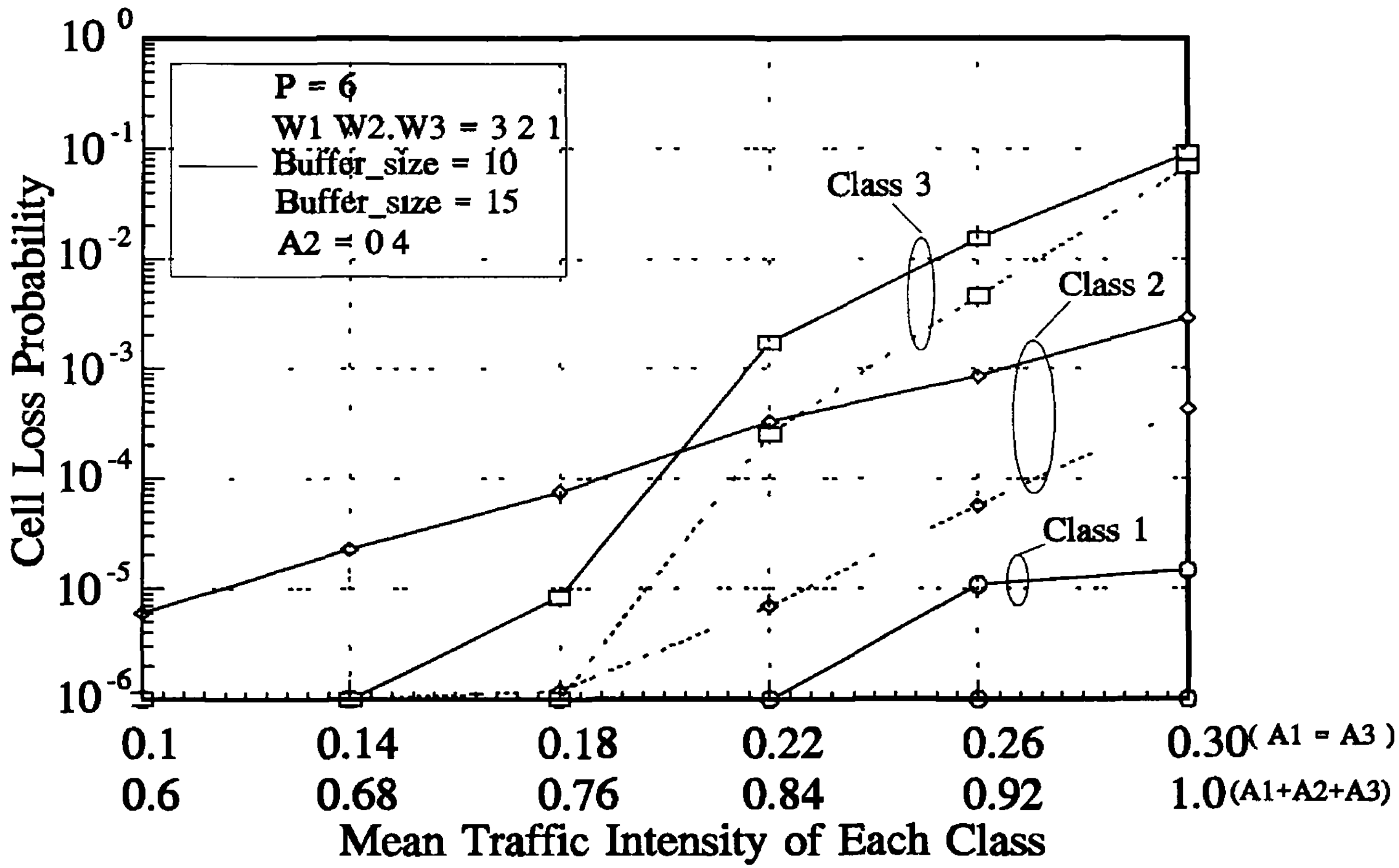


Fig 5 10 Cell loss probabilities of each class versus A ($A = A_1 + A_2 + A_3$) where A_2 is a constant

