PERFORMANCE ANALYSIS OF A MULTISERVER DISCRETE-TIME QUEUEING SYSTEM WITH CORRELATED BATCH ARRIVALS FOR ATM BASED B-ISDN

by

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Abstract

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Arshad Hussain

Advisor: Professor Mohamed A. Ali

Most queueing analyses performed in literature are based on characterizations of queueing phenomena in continuous-time terms. Recently in telecommunication industries, Broadband Integrated Services Digital Network (B-ISDN) has received considerable attention for its capability of providing a common interface for future communication needs including video, data and voice. Since information in B-ISDN is transported by means of discrete units of 53-octets Asynchronous Transfer Mode (ATM) cells, interests in discrete-time systems have increased.

This research dissertation presents general discrete-time analytical models for the characterization of the queueing performance of an ATM network in terms of the overall performance characteristics such as buffer occupancy (in cells), jitter distribution and its impact from spatial correlation of the mono- media streams. Multistream arrivals within the same time slot are assumed to be generally correlated batch processes, but batches are independent from slot to slot. Traffic generated from mono-media sources in a multi-

media session, such as voice and video, are of a stream type, and tend to be correlated. Lack of independence among mono-media sources in a multi-media environment, and the related temporal characteristics embedded in the behavior of mono-media sources, introduce cross-correlation among various mono-media streams. These correlations if not properly taken into account, as this dissertation shows, can impact network performance, such as jitter and synchronization.

The main characteristics of the present treatment lies in introducing two critical new features crucial for the accurate assessment of the queueing performance of an ATM network with multi-media traffic handling capabilities. First, unlike most of the conventional analysis which assume a 1-D known arrival process, e.g., MMPP, BMAP, DBMAP, etc., this dissertation presents 2-D general independent[1] (GI) arrival process models. Second, the model assumes multiple correlated batches of information units , arriving in the same time slot, enter multiple queues with multiple servers. The media servers are of different types, such as audio server, and video server. Specifically, spatial correlations present in multi-media traffic are taken into account and numerical examples showing the overall performance of the proposed queueing model are presented. These results are then compared with that of the conventional single queue models, i.e., GI-D-c. This dissertation has established the fact that GI-D-c model and the like *under or overestimate* the average system contents depending on the degree of correlation among the mono-media streams and the server configurations.

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[1] The numbers of messages entering the system during the consecutive time-slots are assumed to be independent and identically distributed (i.i.d) non-negative discrete random variables with an arbitrary (general) probability distribution.

Several experimental measurements in multi-media videoteleconference sessions are performed to explore and characterize the spatial correlations among mono-media sources. All experiments were performed in a Local Area Network (LAN) employing CSMA/CD. In all the experiments described in this dissertation, such correlations are demonstrated. It is envisioned that such correlations would also exist in Wide Are Networks (WAN) such as B-ISDN/ATM. Conclusions are reached regarding the influence of spatial correlation on ATM network performance by employing the experimental data into the proposed two-queue analytical model.

Preface

Except where otherwise stated in the text, this dissertation is the result of my own work and is not the outcome of work done in collaboration.

This dissertation is not substantially the same as any I have submitted for a degree or diploma or any other qualification at any other university.

No part of this dissertation has already been, or is being currently submitted for any such degree, diploma or other qualification.

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Chapter 1

Introduction

1.1 Introduction

Asynchronous Transfer Mode (ATM) is emerging as the technology of choice for future Broadband Integrated Services Digital Network (B-ISDN). These B-ISDN networks are either synchronous transfer mode (STM) or asynchronous transfer mode (ATM), to handle both constant and variable bit-rate traffic applications. STM provides fixed bandwidth channels, and therefore is not flexible enough to handle the different types of traffic typical in multimedia applications. On the other hand, ATM is suitable for multimedia traffic, since it provides great flexibility in bandwidth allocation by assigning fixed length packets, called cells, to virtual connection. ATM can also increase the bandwidth efficiency by buffering and statistically multiplexing bursty traffic at the expense of cell delay and loss. The defining characteristics of this technology lies in its cell-based structure that allows ATM to carry voice, video and data in the same transport and switching equipment. An ATM network may be viewed as a collection of network nodes connected by a set of transmission links, whereby the Store-and- forward principle is used to transmit ATM cells from a source to a destination.

Multimedia systems combine a variety of mono-media sources, such as, data, voice, audio, graphics, animation, images, and full motion video, into a wide range of applications. Traffic generated from mono-media sources in a multimedia session, such as voice and video, are of a stream type, and tend to be correlated. Lack of independence among mono-media sources in a multimedia environment, and the related temporal characteristics embedded in the behavior of mono-media sources, introduce cross-correlation among various mono-media streams. These correlation if not properly taken into account, as this paper will show, can impact network performance, such as jitter and synchronization. For instance, packets (such as ATM cells) that are generated from voice and video in a teleconference environment are generally bursty in nature and cross-correlation among batches of voice and video, if not properly taken into account, may result in network's performance degradation.

Several attempts have been made to characterize the queueing performance of ATM networks transporting multimedia traffic [2], [44], [45], and [51]. The difficulty with analytical modeling of multimedia sources and their treatment over an ATM network is that the traffic behavior is not known as well as that of a mono-media source. Most of these analyses have assumed a known arrival process, e.g., semi- Markov arrival process and bursty model [43], and Point Process [44], to represent each of the many mono-media sources and the corresponding multi-media aggregate. The main problem associated with these approaches is that the assumed process cannot capture the source characteristics and therefore multimedia traffic is treated as independent for the most part. These correlations if not properly taken into account, as this dissertation shows, can impact/deteriorate the network performance. The penalty is in terms of jitter and loss of synchronization.

1.2 Thesis Objectives

This thesis seeks to develop novel multimedia traffic models for the characterization of the queueing performance of an ATM network in terms of the overall performance characteristics, such as, buffer occupancy (in cells), cell delays, and jitter behavior. The main characteristics of the present treatment lies in introducing two critical new features crucial for the accurate assessment of the queueing performance of an ATM network with multi-media traffic handling capabilities. First, unlike most of the conventional analysis which assume a 1-D known arrival process [43]-[45], this thesis presents 2-D general independent[1] (GI) arrival process models. Second, unlike most of the conventional analysis which have always assumed a single queue for all the incoming multi-media traffic, this thesis assumes a separate queue for each type of mono-media traffic comprising the multi-media traffic aggregate. Specifically, considering the problem in the context of discrete-time queueing models, and without lack of generality, we assume a multi-media session consisting of two mono-media correlated sources (voice and video), where each stream consists of independent batches. The media servers are of different types, such as audio server, and video server.

We consider two queueing approaches. In the first approach we assumes multiple queues with separate group of servers. In this model we assume that the two mono-media streams are transmitted in parallel and join two separate queues at the multiplexer or switching

[1] The numbers of messages entering the system during the consecutive time-slots are assumed to be independent and identically distributed (i.i.d) non-negative discrete random variables with an arbitrary (general) probability distribution.

nodes in the ATM network. C_i servers in parallel serve each queue, where "i" is the mono-media source index within the multi-media session. In the second approach we consider a single queue with a group of servers. Here it is assumed that the multi-media source traffic over an ATM network is transmitted in a single high speed channel and handled at the multiplexer / switch using a single queue with a group of (parallel) servers, e.g., GI-D-c [43]. Several numerical examples illustrating the main features and the overall performance analysis of our proposed model are presented. These results are then compared with that of the conventional analysis results.

These objectives are carried out in two phases. First, the intuition that mono-media traffic comprising a multi-media session are correlated, is experimentally substantiated under different temporal conditions for a given LAN architecture with video-teleconferencing sessions. Specifically, we explore and characterize the correlation among audio and video streams, within a given time slot, in a real-time video-conferencing session. Second, the measured data of the experimental phase are used to calculate the spatial correlations among mono-media streams and the joint probability density functions (pdf). It is important to emphasize that even though the measurements and computation of spatial correlation are carried out in an Ethernet LAN environment, it is envisioned that such correlations would also exist in Wide Area Networks (WAN) such as B-ISDN/ATM and multi-media wireless systems.

Chapter 2

ATM and the Future of Multi-media

Standardization work on the Broadband ISDN (B-ISDN) has been underway within the CCITT SG XVIII now ITU-T SG13), and the basic recommendations are now complete. This section will examine the technical developments behind the CCITT Recommendations, focusing on basic principles of ATM, B-ISDN services and the network as seen from the user's point of view.

2.1 What is ATM?

In ATM, all types of information are segmented and input to information fields of short blocks called cells as shown in Figures (2.1-2.2). These cells are transferred with a header containing the destination address and other information. ITU-T specifies a cell configuration with a 5-octet (1 octet = 8 bits) header length and a 48-octet information field length.

The types of information to be handled by the communications networks can be roughly classified into two categories. The first type is handled as constant generation of information such as telephone communications, while the second type is generated as burst (packet) information, such as PC communications. ATM can accommodate both constant and burst information by means of cell-based transfer techniques.

In the case of constant data, cells are assigned at certain cycles, while for burst data cells are assigned every time information is generated. One of the features of ATM is that it maintains both the real-time data transfer characteristics of circuit switching and the flexible speed assignment of packet switching. This is achieved by dividing all of the information into very small cell units and transferring and switching these units at super-high speed. Thus, ATM can be considered to be a method in which circuit switching and packet switching are integrated. With ATM, any type of information can be transferred and switched in real time or non-real time at either fixed or variable speed, according to the characteristics of the constant or burst data sources. Therefore, it can handle various information transfer patterns, and offers high flexibility with regard to changes that may take place in communications of the future. Information from several terminals can be multiplexed to a single ATM line, allowing an efficient network to be realized by utilizing the maximum transfer capacity of the line.

In ATM, cells are multiplexed on a single ATM line from several terminals, as shown in Figure (2.3). To identify the cells representing the same communications, Virtual Channel Identifiers (VCIs) contained in the headers are used. VCIs are defined on a link-by-link basis between terminal and switching system, and between switching systems. In addition to switching cells from an input link to an output link, an ATM switching system also changes the values of VCIs.

2.2 Background and Objectives of the B-ISDN

As opposed to the B-ISDN, the present ISDN networks are referred to as Narrowband ISDN (N-ISDN). In the N-ISDN, the rates of information transfer speeds are based on the basic rate of 64kbit/s, ranging up to 1536/1920kbit/s. In the B-ISDN, the maximum transfer rates will be two orders of magnitude higher, at 155.52Mbit/s or 622.08Mbit/s. The reason why these transfer rates are needed is that business communications and office automation through workstations and LANs is expected to make severe demands on the communications network. In particular, the B-ISDN with its ATM technology will be an efficient way to implement multimedia communications, with video information added to the conventional voice and data information. Also in the background is recent progress in optical technology.