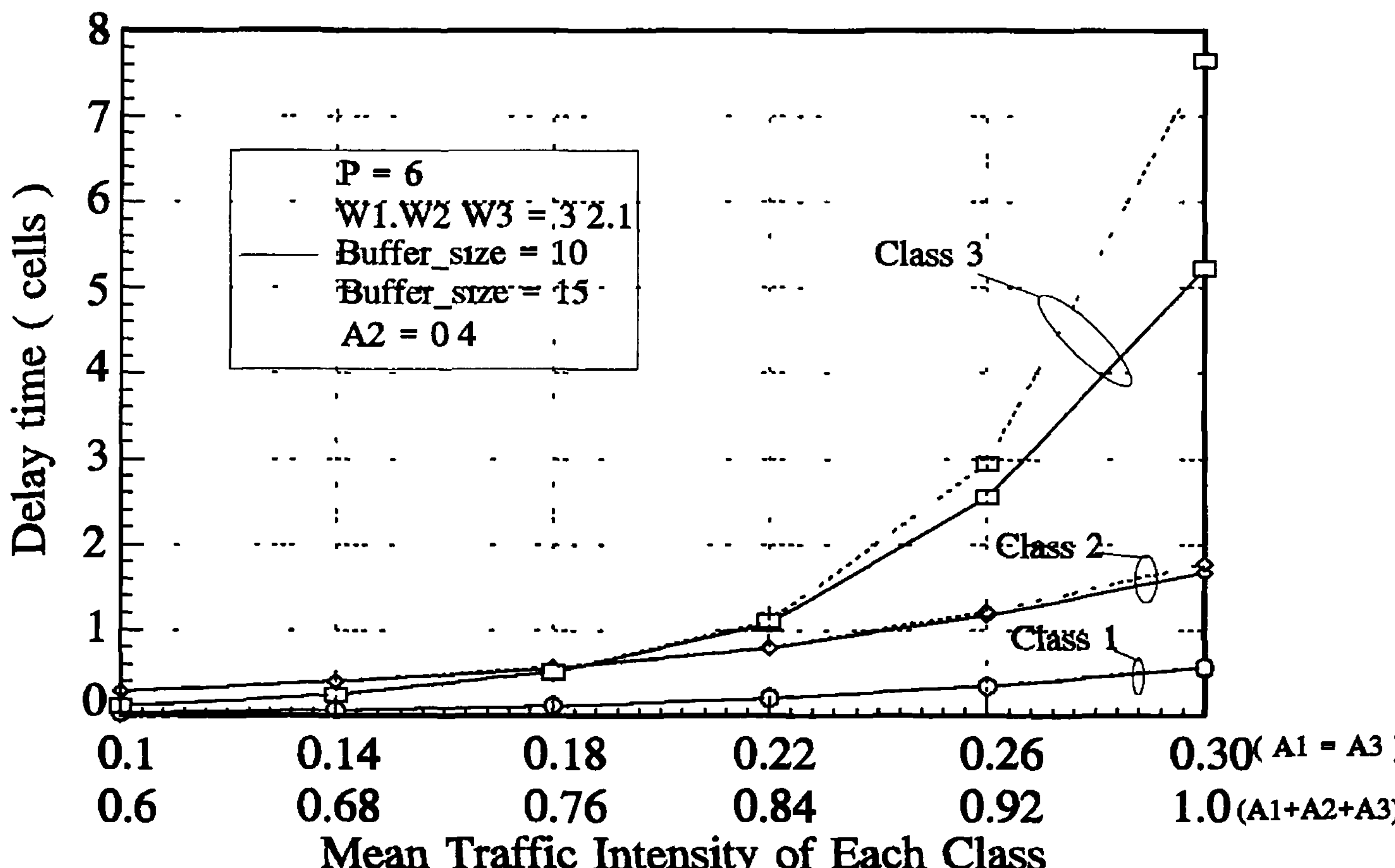


Fig 5 11 Delay times at each buffer versus A ($A = A_1 + A_2 + A_3$) where A_2 is a constant