It is now possible to construct high-quality, low-cost communication channels directly to the user's home or workplace, raising the possibility of B-ISDN services such as High Definition Television (HDTV) broadcasting offered at 155.52/622.08Mbit/s. These developments in very-high-speed switching are due to progress in optical-fiber, high-integration LSI and ATM switching technology. The B-ISDN will offer many benefits to both user and carrier. For the user, it will answer the need for faster and more varied kinds of communications services. For the carrier, it will be flexible enough to cope economically with unforeseen developments in future communications networks. .

2.3 Types of Communications Services

In addition to all of the services offered by conventional telephone and packet switched networks, the N-ISDN offers digital switching at speeds ranging from 64kbit/s to 1536/1920kbit/s. In the same way, the B-ISDN will offer all of the services provided by the N-ISDN, in addition to high-speed broadband services.

B-ISDN services will be available in two broad categories, interactive and distribution. The following offers some examples of services that could be provided under these categories.

Interactive Services: Interactive services feature bi-directional communications between terminals. Communications can be simultaneous, as in telephone conversations, monitoring by remote cameras or TV conferences, or non simultaneous, as in the case of store-and forward message transfer services or database retrieval services.

Distribution Services: Distribution services refer to television broadcasting and other kinds of services in which information is passed from a provider to an indeterminate number of recipients.

2.4 Characteristics of B-ISDN Services

As related above, the B-ISDN can offer a wider variety of services than conventional communications networks. But another of its outstanding characteristics is a new approach to transmission speeds, multi-media communications and quality control.

Transmission Speeds: Transmissions over the B-ISDN can be conducted using either a Constant Bit Rate (CBR) or a Variable Bit Rate (VBR). CBR services are provided at a fixed transmission speed. Within the B-ISDN, for example, the 64kbit/s to 1536/1920kbit/s services offered by the N-ISDN will be provided under this category. But thanks to ATM technology, the B-ISDN will also be able to offer Variable Bit Rate services for cases in which the amount of information to be transmitted is not constant. In video conference, for example, the amount of information varies depending on the amount of movement within the pictures.

2.5 Service Categories and Example B-ISDN Services

Multimedia Communications: The N-ISDN has provided a new framework for research into a variety of integrated, multimedia services. The B-ISDN will inherit the results of this research, and develop them further in the direction of distribution and variable-speed applications. As the cost of ISDN terminals falls, entirely new ways of using the network are expected to emerge. In order to support these new services, the B-ISDN will support functions such as the establishment of multiple media, and the addition or deletion of new media during a communications session.

Service Quality: The quality demand of communications varies according to the type of application. In a telephone conversation, for example, the transmission must be carried out in real time. In data communications, on the other hand, the paramount consideration is the elimination of transmission errors. In the past, telecommunications networks have been designed to assure one of these parameter sets. But the B-ISDN will be able to handle different parameter sets.

Using a single network based on the ATM technology, it will be possible to offer a variety of communication speeds and fixed or variable bit rates for a variety of media, with the right kind of communications quality for every application, and even different bit error requirements for low-speed and high-speed applications.

For this reason, the B-ISDN does not make fixed assumptions about service quality. Instead, it allows the terminal to request a certain kind of quality. For example, one method would be to combine a certain set of quality parameters and offer this set as a service class. During the call establishment phase, the terminal could examine a menu of available classes and choose the one most suited for the application. A good deal of work remains to be done on the formulation of service classes and quality negotiation procedures between the terminal and network, but this flexibility will make it easier to obtain the maximum benefits from B-ISDN communications.

2.6 Outline of the B-ISDN Protocols

Communications within the B-ISDN are regulated by protocols for transmission, reception and communications between terminals and the network. Figure (2.4) shows the B-ISDN protocols reference model for each layer of the B-ISDN.

Connections between terminals and switches, or between the different switches in the network, are achieved via the Physical Layer. Physical connections will be made over optical fibers, but during the introductory stage use will also be made of existing coaxial cables.

Above the Physical Layer is the ATM Layer, which executes the transmission and exchange of user information. In the ATM Layer, all information is converted into blocks of 53 octets, called cells. For continuously generated information, cells are allocated over a fixed period, while for information that comes in bursts the cells will be allocated whenever they are needed.

The ATM Layer carries out the processing and exchange of cell for all types of communications, but for some applications it will be necessary to use another layer, called the ATM Adaptation Layer (AAL). An AAL can be designed with functions to handle each kind of message or media supported by the network. In the case of voice communications, for example, the call originating side will detect talking periods and assemble cells, while the receiving side will convert the cells back into voice information.

An important characteristic of the B-ISDN is that *the network does not request retransmission of cells*. If an error is detected, the cell is simply discarded. This method makes it possible to achieve very high transmission rates with little degradation, thanks to the high quality of recent optical-fiber technology. On the other hand, however, the AAL must be able to compensate for not-received cells or erroneous cells. One very simple way to compensate would be to simply use previously received information. But as pointed out above, different applications require different types of communications quality. One aspect of the interface which should be understood is the relation to the Synchronous Optical Network (SONET) or Synchronous Digital Hierarchy (SDH). The SONET is a standard technology for multiplexing existing digital PCM channels, in order to serve the Network Node Interface (NNI). A SONET STS-1 frame is organized as a two-dimensional structure comprising of 9 (rows) by 90 (columns). The signal rate can be given as follows:

$$810 \ \frac{bytes}{frame} \times 8000 \frac{frames}{second} \times 8 \frac{bits}{byte} = 51.84 \ Mbps$$

The derivation given above can be applied to any signal, for example, an STS-12 signal, there are 1080 (12x90) columns. One can easily obtain the STS-12 signal rate from 12x90x0.576 Mbps, or 622.08 Mbps. In other words, SONET signals are modular.

2.7 SONET/SDH

SONET (Synchronous Optical Network) is an optical transmission interface originally proposed by Bellcore and standardized by ANSI. A compatible version, referred to as synchronous digital hierarchy (SDH), has been published by CCITT in Recommendations G.707, G.708, and G.709[2]. SONET is intended to provide a specification for taking advantage of the high-speed digital transmission capability of optical fiber.

The SONET standard addresses the following specific issues [76]:

[2] In what follows, we will use the term SONET to refer to both specifications. Where differences exist, these will be addressed.

1. Establishes a standard multiplexing format using any number of 51.84-Mbps signals as building blocks. Because each building block can carry a DS3 signal, a standard rate defined for any high-bandwidth transmission system that might be developed.

2. Establishes an optical signal standard for interconnecting equipment from different suppliers.

3. Establishes extensive operations, administration, and maintenance (OAM) capabilities as part of the standard.

4. Defines a synchronous multiplexing format for carrying lower-level digital signals (DS1, DS2, CCITT standards). The synchronous structure greatly simplifies the interface to digital switches, digital cross-connect switches, and add-drop mulrtiplexers.

5. Establishes a flexible archtiecture capable of accomodating future applications such as broad ISDN with a variety of transmission rates.

Three key requirements have driven the development of SONET. First was the need to push multiplexing standards beyond the existing DS-3 (44.736-Mbps) level. With the increasing use of optical transmission systems, a number of vendors have introduced their own proprietary schemes of combining DS-3s into an optical signal, for details please refer to Tables (2.1) & (2.2). In addition, the European schemes, based on the CCITT hierarchy, are incompatible with North American schemes. SONET provides a standardized hierarchy of multiplexed digital transmission rates that accommodates existing North American and CCITT rates.

A second requirement was to provide economic access to small amounts of traffic within the bulk payload of an optical signal. For this purpose, SONET introduces a new approach to time-division multiplexing.

A third requirement is to prepare for future sophisticated service offerings, such as virtual private networking, time-of-day bandwidth allocation, and support of the broadband ISDN ATM transmission technique. To meet this requirement, a major increase in network management capabilities within the synchronous time-division signal was needed.

2.7.1 Signal Hierarchy

The SONET specification defines a hierarchy of standardized digital data rates (Table 2.1). The lowest level, referred to as STS-1 (synchronous transport signal level 1), is 51.84 Mbps. This rate can can be used to carry a single DS-3 signal or a group of lower rate signals, such as DS1, DS1c, DS2, plus CCITT rates (e.g., 2.048 Mbps).

Multiple STS-1 signals can be combined to form an STS-N signal. The signal is created by interleaving bytes from N STS-1 signals that are mutually synchronized.

For the CCITT synchronous digital hierarchy, the lowest rate is 155.52 Mbps, which is designated STM-1. This corresponds to SONET STS-3. The reason for the discrepancy is that STM-1 is the lowest signal that can accommodate a CCITT level 4 signal (139.264 Mbps).

2.7.2 System Hierarchy

SONET capabilities have been mapped into a four-layer hierarchy Figure (2.5):

- Photonic: This is the physical layer. It includes a specification of the type of optical fiber that may be used and details such as the required minimum poers and dispersion characteristics of the transmitting lasers and the required sensitivity of the receivers.
- Section: This layer creates the basic SONET frames, converts electronic signals to photonic ones, and has some monitoring capabilities.
- Line: This layer is responsible for synchronization, multiplexing of data onto SONET frames, protection and maintenance functions, and switching.
- Path: This layer is responsible for end-to-end transport of data at the appropriate signaling speed.

Figure (2.6) shows the physical realization of the logical layers. A section is the basic physical building block and represents a single run of optical cable between two optical fiber transmitter/receivers. For shorter runs, the cable may run directly between two end units. For longer distances, regenerating repeaters are needed. The repeater is a simple device that accepts a digital stream of data on one side and regenerates and repeats each bit out of the other side. Issues of synchronization and timing need to be addressed. A line is a sequence of one or more sections such that the internal signal or channel structure of the signal remains constant. Endpoints and intermediate switches/multiplexers that may add or drop channels terminate a line. Finally, a path connects to end terminals; it corresponds to an end-to-end circuit. Data are assembled at the beginning of a path and are not accessed or modified until they are disassembled at the other end of the path.

2.7.3 Pointer Adjustment

In conventional circuit-switched networks, most multiplexers and telephone company channel banks require the demultiplexing and remultiplexing of the entire signal just to access a piece of information that is addressed to a node. For example, consider that T-1 multiplexer B receives data on a single T-1 circuit from T-1 multiplexer A and passes the data on to multplexer C. In the signal receive, a single DS0 channel (64 kbps) is addressed to node B. The rest will pass on to node C and further on into the network. To remove that single DS0 channel, B must demultiplex every bit of the 1.544 Mbps signal, remove the data, and remultiplex every bit. A few proprietary T-1 multiplexers allow for add-drop capability, meaning that only part of the signal has to be demultiplexed and remultiplexed, but this equipment will not communicate with that of the other vendors.

SONET offers a standard add-drop capability, and it applies not just to 64-kbps channels but to higher data rates as well. SONET makes use of a set of pointers that locate channels within a payload and the entire payload within a frame, so tha tinformation can be accessed, inserted, and removed with a simple adjustment of pointers. Pointer information is contained in the path overhead that refers to the multiplex structure of the channels contained within the payload. A pointer in the line overhead serves a similar function for the entire payload.

2.8 Transmission of ATM Cells

The B-ISDN specifies that ATM cells are to be transmitted at a rate of 155.52-Mbps or 622.08-Mbps. As with ISDN, we need to specify the transmission structure that will be used to carry this payload. For 622.08-Mbps, the matter has been left for further study. For the 155.52-Mbps interface, two approaches are defined in I.413: a cell-based physical layer and an SDH-based physical layer.

2.8.1 Cell-based Physical Layer

For the cell-based physical layer, no framing is imposed. The interface structure structure consists of a continuous stream of 53-octect cells (Figure 2.1). Since there is no external frame imposed in the cell-based approach, some form of synchronization is needed. Synchronization is achieved on the basis of the header error check (HEC) field in the cell header. The procedure is as follows Figure 2.6:

1. In the HUNT state, a cell-delineation algorithm is performed bit by bit to determine if the HEC coding law is observed (i.e., match between received HEC

and calculated HEC). Once a match is achieved, it is assumed that one header has been found, and the method enters the PRESYNCH state.

2. In the PRESYNCH state, a cell structure is now assumed. The cell delineation algorithm is performed cell by cell until the encoding law has been confirmed *delta* times consecutively.

3. In the SYNCH state, the HEC is used for error detection and correction. Cell delineation is assumed to be lost if the HEC coding law is recognized as incorrect *alpha* times consecutively.

Finally, ATM cells are used to convey operations, administration, and maintenance (OAM) information. A virtual-path identifier of 0 and a virtual-channel identifier of 9 identifies OAM cells.

The advantage of cell-based transmission scheme is the simplified interface that results when both transmission and transfer mode functions are based on a common structure.

2.8.2 SDH-based Physical Layer

For the SDH-based physical layer, framing is imposed using the STM-1 (STS-3) frame. Figure 2.7 shows the payload portion of an STM-1 frame. This payload may be offset from the beginning of the frame, as indicated by the pointer in the section overhead of the frame. As can be seen, the payload consists of a 9-octet path overhead portion and the remainder, which contains ATM cells. Since the payload capacity (2,340 octets) is not an integer multiple of the cell length (53 octets), a cell may cross a payload boundary.

The H4 octet in the path overhead is set at the sending side to indicate the next occurrence of a cell boundary. That is, the value in the H4 field indicates the number of octets to the first cell boundary following the H4 octet. The permissible range of values is 0 to 52.

The advantages of SDH-based approach include [77]-[78]:

- it can be used to carry either ATM-based or STM-based (synchronous transfer mode) payloads, making it possible to initially deploy a high-capacity fiber-based transmission infrastructure for a variety of circuit-switched and dedicated applications and then to readily migrate to the support of B-ISDN.

- Some specific connection can be circuit-switched using an SDH channel. For example, a connection carrying constant-bit-rate video traffic can be mapped into its own exclusive payload envelope of the STM-1 signal, which can be circuit switched. This may be more efficient than ATM switching.

- Using SDH synchronous multiplexing techniques, several ATM streams can be combined to build interfaces with higher bit rates than those supported by the ATM layer at a particular site. For example, four separate ATM streams, each with a bit rate of 155-Mbps (STM-1), can be combined to build a 622-Mbps

(STM-4) interface. This arrangement may be more cost-effective than one using a single 622-Mbps ATM streams.

2.9 Multi-media systems

Several definitions for the terms multi-media application and multi-media system are described in the literature. Three criteria for the classification of a system as a multi-media system can be distinguished: the number of media, the types of supported media, and the degree of media integration [1].

The most simple criterion is the number of media used in an application. Using only this criterion, even a document processing application that supports text and graphics can be regarded as a multi-media system [33].

The types of supported media are an additional criterion [1]. In this case, we distinguish between time-dependent and time-independent media. A time-independent media object is usually presented using one presentation unit. An example is a bitmap graphic. Time-dependent objects are presented by a sequence of presentation units. An example is a motion picture sequence without audio (i.e., a video sequence) presented frame after frame. Because the integration of time-dependent media objects is a new and essential aspect in information processing, some authors define a multi-media system as a system that supports the processing of more than one medium with at least one time-dependent medium [1].

The degree of media integration is the third criterion [1]. In this case, integration means that the different types of media remain independent but can be processed and presented together.

Combining all three criteria, we propose the following definition of a multi-media system: a system or application that supports the integrated processing of several media types with at least one time-dependent medium [1].

2.10 ATM and the Future of Digital Video

One clear advantage of ATM is its ability to support video, voice, and data simultaneously[3]. As ATM deployment begins, however, ATM and existing networks, i.e., Ethernet, Token Ring, and circuit-switched WANs such as T1 and Integrated Services Digital Network (ISDN), will coexist for many years. To fully take advantage of ATM and existing networks, video applications must therefore interoperate over these network types, and these networks must interoperate with each other.

There are numerous considerations and tradeoffs to consider when developing an infrastructure for video applications, and no two networks will be identical. However, in the emerging milieu of hybrid shared LAN, switched LAN, and WAN networks, many of which will include ATM and other technologies, there are guidelines to follow in the creation and deployment of these networks. Moreover, the addition of ATM as a common

[3] ATM has the capability to support any traffic type that has been adapted to it with the appropriate ATM Adaptation Layer (AAL). All of these cells of various traffic types are switched along virtual connections through an ATM network.

video, voice, and data transport mechanism that transparently spans the LAN and the WAN creates additional options.

From the network perspective, video applications fall into one of the three areas:

- *Packetized video*, which runs over traditional LANs at the MAC (Medium Access Control; e.g., Ethernet) or network (e.g., IP) layers, and that will therefore be transported into ATM networks through LAN emulation, layer 3encapsulation, or multiprotocol over ATM (MPOA). These applications will use LAN switches, routers, and ATM switches, such as LightStream® ATM switches, Cisco routers, and Catalyst(TM) and EtherSwitch® LAN switches.

- *Constant-bit-rate video*, which runs over traditional 64-kbps or multiple 64-kbps (or ISDN) lines, and will be transported over ATM using circuit emulation. These applications will use ATM service multiplexers, ATM switches, or existing leased-line facilities.

- *Packetized video*, which will run natively over ATM using an MPEG2-to-ATM convergence layer. Standards for doing such video transmission are still in the nascent stages of development and probably will not be widespread for several years. These applications will make use of specialized MPEG2 video to ATM codecs[4], still in early stages of development.

[4] A codec is a device that provides compression and decompression of a digital video signal, although it is commonly used to refer to a product that also digitizes an analog video signal, such as NTSC, PAL, or SECAM signal from a television or VCR, and then applies compression.

2.11 Packetized Video vs. Constant Bit Rate (CBR) Video

Video applications are either packetized, which means that they are *bursty or variable bit rate*, or they are *constant bit rate*. The packetized video types were designed to run over traditional LANs, and make use of some type of compression algorithm whose output is dropped into traditional packets, whether they are IP packets (or the equivalent) or Ethernet frames (or the equivalent). Because packetized video runs over LAN infrastructures, it can perform even better when the LAN infrastructure has protocol support and enhanced features to guarantee quality of service. In addition, when LAN-based video will run over ATM, there must be some translation from ATM to the legacy LAN, whether by LAN Emulation or by Multiprotocol over ATM.

When packetized video runs natively over ATM, there must be a convergence layer between the video stream and the ATM Adaptation Layer (AAL), most likely ALL5, according to the most recent work of the ATM Forum. Such work is in the early stages and the development of standards-based video compression algorithms for two-way, low-bandwidth, high-quality video in the form of MPEG4 is also still underway. The ATM Forum is tackling MPEG2 over ATM, which is well suited for applications such as video broadcasting in LANs or by cable TV.

In the case where video must run over px64-kbps lines for wide-area transmission, the compressed bursty video bitstream must be transformed into a CBR bitstream to meet the requirements of transmission over these CBR digital lines. Traditional videoconferencing equipment, such as made by Intel, PictureTel, VTel, and Compression Labs, uses some variant of the H.320/H.261 protocol suite. This protocol suite uses buffers to ensure that bits are always sent at every "clock-tick", regardless of the original structure or traffic shape of the bursty video stream. Such CBR video, when transmitted over an ATM network requires *circuit emulation*, where the ATM transport is emulating the traditional px64-kbps or ISDN circuits that the H.320 equipment is expecting as its underlying network.

AAL5 will be used for real-time variable bit rate (VBR) and available bit rate (ABR) traffic to deliver bursty data as LANs today. AAL1 will be used for constant bit rate (CBR) traffic (H.320-compliant videoconferencing equipment) where a time-stamp is needed for precise transmission epochs.

2.12 Multicasting and Service Multiplexers in ATM

ATM networks inherently support multicasting, where a single source can set up connections to multiple users. This multicasting is designed to be able to copy cells down the multicast tree only at the last possible branch point, so that multiple individual connections are not set up through the network to each user. This conserves network bandwidth.

Because video streams can be compressed, they do not require the full dedicated bandwidth given to an ATM 155-Mbps (OC-3) port, even though MPEG or MPEG2 streams can require at least 1.5-Mbps each. To maximize the investment payoff in ATM switches, multiple input and output feeds can be multiplexed up to full ATM DS3/E3 or even OC-3 speed, to be fed into the ATM switch. Such an approach allows fanout of the video ports, giving high port density per ATM connection without tying up switch ports for individual video connections. The benefits of ATM, however, are presented through the service multiplexer.

2.13 Usage Parameter Control/Traffic policing

Given that there will likely be numerous services deployed by an ATM network, including data, voice, and video, to ensure performance guarantees for different services, call admission control will be necessary in ATM networks. In particular, it is not possible to speak of large-scale ATM networks deployed for high-end video and ignore the requirements for usage parameter controls. The usage parameter control (UPC), or traffic policing function, ensures that incoming traffic does not exceed the contract negotiated between the user and the network. Without such a function, already-established connections may incur performance degradation from users who violate their contract. The use of UPCs, therefore, is required for ATM networks that support video traffic. The UPCs ensures that all traffic types conform to their contract and receive their requested level of performance from the network.