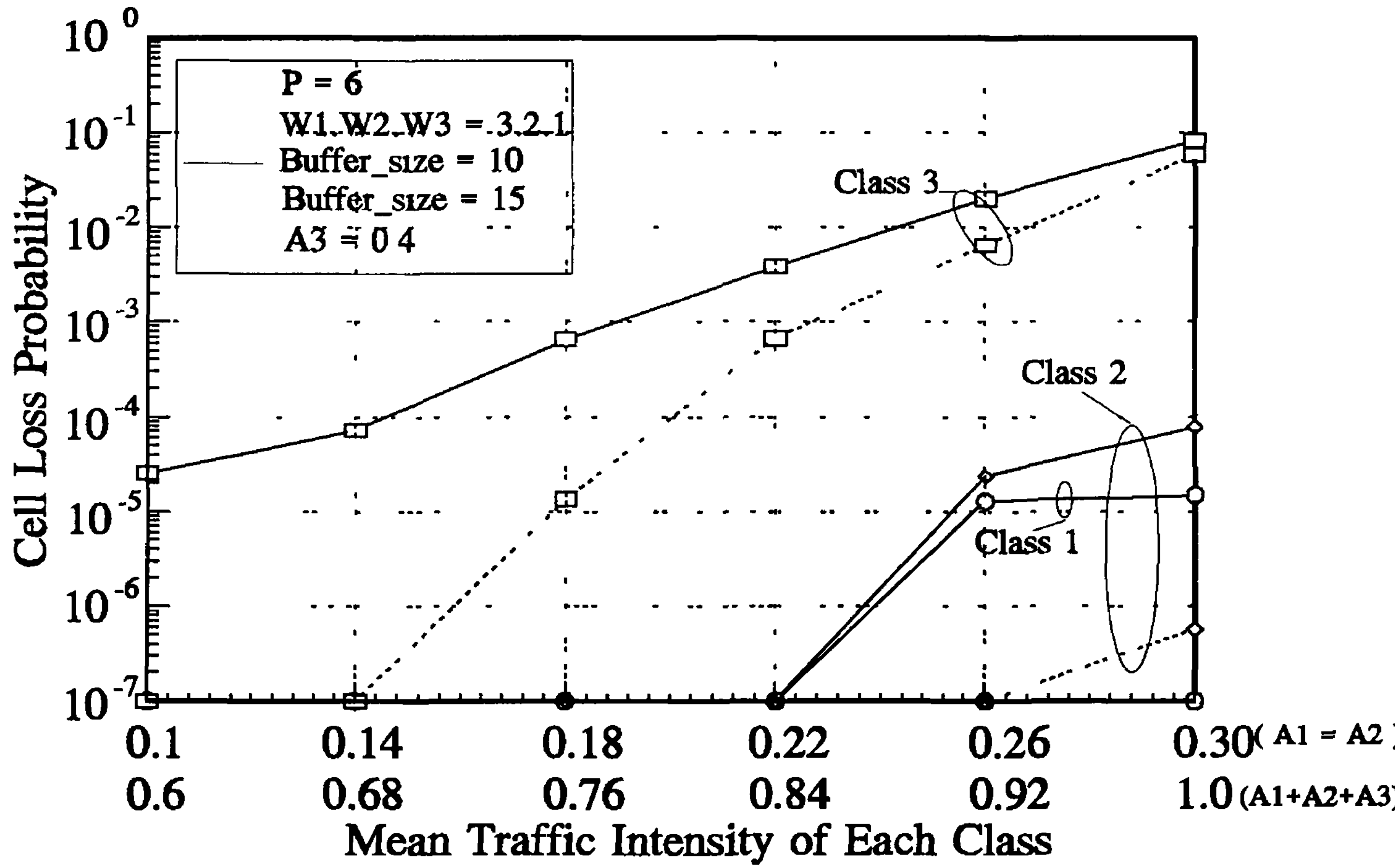


Fig 5 12 Cell loss probabilities of each class versus A $(A = A_1 + A_2 + A_3)$ where A_3 is a constant

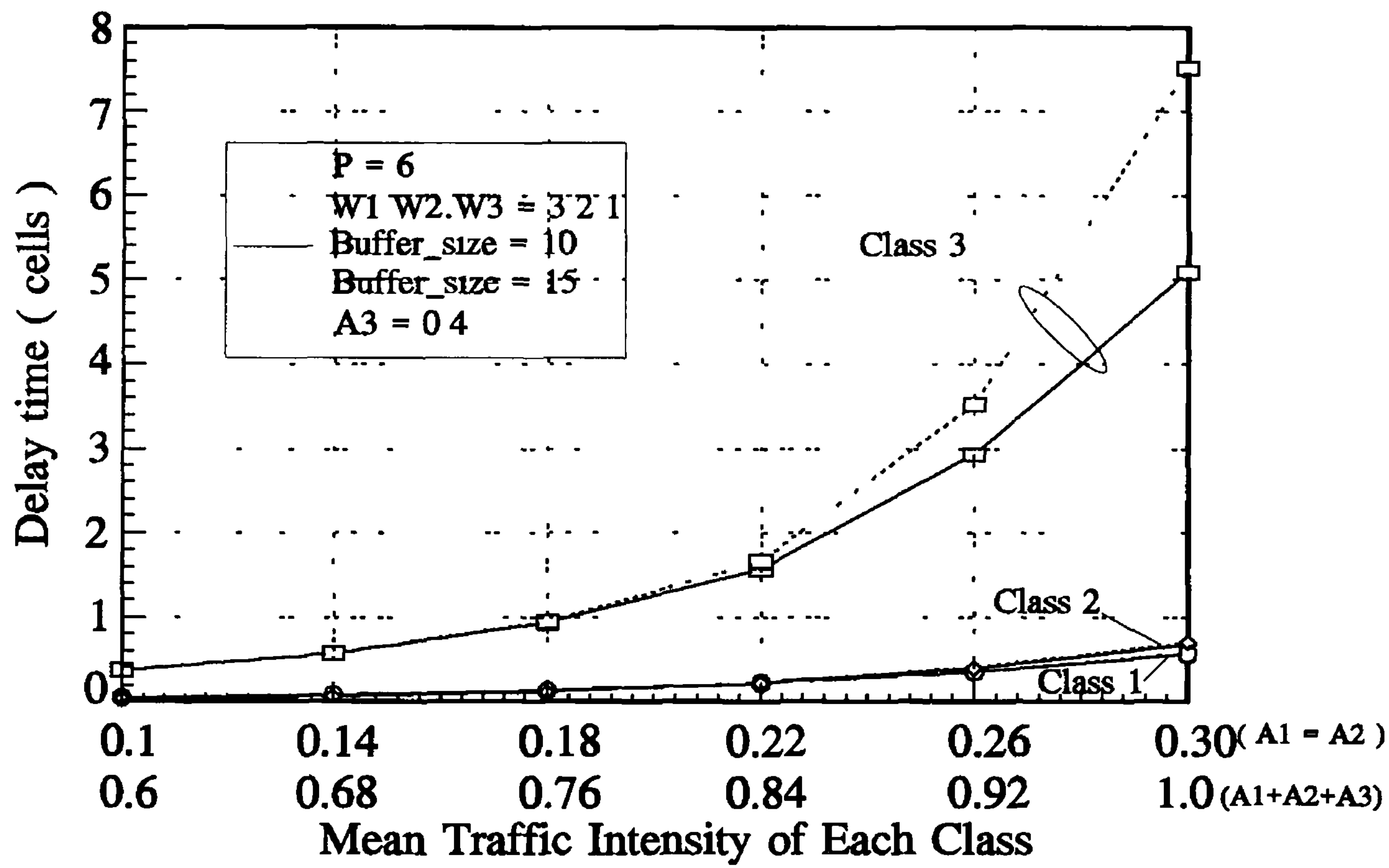


Fig 5 13 Delay times at each buffer versus A $(A = A_1 + A_2 + A_3)$ where A_3 is a constant

Chapter 6 Conclusions

The specific focus of this thesis has been on the performance of an ATM multiplexer for bursty sources There are two themes in the thesis 1) simulating an ATM multiplexer without priority class for bursty sources 2) simulating an ATM multiplexer with three priority classes for bursty sources All of the simulations reported are based on the assumptions of a fixed cell size of D = 512 bits/cell, an average bit rate of each source of A_0 = 150 kb/s, a multiplexer output capacity of C = 150 Mb/s

In the first theme, The distribution of the queue length, for different Buffer sizes and values of the average offered load ρ has been analyzed Thus, a inherent phenomenon of the distributions of the queue length has been brought to light in statistical multiplexing ------ a bimodal behaviour However, a threshold value of the buffer size is found through observing the bimodal behaviour When the buffer size K is less than the threshold, the cell loss probability is apparently decreased by increasing K and reducing ρ, especially when ρ is lower But, changing K is practically ineffective for the cell loss probability when K is bigger than the threshold Such the effect is more pronounced under increasing ρ Only if K is increased up to extremely high values can the cell loss values be significantly reduced It should also be taken into account that a maximum delay T_{max} constraint implies an upper bound on K, for the chosen value of C As a consequence, the matching between the desired QOS and the performance parameters of the ATM multiplexer can be obtained only by limiting the value of ρ when K is bigger than the threshold these results are very useful in the design of an economic and effective ATM multiplexer

In the second theme, two different input categories, which share an ATM multiplexer with three priority classes respectively are considered using the priority assignment control method of [15] The relationship between the traffic balance of the three classes and buffer size of each class is studied The cell loss probability and delay time of each class are evaluated