Chapter 3

Spatial correlation in multi-media traffic

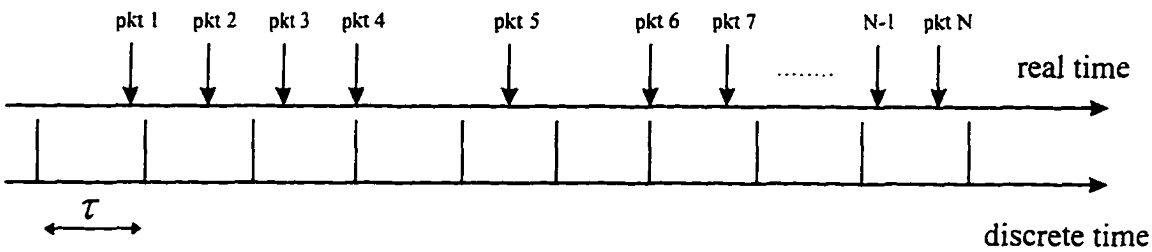
3.1 Introduction

The intuition that mono-media traffic comprising a multi-media session are correlated, is experimentally substantiated under different temporal conditions for a given Local Area Network (LAN) architecture with videoteleconferencing sessions. Specifically, we explore and characterize the correlation among audio and video streams, within a given time-slot, in a real-time videoteleconferencing session. Experimental measurements focused on multi-media traffic streams produced by individual workstations engaged in a videoteleconferencing session. The objective is to characterize the degree of spatial correlation between audio and video streams. For a pair of audio and video batch (α_i, v_i) bytes, respectively, in the (i^{th}) time slot, $i = 1,2,\ldots N$, the correlation coefficient ρ, [34], can be computed if the joint probability density function (pdf) of the arrival process is known. A detailed example is solved in Appendix-I.

It is important to be able to develop a method of measuring traffic data for a videoteleconference session as well as to be able to produce statistical estimates. Furthermore, in order to build a mathematical model for a real mono-media packet (or cell in ATM) arrival streams, we need some statistics such as the average, the variance of the number of arriving packets, and the corresponding cross-correlation function. This is why we propose here how to collect and process data in order to produce these statistics.

We are interested in a set of parameters which can be obtained from real data with high accuracy. These parameters are interarrival time, type of information (audio or video) transferred between the two participants of a conferencing session, and the number of bytes (size) of information carried by the packet. In section 3.3 we have given a detailed account of the information that can be obtained through *protocol analysis* for a conference session.

A conventional method of discretizing the real time line into fixed length slots is given below, where "*pkt 1*" through "*pkt N*" represent the total number of packets generated during the videoconferencing session.



During the analysis, we chose fixed length time intervals, henceforth referred to as time slots, containing batches of audio and video packets. In particular, a batch of audio or video represents the number of bytes in a given time slot. If a packet arrival time falls on the boundary of the time slot then it is put in the next time slot. The pdf of the conference data for a time slot of length τ and for batch sizes "*i*" and "*j*", is the frequency distribution of the joint occurrence of the form $\Pr[\alpha = i, v = j]$, where α represents audio and v represents video respectively. A typical pdf is shown in figure (3.3).

The pdf curves are sensitive to time slot length. This is because we used Ethernet (CSMA/CD) as the media access mechanism. As is well known, that a non-deterministic media access mechanism, like that of Ethernet, backs off exponentially in the face of contention, and hence causes gaps in the interarrival time of the packets. It is obvious that if we choose a smaller time slot length then the arrival epoch of the packet might not fall in this time slot and hence most of the small size time slots remain either empty or has smaller information carrying bytes. But as the time slot length is increased in steps then the arrival epochs of the packets will fall in the time slots which were missed previously. This leads to variations in the batch sizes of audio and video inside the time slots and hence the frequency distributions of the joint occurrence results in pdf's with longer tails. Furthermore, statistics such as the average, the variance of the number of arriving packets, and the corresponding cross-correlation between the mono-media packet arrival streams are affected by the time slot variations.

The cross-correlation between audio and video batches can be computed by using:

$$\rho_{a,v} = \frac{Cov(\alpha, v)}{\sigma_a \sigma_v}$$

Where α represents audio and v represents video respectively. The cross-correlation thus computed oscillates between positive and negative values. A *negative correlation* means that, within a short time slot (of the order of 10 msec - 80 msec), the probability of having video bytes is much higher than that of having audio bytes. In fact, within a short time slot one might get only video bytes and no audio at all. However, as the time slot length is increased, the spatial correlation takes on positive values and increases with time. Keep in mind that even though we are varying time slot lengths, it does not mean that the temporal characteristics of the arriving packets are changed. Also remember that we are computing cross-correlation between the batches of audio and video information bytes which is not temporal correlation.

Teleconferencing sessions were established by using a commercially available videoconferencing product, ProShare Personal Conferencing Video System 150® [6], designed by Intel Corporation. Several experiments were carried out to explore and characterize the spatial correlations among mono-media streams. The experiments have confirmed the feasibility of the approach and illustrate the presence of multi-media traffic correlations. Prior to describing the multi-media traffic analysis in Section 3.3, an outline of the experimental environment is given in Section 3.2. Section 3.4 details the Experimental measurements for different temporal conditions.

3.2 Experimental Environment

This section presents relevant background information about the environment used for development and execution of the experiments. A summary of the elements and organization of the system is first given, followed by a technical abstract in Section 3.2.2. describing the ProShare system 150® used for videoconferencing. Section 3.3, details the experimental results and analysis.

3.2.1 Synopsis

The environment used for development and experimentation consists of a collection of four dedicated workstations, which are physically connected to a Local Area Network as shown in Figure (3.1). Workstations running Windows for Workgroups operating system have Ethernet access and provide support for development, as well as a platform for executing applications. Two workstations are used for ProShare Personal Conferencing Video System 150® [6]. This allows audio and video data to be transferred over the LAN. The third workstation is used as a conferencing manager running LANDesk Personal Conferencing Manager® [35]. This workstation monitors the conferencing activities, i.e., displays details about each active call, amount of bandwidth allocated and in use, number of active calls, and service level for videoconferencing. The fourth workstation is used as a protocol analyzer running LANDesk Traffic Analyst® [36], which is a full featured packet monitoring product. It supports packet capture and decode from both the local and the remote segments. In addition, it displays real-time statistics such as packet rate, and error rate for all or part of the network traffic. We used it to develop our own protocol decoder and packet capture filters for PSVideo by using the compiler that comes with it [37].

3.2.2 ProShare System 150®

PSVideo system is equipped with a video and audio capture board , camera, headset, external speakers and software. The connection established by one PSVideo system to another operates on top of datagrams. With datagram protocol, packets do not have to be acknowledged by the receiver and therefore are not guaranteed to be delivered in the same order that they are sent. The PSVideo software time stamps successive audio and video packets. After reception, packets are resequenced before playback. Audio packets are given priority while video packets are either delayed or dropped in order to keep audio delivery in synch. The received packets are buffered by the software to reduce jitter and perform resequencing. Dropped packets are not retransmitted, because packets would arrive late and not be of any use [38].

Multi-media applications depend on *Codecs* (compressor/decompressors), which may be voice, video, or data. Codecs are algorithms amenable to software-only or hardware-assisted playback and provides bit rate reduction by removing superfluous information. Video coding schemes are categorized into *source* and *entropy coding*. Source coding is lossy and produces low quality video pictures. Source coding can be further divided into *intraframe* and *interframe coding*. Interframe coding removes both spatial and temporal redundancies from sequences of similar pictures, including those containing moving objects, whereas intraframe removes only the spatial redundancy within a picture [39], [40]. Entropy coding achieves bit-rate reduction by using the statistical properties of the signals and, in theory, is lossless. ProShare Video System uses a software assisted capture board with embedded i750 chip, as video Codec, called *Indeo Video* [40]-[41], which employs both interframe and intraframe compression by dividing the video stream, respectively, into *delta* and *key frames* [42]. A key frame is not subjected to interframe coding. They are used only to provide a reference for subsequent delta frames and a point of access to the video. Delta frames are smaller than key frames and removes superfluous information from previous frames, creating the interframe compression. To maintain its data rate during high motion sequences, Indeo Video Codec starts *subsampling* using interframe compression, producing delta frames with less information. Subsampling reduces the resolution of the image and the Codec starts to work with bigger pixel blocks, hence reducing the number of pixels in an image. We don't know the exact subsampling rate of Indeo Video Codec but a comparison of Indeo Video and other products is presented in [41], which clearly states that Indeo Video utilizes subsampling to reduce the number of pixels in an image by less than four times. On the other hand if every frame is a key frame then the Codec drops frames in order to maintain its data rate.

Audio compression in PSVideo is handled by *GSM Algorithm* which is similar to the one employed by the European cellular phone system known as Global System for Mobile Communications (GSM) [38].

3.3 Multi-media Traffic Analysis

Protocol analysis involves recording network activity and capturing, filtering, and viewing packets. In a real-time videoteleconferencing session, audio and video packets are interleaved before transmission. In order to capture and discriminate the contents of various packets exchanged during a videoteleconferencing session, the author developed protocol decoder and packet capture filters. The main communication protocol is IPX over NetBEUI® carrying Ethernet_II frames. Multi-media traffic measurements were carried out in two different phases. In the first phase, traffic measurements were carried out in a videoteleconference session to analyze and isolate the contents of the captured conference packets. After careful and detailed analysis of the videoconference packets, Figure (3.2), we observed that:

- Each audio and video packet has three ASCII characters, *Media Dependent Module* (*MDM*), as signature. This signature is three bytes long and is present at offset 44 in every videoconference packet.
- A single byte follows at offset 47, which is a *function code*, where the hexadecimal value represents the function
    - 00 = audio/video/data,
    - 01 = request connection,

- 02 = accept connection,
- 03 = reject/close connection, and
- 04 = clear to send user data.

- The following ten bytes at offset 48 through 57 are *reserved* for protocol information.

- The next byte at offset 58 is very important as far as *packet contents* are concerned. The hexadecimal value identifies the receive channel contents as:

    - 03 = audio (caller),
    - 04 = video (caller),
    - 06 = audio (callee), and
    - 07 = video (callee).

- We have labeled the next two fields as *Reserved Control*, in our protocol decoder since their function is not known.

- Each captured packet, contains video or voice data (but not both) starting at offset 64 till the end of packet. Regular audio packets are 234 bytes long, while muted audio packets are 54 bytes in length. Video packets vary in size (between 62 and 576 bytes).

In addition to video and voice, streams of 3 to 4 control packets are occasionally exchanged between the conferencing stations. All control packets were filtered out and only audio and video information is processed in our analysis. Packets are captured to a RAM buffer, before they can be permanently stored to a hard disk. To capture data for a videoconference session a large amount of RAM is needed. Typically to store 30 minutes of *single line decoded* data, for a single conference, one needs 32 Mbytes of RAM. The output generated by our protocol decoder is an ASCII file, typically 35 Mbytes in size which contains the captured packets information. In the second phase, statistical analysis were carried out in order to obtain the correlation coefficient from the joint probability distribution for the multi-media arrival stream.

3.4 Experimental Results

In this section we present experimental results and describe the details of the experiments.

3.4.1 Experiment 1

Extensive audio exchanged between two “talking head” participants

This experiment simulates a scenario of extensive audio exchange between the two "talking head" participants. The phrase "talking head" is used when the video camera is focused on the participant's faces. The video-conference session in this experiment was 30 minutes long. The participants were continuously speaking thus generating a lot of audio packets. A significant number of control packets are exchanged between the two workstations during the course of a conference. These control packets were removed from the captured file during the data analysis. Table (3.1) lists the total number of captured packets, the total number of control packets for both Caller and Callee. For a given time slot length, the number of bytes contained within batches of video/audio were measured. Typical pdf curves are shown for varying time slot lengths in Figures (3.3)-(3.10). The cross-correlation between audio and video is shown in Figures (3.11)-(3.12).

3.4.2 Experiment 2

Extensive audio and video exchanged between two participants

This experiment simulate a scenario of extensive audio and video exchanged between the two participants. It contains excessive high motion sequences and video transitions. Traffic measurements and analysis carried out in Experiment #1 were repeated again in this experiment and the results for short time slots are also included in Figures (3.11)-(3.12). The total number of captured packets and control packets for both Caller and Callee are also included in Table (3.1). Note that, although this experiment contains excessive high motion sequences and video transitions, the total number of captured packets in this experiment is less than that of Experiment #1. This is because the codec is no longer able to keep up with the high motion sequences, and consequently, starts to drop video frames in order to maintain its data rate at the expense of picture quality. As expected, and shown in Figure (3.11), the results and data analysis of this experiment are generally in good agreement with experiment #1. This indicates that the correlations among mono-media traffic streams in a multi-media session seem to always exist regardless of the temporal behavior.

3.4.3 Experiment 3

Regular videoconference

This experiment simulate a scenario of a regular conference session. The two participants talk and move as they would in a normal conversation. Traffic measurements and data analysis carried out in the previous experiments were repeated and the results are included in Figures (3.11). The total number of captured packets and control packets for Caller and Callee are included in Table (3.1). The results and data analyses of this experiment are generally in agreement with the previous results.

Chapter 4

Discrete-Time Models for Multi-media Traffic

4.1 Introduction

In the literature, authors have considered time-correlated (mono-media traffic) arrival processes. Multi-media traffic behavior is not known as adeptly as mono-media traffic, which makes it difficult to model it analytically and handle over ATM networks. Most analyses have assumed a known arrival process, e.g., semi- Markov arrival process and bursty model [43], Point Process [44], or among the most popular Markov-modulated arrival processes, Markov-Modulated Poisson Process (MMPP) [17], to represent each of the many mono-media sources and the corresponding multi-media aggregate. In [18], Lucantoni introduced the Batch Markovian Arrival Process (BMAP), in order to improve the accuracy of the approximations. The BMAP has the same characteristics as the MMPP, except that cells arrive in batches. Discrete-time versions of MMPP and BMAP were developed by Blondia and were referred to as Discrete-time Markovian Arrival Process (DMAP) [14] and Discrete-time BMAP (DBMAP) [15], respectively. Aside from the possible difficulties in modeling mono-media sources, the mélange of such sources producing multi-media traffic on a network introduces additional problems. A multi-media source, comprised of mono-media sources, cannot be treated independent for the most part. For example, during a multicast/broadcast conference, usually the active participants information (combined voice, video, data, etc.) are broadcasted. Lack of independence among various mono-media streams and their temporal characteristics introduces cross-correlation's between them. In [45], La Corte presents an ON/OFF correlated traffic model for a multi-media arrival process. A Markov process model is used to represent each of the mono-media sources and the corresponding multi-media aggregate. The difficulty with that approach is the limitation of a Markovian process in representing multi-media traffic sources, which puts further constraints on the queueing model representing the underlying communication network. However, the source model captures the cross- and auto-correlation of the mono-media sources in a multi-media environment. The problem of synchronization control is also not addressed in that reference. In [44], Blondia suggests a versatile point process for the analysis of a class of discrete-time batch Markovian arrival process (DBMAP).

4.2 Multiple Queues with separate Groups of Servers

In this model we assume that the two mono-media streams are transmitted in parallel and join two separate queues at the multiplexer or switching nodes in the ATM network as shown in Figure (4.1) . Each queue is served by C_i servers in parallel, where "i" is the mono-media source index within the multi-media session. For example, in an ATM network, this may correspond to separate virtual channels each assigned to a different mono-media source. In the wireless case, this model may be interpreted as having (multiple) frequency channels assigned to several monomedia sources all transmitting in the same time slot (e.g., TDMA).

4.3 The Two-queue Model

In a correlated batch arrival model to be considered in the following section, we assume that the arrival process consists of two cross-correlated mono-media batch traffic streams, representing a multi-media session. We consider the case of two separate queues for each type/class of mono-media arrival streams.

4.3.1 General Assumptions and Terminology

A brief summary of the assumptions for the multiple queues with separate servers queueing model is given below.

- The buffers have unlimited storage capacity.
- The buffers are emptied through the transmission of the cells they contain. The number of output links via which the cells are removed (i.e., the number of servers in the multiple queueing model) are equal to $C_1 > 0$ and $C_2 > 0$.
- Time is divided into fixed-length intervals, referred to as *slots*, such that one slot suffices for the transmission of one cell via each of the output channels of the buffers (*i.e.*, the service times of cells are deterministic). The transmission of a cell via an

output buffer starts at the beginning of a slot and ends at the end of this slot. Cells cannot leave the buffer at the end of the slot during which they arrived in the buffer.

- New streams of cells from a multi-media session consisting of two cross-correlated mono-media sources, where each stream of cells consists of independent batches, enters the buffers according to a joint general independent arrival process. Batches of cells arriving in each stream are assumed to be independent and identically distributed (*i.i.d.*) non-negative discrete random variables (*r.v.'s*).

- The exact locations of the batch arrival instants within the slot length are not specified here. It is even irrelevant as long as the system is observed at slot boundaries only. If results are desired at other time instants (such as arrival instants or random time points), then a more detailed description of the arrival process is necessary.

- It is assumed that the service and the arrival processes are mutually independent.

Let

$$P(a_1, a_2) = \Pr[a_1 = i, a_2 = j \quad messages\ arrive\ during\ one\ time - slot], \quad (i, j) \geq 0$$

represent the common joint probability mass function of the batches of size a_1 and a_2 arriving in the same time slot. The corresponding probability generating function (pgf), [67]-[70], of the joint arrival process from a multi-media session consisting of two cross-correlated mono-media sources, is given by

Eq 4.3.1.1 $$A(Z_1,Z_2)=E\left[Z_1^{A_1}Z_2^{A_2}\right]=\sum_{i=0}^{\infty}\sum_{j=0}^{\infty}P\left[a_1=i,a_2=j\right]Z_1^iZ_2^j$$

The marginal probability generating function for mono-media stream of type 1, represented by $A_1(Z_1)$, is given by:

Eq 4.3.1.2 $$A_1(Z_1)=A(Z_1,1)=E\left[Z_1^{A_1}\right]=\sum_{i=0}^{\infty}\sum_{j=0}^{\infty}P\left[a_1=i,a_2=j\right]Z_1^i$$

and the marginal probability generating function for mono-media stream of type 2, represented by $A_2(Z_2)$, is given by:

Eq 4.3.1.3 $$A_2(Z_2)=A(1,Z_2)=E\left[Z_2^{A_2}\right]=\sum_{i=0}^{\infty}\sum_{j=0}^{\infty}P\left[a_1=i,a_2=j\right]Z_2^j$$

In the case of independent arrivals where, $P\left[a_1=i,a_2=j\right]=P\left[a_1=i\right]P\left[a_2=j\right]$, the expression for the probability generating function of the joint arrival process is:

Eq 4.3.1.4 $$\begin{aligned}A(Z_1,Z_2)&=E\left[Z_1^{A_1}Z_2^{A_2}\right]=\sum_{i=0}^{\infty}\sum_{j=0}^{\infty}P\left[a_1=i\right]P\left[a_2=j\right]Z_1^iZ_2^j\\&=E\left[Z_1^{A_1}\right]E\left[Z_2^{A_2}\right]=\left[\sum_{i=0}^{\infty}P\left[a_1=i\right]Z_1^i\right]\left[\sum_{j=0}^{\infty}P\left[a_2=j\right]Z_2^j\right]\\&=A_1(Z_1)A_2(Z_2)\end{aligned}$$

4.3.2 System Contents

Let us denote, respectively, by S_1^k and S_2^k the system contents at the beginning of time slot k of queue 1 and queue 2. $S_1^k(Z_1)$ and $S_2^k(Z_2)$ denotes, respectively, the pgf of S_1^k and S_2^k. The system contents S_1^{k+1} and S_2^{k+1} of both queues just before the k^{th} service instant evolves according to the following equations; where C_i, i = 1,2; represents the number of servers in queue 1 and queue 2, respectively.

Eq 4.3.2.1
$$S_1^{k+1} = \begin{cases} S_1^k - C_1 + A_1^k & if \;\; S_1^k \geq C_1 \\ A_1^k & if \;\; S_1^k < C_1 \end{cases}$$

Eq 4.3.2.2
$$S_2^{k+1} = \begin{cases} S_2^k - C_2 + A_2^k & if \;\; S_2^k \geq C_2 \\ A_2^k & if \;\; S_2^k < C_2 \end{cases}$$

Where A_1^k and A_2^k represents the total number of cell arrivals in mono-media streams of type 1 and type 2, in the interval between the k^{th} and $(k+1)^{th}$ service instants, respectively.

We can write equations (4.3.2.1) and (4.3.2.2) as

Eq 4.3.2.3
$$S_1^{k+1} = \left(S_1^k - C_1\right) \cdot U\left(S_1^k - C_1\right) + A_1^k$$

Eq 4.3.2.4
$$S_2^{k+1} = \left(S_2^k - C_2\right) \cdot U\left(S_2^k - C_2\right) + A_2^k$$

where $U(x) = \begin{cases} 0 & if \quad x < 0 \\ 1 & if \quad x \geq 0 \end{cases}$.