In the first input category, the reference sources of each class have the same parameter values, and the average offered loads of each class A_1, A_2, and A_3 have the same values, i e , $A_1 = A_2 = A_3$ In the second input category, the reference sources of each class are the same as 1), but the average offered load (A_1, A_2, and A_3) of one of these three classes is a constant It is found out that these two different input categories have some generalities in the QOS parameters respect(the cell loss probability and delay time of each class)

For the cell loss probability, when the buffer size of each class is increased, then the cell loss probability of each class must also be raised accordingly. With higher priority classes, the rate of increase becomes more obvious. Usually, the cell loss probability of each class steadily became higher with the increase of the average offered load of these three classes. But, after the total average offered load A reaches a certain value, which is called "the critical point", despite the increasing A, the cell loss probability Γ_1 remains approximately constant. The value of the constant depends to a large extent on the priority assignment period P, the priority assignment ratio of each class, and the value of A_1 when the total average offered load A reaches "the critical point". The right of the critical point is called "the steady region". In this region, only by increasing K_1 can the cell loss probability Γ_1 be evidently decreased. To satisfy the QOS requirements of each class and to allocate the bandwidth of each class effectively, the bandwidth of the highest priority class must first be allocated in the "steady region", then the cell loss probability Γ_1 is satisfied by changing only the buffer size, because the effect of changing A_1 is not evident in the "steady region". But, for the other two classes the cell loss probability can be satisfied by changing either of the buffer size and the average offered load.

For the average delay time, which varies evidently with the buffer size at the lowest priority class when the average offered load of this priority class is very large, while the variations of the average delay times for the different buffer sizes at higher priority classes(or at lowest priority class when A_{lowest} is very small) are not apparent These phenomena can be explained by the fact that the probability distribution of longer queue lengths is generally higher for the lowest priority class, especially under higher average offered loads For the highest priority classes the probability distribution of smaller queuing length is usually higher, particularly under lower average offered load

The above results are very useful to allocating bandwidth of each class effectively in ATM multiplexer with the priority class

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APPENDIX

APPENDIX A

Critical Values of the t Distribution

			α		
n	0 10	0 05	0 025	0 01	0 005
1	3 078	6 314	12 706	31 821	63 657
2	1 886	2 920	4 303	6 965	9 925
3	1 638	2 353	3 182	4 541	5 841
4	1 533	2 132	2 776	3 747	4 604
5	1 476	2 015	2 571	3 365	4 032
6	1 440	1 943	2 447	3 143	3 707
7	1 415	1 895	2 365	2 998	3 499
8	1 397	1 860	2 306	2 896	3 355
9	1 383	1 833	2 262	2 821	3 250
10	1 372	1 812	2 228	2 764	3 169
11	1 363	1 796	2 201	2 718	3 106
12	1 356	1 782	2 179	2 681	3 055
13	1 350	1 771	2 160	2 650	3 012
14	1 345	1 761	2 145	2 624	2 977
15	1 341	1 753	2 131	2 602	2 947
16	1 337	1 746	2 120	2 583	2 921
17	1 333	1 740	2 110	2 567	2 898
18	1 330	1 734	2 101	2 552	2 878
19	1 328	1 729	2 093	2 539	2 861
20	1 325	1 725	2 086	2 528	2 845
21	1 323	1 721	2 080	2 518	2 831
22	1 321	1 717	2 074	2 508	2 819
23	1 319	1 714	2 069	2 500	2 807
24	1 318	1 711	2 064	2 492	2 797
25	1 316	1 708	2 060	2 485	2 787
26	1 315	1 706	2 056	2 479	2 779
27	1 314	1 703	2 052	2 473	2 771
28	1 313	1 701	2 048	2 467	2 763
29	1 311	1 699	2 045	2 462	2 756
inf	1 282	1 645	1 960	2 326	2 576

Extract this table from [5].

APPENDIX B

Critical Values of the Chi-Square Distribution

α	0 05	0 025	0 01	0 005
x^2				
k				
1	3 8415	5 0239	6 6349	7 894
2	5 9915	7 3778	9 2103	10 597
3	5 9915	7 3778	11 345	12 838
4	9 4877	11 143	13 277	14 860
5	11 071	12 833	15 086	16 750
6	12 592	14 449	16 812	18 548
7	14 067	16 013	18 475	20 278
8	15 507	17 535	20 090	21 955
9	16 920	19 023	21 666	23 589
10	18 307	20 483	23 209	25 188

Extract this table from [5].