The joint probability generating function (pgf) of the system contents at steady state is defined as

Eq 4.3.2.5 $$S(Z_1, Z_2) = \underset{k \to \infty}{Lim} E\left[Z_1^{S_1} Z_2^{S_2}\right] = \sum_{i=0}^{\infty} \sum_{j=0}^{\infty} P\left[s_1 = i, s_2 = j\right] Z_1^i Z_2^j .$$

At steady state (i.e., when $k \to \infty$), equations (4.3.2.3) and (4.3.2.4) can be written as

Eq 4.3.2.6 $$S_1 = (S_1 - C_1) \cdot U(S_1 - C_1) + A_1$$

Eq 4.3.2.7 $$S_2 = (S_2 - C_2) \cdot U(S_2 - C_2) + A_2$$

Using standard 2-D z-transform techniques, it is possible to derive from equations (4.3.2.6) and (4.3.2.7) the expression for the system contents. In the following we have

Eq 4.3.2.8 $$\begin{aligned} S(Z_1, Z_2) &= E\left[Z_1^{(S_1 - C_1) \cdot U(S_1 - C_1) + A_1} . Z_2^{(S_2 - C_2) \cdot U(S_2 - C_2) + A_2}\right] \\ &= E\left[Z_1^{A_1} Z_2^{A_2}\right] \cdot E\left[Z_1^{(S_1 - C_1) \cdot U(S_1 - C_1)} . Z_2^{(S_2 - C_2) \cdot U(S_2 - C_2)}\right] \end{aligned}$$

We note that the mono-media streams A_1, A_2; and the system contents S_1, S_2 are statistically independent. Let us define $S(Z_1, Z_2) = A(Z_1, Z_2) \cdot G(Z_1, Z_2)$ where

Eq 4.3.2.9 $$G(Z_1, Z_2) = E\left[Z_1^{(S_1 - C_1) \cdot U(S_1 - C_1)} . Z_2^{(S_2 - C_2) \cdot U(S_2 - C_2)}\right]$$

The above expression for $G(Z_1, Z_2)$ can be written as, Eq 4.3.2.10

$$G(Z_1,Z_2)=\sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1}P[s_1=i,s_2=j]+\sum_{i=C_1}^{\infty}\sum_{j=0}^{C_2-1}P[s_1=i,s_2=j]Z_1^{i-C_1}+$$
$$\sum_{i=C_1}^{\infty}\sum_{j=C_2}^{\infty}P[s_1=i,s_2=j]Z_1^{i-C_1}Z_2^{j-C_2}+\sum_{i=0}^{C_1-1}\sum_{j=C_2}^{\infty}P[s_1=i,s_2=j]Z_2^{j-C_2}$$

Solving for $S(Z_1,Z_2)$, the system contents, we finally have

Eq 4.3.2.11

$$S(Z_1,Z_2)=\frac{A(Z_1,Z_2)\left[Z_1^{C_1}Z_2^{C_2}-A_1(Z_1)A_2(Z_2)\right]\sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1}P(i,j)\left(Z_1^{C_1}-Z_1^{i}\right)\left(Z_2^{C_2}-Z_2^{j}\right)}{\left(Z_1^{C_1}Z_2^{C_2}-A(Z_1,Z_2)\right)\left(Z_1^{C_1}-A_1(Z_1)\right)\left(Z_2^{C_2}-A_2(Z_2)\right)}$$

Where we have used $P(i,j)=P[s_1=i,s_2=j]$. Equation (4.3.2.11) contains C_1C_2 unknown constants $P(i,j)$ for $0\le i\le C_1-1$ and $0\le j\le C_2-1$. Even though, there is no analogous theorem for the *fundamental theorem of algebra* [58]-[61] in 2-D [46], we can still solve for the unknown $P(i,j)$'s. From the theory of Algebraic Geometry [47], Multidimensional Residues [48]-[50], and Several Complex Variables [62]-[66], we present a method for the determination of the unknown $P(i,j)$'s.

4.3.3 Determining the unknown joint probabilities *P(i,j)*

The unknown $P(i,j)$ *'s* in equation (4.3.2.11) can be found by using the notion of union *of the analytic lines* described in [48], under the conditions of the *local description theorem.* It is assumed that the joint arrival process of the multi-media session is known. The values of C_1 & C_2 are fixed based on the mean arrivals of the mono-media streams in a multi-media session. In the closed unit bidisk $\overline{U}^2 = \{(Z_1, Z_2): |Z_i| \le 1, \ \ i = 1,2\}$ [46], of 2-D complex z-plane we choose a coordinate system such that Z_1 and Z_2 represent the two coordinate axes. With reference to equation (4.3.2.11), the zeros of the polynomials $[Z_1^{C_1} - A_1(Z_1)]$ and $[Z_2^{C_2} - A_2(Z_2)]$ counting multiplicity lie on the two coordinates Z_1 and Z_2 respectively. The polynomial $[Z_1^{C_1} Z_2^{C_2} - A(Z_1, Z_2)]$ represents an analytic curve in the coordinate system. As shown in Figure (4.2), a *union* of the zeros along the coordinates Z_1, Z_2 and the analytic curve represents $m \ge C_1 C_2$ poles (zeros of denominator) of $S(Z_1, Z_2)$. The be a proper probability generating function the joint pgf $S(Z_1, Z_2)$ converges at least inside the unit bidisk. It then follows that the zeros of the denominator of equation (4.3.2.11) are also zeros of its numerator. We need C_1C_2 equations (one equation is $S(1,1) = 1$) in order to solve for the unknown $P(i, j)$ *'s* present in the numerator. Since m is usually larger than C_1C_2, the remaining (m - C_1C_2) zeros comes from the terms outside the double sum in the numerator of $S(Z_1, Z_2)$, i.e., equation (4.3.2.11). The terms inside the double sum are divided by $(Z_1 - 1)(Z_2 - 1)$, since both $|Z_1| = 1$, and $|Z_2| = 1$, are poles of $S(Z_1, Z_2)$, we call it normalization of the numerator. To proceed further, we find all the zeros necessary in order to obtain the equations in the unknowns. Please remember that one of the equation comes from the total probability condition, i.e., $S(1,1) = 1$, and can be written as follows,

$$\left[(C_1 - A_1'(1))(C_2 - A_2'(1))\right] - \left[\sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1} P(i,j)(C_1 - i)(C_2 - j)\right] = 0.$$

Simultaneous solution of all the equations thus obtained gives the unknown $P(i,j)$'s.

4.3.4 Average System Contents for the two-queue case

For the sake of completeness, comparison and without lack of generality we assume two cases. In the first case we assume that the mono-media streams comprising the multi-media aggregate arrival stream are correlated, i.e., the joint arrival stream is given by equation (4.3.1.1). In the second case we assume that the multi-media arrival stream is composed of two mono-media streams which are independent, i.e., the joint arrival stream is given by equation (4.3.1.4).

4.3.4.1 Case I

By making the substitution $Z_1 = Z_2 = Z$ in equation (4.3.2.11) we solve for $S'(1)$, i.e., the average system contents. The resulting equation is given in terms of the number of servers, the joint probabilities $P(i,j)$'s, the first and second moments of the joint arrival stream $A(Z_1, Z_2)$ and the mono-media streams $A_1(Z_1)$ and $A_2(Z_2)$. The general expression is given below.

$$K = \sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1}\left\{\left(\tfrac{(C_1-1)C_1}{2} - \frac{i(i-1)}{2}\right)(C_2 - j) + \left(\tfrac{(C_2-1)C_2}{2} - \frac{j(j-1)}{2}\right)(C_1 - i)\right\}P(i,j)$$

$$\sigma = \frac{\left(C_1 + C_2 - A_1'(1) - A_2'(1)\right) \cdot \kappa}{\left(C_1 + C_2 - A'(1)\right)\left(C_1 - A_1'(1)\right)\left(C_2 - A_2'(1)\right)}$$

$$\omega = \sum_{i=0}^{C_1-1} \sum_{j=0}^{C_2-1} \left[(C_1 - i)(C_2 - j)\right] P(i,j)$$

$$\alpha = \frac{\dfrac{A'(1)}{C_1 + C_2 - A'(1)} - \dfrac{\frac{1}{2}(C_1 + C_2 - 1)(C_1 + C_2) - \frac{A''(1)}{2}}{\left[C_1 + C_2 - A'(1)\right]^2}}{C_1 - A_1'(1)}$$

$$\beta = \frac{\dfrac{1}{2}(C_1 - 1)C_1 - \dfrac{A_1''(1)}{2}}{\left(C_1 + C_2 - A'(1)\right)\left[C_1 - A_1'(1)\right]^2}$$

$$\chi = \frac{\dfrac{1}{2}(C_2 - 1)C_2 - \dfrac{A_2''(1)}{2}}{\left(C_1 + C_2 - A'(1)\right)\left(C_1 - A_1'(1)\right)\left[C_2 - A_2'(1)\right]^2}$$

$$\gamma = \frac{\dfrac{1}{2}(C_1 + C_2 - 1)(C_1 + C_2) - A_1'(1)A_2'(1) - \dfrac{A_1''(1)}{2} - \dfrac{A_2''(1)}{2}}{\left(C_1 + C_2 - A'(1)\right)\left(C_1 - A_1'(1)\right)\left(C_2 - A_2'(1)\right)}$$

Eq 4.3.4.1
$$S'(1) = \sigma + \omega\left\{\gamma + \left(C_1 + C_2 - A_1'(1) - A_2'(1)\right)\left(\frac{\alpha - \beta}{C_2 - A_2'(1)} - \chi\right)\right\}$$

The above result, i.e., equation (4.3.4.1) is used in chapter 5 for numerical results representing the average system contents for the correlated batch arrivals.

4.3.4.2 Case II

Here we use a similar procedure as in case I above, to generate an expression for the average system contents when the multi-media arrival stream is treated as independent batch arrivals. The average system contents, i.e., $S'(1)$, can be represented by:

$$\kappa = \sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1}\left\{\left(\frac{(C_1-1)C_1}{2}-\frac{i(i-1)}{2}\right)(C_2-j)+\left(\frac{(C_2-1)C_2}{2}-\frac{j(j-1)}{2}\right)(C_1-i)\right\}P(i,j)$$

$$\sigma = \frac{\kappa}{(C_1-A_1'(1))(C_2-A_2'(1))}$$

$$\omega = \sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1}[(C_1-i)(C_2-j)]P(i,j)$$

$$\alpha = \frac{A_2'(1)}{(C_1-A_1'(1))(C_2-A_2'(1))}$$

$$\beta = \frac{\dfrac{A_1'(1)}{C_1-A_1'(1)}-\dfrac{\frac{1}{2}(C_1-1)C_1-\frac{A_1''(1)}{2}}{[C_1-A_1'(1)]^2}}{C_2-A_2'(1)}$$

$$\chi = \frac{\frac{1}{2}(C_2-1)C_2-\frac{A_2''(1)}{2}}{(C_1-A_1'(1))[C_2-A_2'(1)]^2}$$

Eq 4.3.4.2
$$S'(1) = \sigma+\omega\{\alpha+\beta-\chi\}$$

The above result, i.e., equation (4.3.4.2) is used in chapter 5 for numerical results representing the average system contents for the independent batch arrivals.

This concludes the proposal for the Two-queue model for the ATM/BISDN network. As a passing remark even though at first this may seem a difficult and cumbersome procedure but it is worth the pain. The reason is that we have gotten a closed form solution for the two-queue model and it only depends upon the multi-media arrival streams and the number of servers to be used for each type/class of mono-media. The rest is just generating numbers for comparison or (pleasure!). The benefit is that we have a General Independent (GI) arrival model which only requires the arrival stream specification.

In the following chapter we present detailed numerical analysis for the proposed two-queue model. Several numerical examples illustrating the main features and the overall performance analysis of our proposed model are presented. These results are then compared with that of the conventional analysis GI-D-c results. Both parametric numerical examples as well as experimental results are presented.

The numerical results represents the graphs of equations (4.3.4.1 & 4.3.4.2). Both equations (4.3.4.1) and (4.3.4.2) depend upon the number of servers C_1, C_2, and the first and second moments of the multi-media stream distribution $A(Z_1,Z_2)$, the mono-media stream distributions $A_1(Z_1)$ and $A_2(Z_2)$. The values of the constants $P(i,j)$'s are determined by using the procedure outlined in subsection 4.3.3. It is a very involved procedure and solving polynomials of the order of degree 30 or even higher is very common. The author used self written C++ code and MATHEMATICA® to solve the system of equations produced by the experimental multi-media arrival streams.

Chapter 5

Numerical Results

In this chapter different numerical examples for average system contents vs correlation coefficient and/or time slot length (in the case of experimental analysis) are given; both correlated and independent multi-media arrival streams are considered. The results are given for different values of C_1 and C_2. A comparison between the proposed two-queue model and the known results of the GI-D-c [43] is also presented. All the results are generated either by self written C code, Numerical Recipes [74], or MATHEMATICA® [73].

5.1 Example # 1

Let the following table represent a parametric pdf for a general independent arrival

Z_2' ⇨

Z_1' ⇩ i \ j	0	1	2
0	4p	p	p
1	p	1/4-2p	1/4-2p
2	p	1/4-2p	1/4-2p

process representing the multi-media traffic. We can write the following equations for the multi-media stream distribution $A(Z_1,Z_2)$, the mono-media stream distributions $A_1(Z_1)$ and $A_2(Z_2)$ respectively.

$$A(Z_1,Z_2)=4p+pZ_1+pZ_1^2+pZ_2+\left(\frac{1}{4}-2p\right)Z_1Z_2+\left(\frac{1}{4}-2p\right)Z_1^2Z_2+pZ_2^2+\left(\frac{1}{4}-2p\right)Z_1Z_2^2+\left(\frac{1}{4}-2p\right)Z_1^2Z_2^2$$

$$A_1(Z_1) = A(Z_1,1) = 6p + \left(\frac{1}{2} - 3p\right)Z_1 + \left(\frac{1}{2} - 3p\right)Z_1^2$$

$$A_2(Z_2) = A(1,Z_2) = 6p + \left(\frac{1}{2} - 3p\right)Z_2 + \left(\frac{1}{2} - 3p\right)Z_2^2$$

The first and second moments of the above equations along with different values of C_1 and C_2 are substituted in equations (4.3.4.1) and (4.3.4.2) to obtain the average system contents. Let us select four different configurations of the output buffers, corresponding to the values of C_1 = 1, 2, and C_2 = 1, 2, respectively. The parameter p should be $0.055 < p < 0.125$. The upper bound is obtained form the joint pdf table while the lower bound is obtained from the stability condition for the average system contents. The correlation coefficient is $\rho = \frac{9p - 81p^2}{0.25 + 12p - 81p^2}$, as explained in Appendix I. Table (5.1) shows the average arrivals, variance of the arrivals, correlation coefficient between the two mono-media streams, and the average system contents for the correlated and independent cases for various server configurations. The correlation coefficient and the average system content plots are shown in Figures (5.1-5.4) for the above mentioned server configurations. All of the figures reveal that, as the correlation coefficient assume values from negative to positive the average system contents increases fairly rapidly. *The system can become unstable if the average number of arrivals in the mono-media streams becomes larger than the number of servers available at the output.* As shown in Table (5.1), the average arrivals in both mono-media streams are less than the number of servers in a given output configuration. Figure (5.1) shows that, as the server configuration changes, the average system contents vs correlation also changes. In particular notice that, the curves for the correlated and the independent arrival streams are fairly closer, the reason is that this particular parametric pdf is highly correlated to begin with. If a different pdf is assumed then the curves might be far apart. Furthermore, since the pdf is symmetric the curves for server configurations [$C_1 = 1$, $C_2 = 2$] and [$C_1 = 2$, $C_2 = 1$] are the same.

5.1.1 Example # 1: Two-queue vs. Single-queue

Figures (5.1-5.4) presents a comparison between the proposed two-queue model and the known results of GI-D-c [43]. Please note that, the multi-media batch arrival streams considered in the literature do not distinguish between the individual batches of the mono-media arrival streams. From the table given in section 5.1, we can write the equation for *A(Z)* by substituting $Z_1=Z_2=Z$, as follows:

$$A(Z) = A(Z,Z) = 4p + 2pZ + \frac{Z^2}{4} + \left(\frac{1}{2} - 4p\right)Z^3 + \left(\frac{1}{4} - 2p\right)Z^4$$

Substituting A(Z) in [equation 4.8, 43], we solve for the system contents of GI-D-c [43]. Figure (5.1) shows the results for $C_1 = C_2 = 1$ server configuration case. The proposed two-queue model clearly outperforms the GI-D-c model. The GI-D-c model under estimates the system contents. GI-D-c model performance is even worst for the $C_1 = 1$, $C_2 = 2$ or $C_1 = 2$, $C_2 = 1$, server configurations, Fig (5.2).

For server configuration $C_1 = C_2 = 2$, Fig (5.3), it seems that the GI-D-c model performs as the proposed two-queue model if the multi-media arrival streams are assumed independent. But it is not true. The reason is that since the assumed multi-media traffic in section 5.1 is symmetrical therefore it might look like the GI-D-c model performance is closer to the proposed two-queue model. Had it not been a symmetrical arrival stream the results would have been different. Figure (5.4) shows all the server configurations for the two-queue model and the single-queue model (GI-D-c).

5.2 Example # 2

In this example we present the results based on pdf's obtained by conducting experiments as explained in chapter 3. A typical pdf is shown in Fig (3.3). The same procedure is applied as explained above. Figures (5.5-5.11) provides details of the numerical analysis. The experimental pdf's are very large and sparse. To solve problems of this magnitude a very fast computer and a true multitasking operating system such as UNIX is needed. MATHEMATICA 3.0 was used to generate all the results. To generate the results of Figures (5.5)-(5.11) the following procedure was used;

- Pdf's of the experimental data were generated for time-slot lengths of 20-50 msec.
- Equations for the multi-media stream distribution $A(Z_1,Z_2)$, the mono-media stream distributions $A_1(Z_1)$ and $A_2(Z_2)$ were obtained from the pdf's.
- The average arrivals in both $A_1(Z_1)$ and $A_2(Z_2)$ are needed since the number of servers for both audio and video depends on the average arrival in a given time-slot.

The rest of the procedure is outlined in section 4.3.3. Solution for the average system contents was obtained by using equation 4.3.4.1 (Correlated Batches) and equation 4.3.4.2 (Independent Batches) for all the pdf's from 20-50 msec. The correlation between the two mono-media streams was also obtained from the same pdf's. Observing the Figures (5.5)-(5.11) might misguide the reader that the two results of the correlated batches and the independent batches are very close. But looking at the cross correlation curves in Figures (3.11)-(3.12) reveals that: (i) the correlation among the mono-media streams oscillates between negative and positive values; (ii) compression and Ethernet as a transport method has to do with it; (iii) we are working on the range of pdf's and hence correlation from 20-50 msecs time-slot range. Figure (5.13) shows that in the range where time-slot length is varied from 20-50 msec the correlation moves from -0.145 to 0.158. The correlation seems to be increasing very rapidly. In such situations the average system contents for the correlated batches and the independent batches will naturally be very close, because even though we say (for the sake of completeness and comparison with the conventional analyses) that the multi-media traffic is independent but the reality is that it is not so naturally the results of equations 4.3.4.1 and 4.3.4.2 will look almost similar.

The comparisons between the proposed two-queue model and the single-queue model reveals the same conclusions we reached earlier, i.e., that the single-queue model under or overestimates the average system contents. Figures (5.9)-(5.11) shows the comparison of the two-queue model and the single-queue models.

Thus we have established the fact that GI-D-c model and the like under or overestimate the average system contents depending on correlation among the mono-media streams and the server configurations.

Chapter 6

Jitter Distribution

6.1 Introduction

Jitter is defined to be the maximum difference between end-to-end delays experienced by any two consecutive packets [4]. Hence jitter implies a varying packet (and LDU) rate at the receiver, i.e., when cells received at the inputs of the buffer in the same time slot are transmitted over different time slots.

Several synchronization control methods have been proposed for solving the problem of stream synchronization. In [2], Sreenan suggests elastic buffers and synchronization server agent models for intra-stream synchronization in continuous media. Similarly, the issue of skew, the time difference between related audio and video LDU's [3], caused by the buffer variations in stream start times; in the context of inter-stream synchronization is treated using group agent models. The use of discrete stream events are also advocated for event based synchronization in that reference. In [51], Li et al. considers synchronization of parallel data streams using the notion of "multi-media data segment" in a real-time multi-media data delivery system (SRTDD). Data in different media are assumed to be transferred through the network over different parallel connections according to their traffic characteristics and transmission requirements. However, possible correlation among the mono-media streams is not considered. In [52], Nicolaou presents a scheme for implementing synchronization among related streams by inserting synchronization points in each individual stream. In [53], Ferrari proposes a delay-jitter control scheme that insures media synchronization as long as a bound on the delay can be guaranteed and the source and sink clocks are kept in synchrony. Every node on the transmission path in the network is involved in the synchronization control by regulating the random delay that has been introduced to an arriving packet by the previous hop. Such a technique might be difficult to implement on an ATM network because it introduces too much workload on the intermediate nodes that carry thousands of connections. In [54], Escobar et al. address a flow synchronization control protocol by implementing an end-to-end synchronization delay. The random delay experienced by each packet is regulated at the sink node. The actual delay experienced by every data packet cannot exceed the preset end-to-end delay for the network connection and a global accurate clock is required. In [55], Little et al. propose computing schedules before transmitting the data packets. The multimedia traffic is delivered according to the schedule to reach the sink in time for the presentation. Because of the random disturbance introduced by the networks, however, scheduling and predicting the traffic is not sufficient to maintain a synchronized multistream data delivery. Therefore, a compensation mechanism at the receiver is necessary when synchronization errors occur.

6.1 Jitter distribution for the two-queue case

Figure (6.1) shows, for example, two mono-media streams containing two test cells received in the same time slot. If the two test cells depart in the same time slot, then there will be no jitter corresponding to this sample. This can occur if the queue lengths for both streams are such that $|T_1 - T_2| = 0$, where $T_i = R_i + Q_i$; Q_i corresponds to the *queue length* at the beginning of the time slot of arrival of the two test cells; and R_i corresponds to the *forward recurrence number* in the mono-media stream "*i*". We can then write

$$\begin{cases} T_1 = R_1 + Q_1 \\ T_2 = R_2 + Q_2 \end{cases}.$$

Assuming that $R(Z_1, Z_2) = E\left[Z_1^{R_1} \cdot Z_2^{R_2}\right]$ represents the joint probability generating function (jpgf) of the forward recurrence numbers. Since R_i and Q_i are independent, and with $S(Z_1, Z_2)$ as the jpgf of queue lengths, we obtain the jpgf of T_1 and T_2 as follows:

Eq 6.1.1 $$T(Z_1, Z_2) = E\left[Z_1^{T_1} \cdot Z_2^{T_2}\right] = R(Z_1, Z_2) \cdot S(Z_1, Z_2)$$

From equation (6.1.1) it is straightforward to determine the distribution of jitter (i.e., $|T_1 - T_2|$). In the following, our goal is to determine the jpgf of the forward recurrence numbers, $R(Z_1, Z_2)$. Suppose A_i represents the batch size, and R_i^n represents the number of cells ahead of the test cell in the mono-media stream "*i*", then:

Eq 6.1.2 $$R_i^{n+1} = R_i^n - C_i \qquad R_i^n \geq C_i$$

Eq 6.1.3 $$R_i^{n+1} = A_i - C_i \qquad R_i^n < C_i$$

Suppose $U(R_i^n) = \begin{cases} 1 & \text{if} \quad R_i^n < C_i \\ 0 & \text{if} \quad R_i^n \geq C_i \end{cases}$

then we can write equation (6.1.2) and (6.1.3) as follows,

Eq 6.1.4 $$R_i^{n+1} = \left[R_i^n - C_i\right]\left[1 - U(R_i^n)\right] + \left[A_i - C_i\right]\left[U(R_i^n)\right]$$

The joint probability generating function (pgf) of the forward recurrence numbers at steady state, (i.e., as $n \to \infty$) can be written from equation (6.1.4) as follows:

Eq 6.1.5
$$\begin{aligned} R(Z_1, Z_2) &= E\left[Z_1^{R_1} \cdot Z_2^{R_2}\right] \\ &= E\left[Z_1^{(R_1 - C_1)(1 - U(R_1)) + (A_1 - C_1)U_1} \cdot Z_2^{(R_2 - C_2)(1 - U(R_2)) + (A_2 - C_2)U_2}\right] \end{aligned}$$

Using equations (4.3.1.2), (4.3.1.3) and the fact that total probability $\sum_{k=0}^{\infty}\sum_{l=0}^{\infty} P[a_1 = k, a_2 = l] = 1$, we can write equation (6.1.5) as:

$$R(Z_1, Z_2) = \frac{1}{Z_1^{C_1} Z_2^{C_2}} \left\{ \begin{array}{l} \sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1} P[r_1 = i, r_2 = j] \cdot A(Z_1, Z_2) + \sum_{i=C_1}^{\infty}\sum_{j=C_2}^{\infty} P[r_1 = i, r_2 = j] Z_1^i Z_2^j + \\ \sum_{i=C_1}^{\infty}\sum_{j=0}^{C_2-1} P[r_1 = i, r_2 = j] Z_1^i \cdot A_2(Z_2) + \sum_{i=0}^{C_1-1}\sum_{j=C_2}^{\infty} P[r_1 = i, r_2 = j] Z_2^j \cdot A_1(Z_1) \end{array} \right\}$$

The above expression for the joint probability generating function for the forward recurrence numbers reduces to:

Eq 6.1.6

$$R(Z_1,Z_2)=\frac{\sum_{i=0}^{C_1-1}\sum_{j=0}^{C_2-1}q(i,j)}{\left[Z_1^{C_1}Z_2^{C_2}-1\right]}\left\{\begin{array}{l} A(Z_1,Z_2)-\left[Z_1^iZ_2^j-Z_1^i-Z_2^j\right]-\left[\dfrac{A_1(Z_1)-Z_1^i}{Z_1^{C_1}-1}\right]- \\ \left[\dfrac{A_2(Z_2)-Z_2^j}{Z_2^{C_2}-1}\right]+A_2(Z_2)\left[\dfrac{A_1(Z_1)-Z_1^i}{Z_1^{C_1}-1}-1\right]+ \\ A_1(Z_1)\left[\dfrac{A_2(Z_2)-Z_2^j}{Z_2^{C_2}-1}-1\right] \end{array}\right\}$$

where we have used $q(i,j)=P[r_1=i,r_2=j]$. The final expression for the joint probability generating function of jitter distribution can be obtained by combining equations (4.3.2.11) and (6.1.6) into equation (6.1.1).

The final result for the jitter distribution $|T_1 - T_2|$ can be obtained numerically by first finding the values of T_1 and T_2. This can be done in a similar way as we did for the average system contents. The numerical results will be published in Globcom,97.

Chapter 7

Conclusions and future work

We have presented novel queueing models for determining the average system contents for the two-queue case. These model can be used to study the impact of correlation among batch arrivals from mono-media sources comprising a multi-media session at the ATM multiplexer. The models have established that when dealing with bursty or variable bit rate (VBR) multi-media traffic it is essential that correlation among the mono-media streams be properly taken into account. Furthermore, assumptions about the arrival processes based upon the known distributions will lead to the wrong conclusions. Our investigations have shown that two-queue model is very general and lends itself to be used in a diverse applications fields. The ATM network would be improperly operated and managed if accurate traffic data were not available, even in the case where network resources were dimensioned and the quality of service was evaluated with high accuracy at the design stage. In the future we will continue analytical modeling and numerical analysis, for the jitter distribution both for the two-queue and single queue cases. Our intuition is that jitter will be higher in the single queue case.

Appendix I

Computing Correlation Coefficient

The correlation coefficient between two random variables X and Y is [34], and [56]-[57]

$$\rho_{x,y} = \frac{Cov(X,Y)}{\sigma_x \sigma_y}$$

The correlation coefficient will be undefined if either of the variances is 0.

EXAMPLE.

Suppose that $A(Z_1, Z_2)$ is represented by the joint pdf given in chapter 5. Then the following table can be used to lists the expected values of the mono-media arrival streams.

u	$f_{z_1}(u)$	$f_{z_2}(u)$	$u \cdot f_{z_1}(u)$	$u \cdot f_{z_2}(u)$
0	6p	6p	0	0
1	0.5-3p	0.5-3p	1(0.5-3p)	1(0.5-3p)
2	0.5-3p	0.5-3p	2(0.5-3p)	2(0.5-3p)
			$E(z_1) = \frac{3}{2} - 9p$	$E(z_2) = \frac{3}{2} - 9p$

From the above table we can find the variance of the two mono-media arrival streams as follows:

$$\sigma_{z_1}^2 = [u - E(z_1)]^2 \cdot f_{z_1}(u) = \sum_{i=0}^{2}\left[i - \left(\frac{3}{2} - 9p\right)\right]^2 \cdot f_{z_1}(u)$$
$$= 0.25 + 12p - 81p^2$$

$$\sigma_{z_2}^2 = [u - E(z_2)]^2 \cdot f_{z_2}(u) = \sum_{i=0}^{2}\left[i - \left(\frac{3}{2} - 9p\right)\right]^2 \cdot f_{z_2}(u)$$
$$= 0.25 + 12p - 81p^2$$

The covariance is computed by the following equation:

$$Cov(z_1, z_2) = E[(z_1 - E(z_1))(z_2 - E(z_2))]$$
$$= 9p - 81p^2$$

And finally the correlation coefficient is given by:

$$\rho = \frac{COv(z_1, z_2)}{\sqrt{\sigma_{z_1}^2 \sigma_{z_2}^2}} = \frac{9p - 81p^2}{0.25 + 12p - 81p^2}$$

ATM

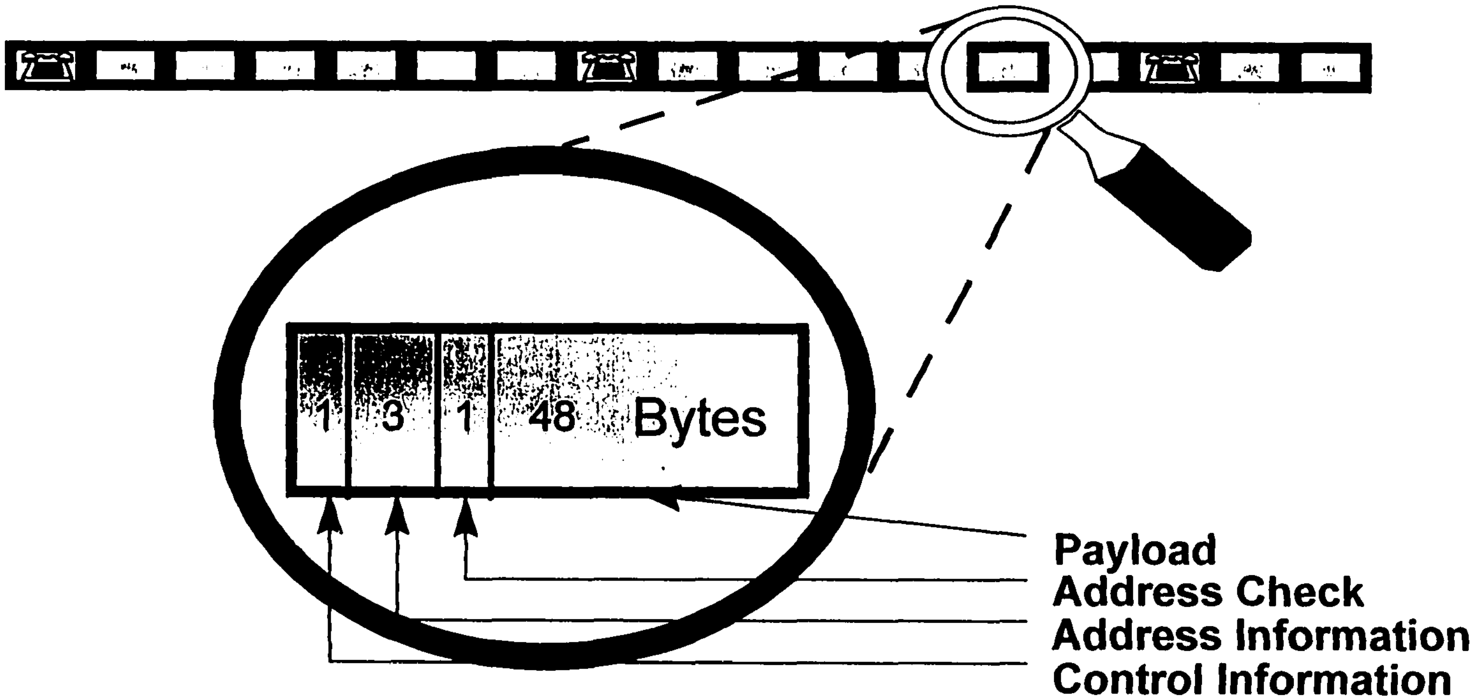


Figure 2.1

ATM Packet

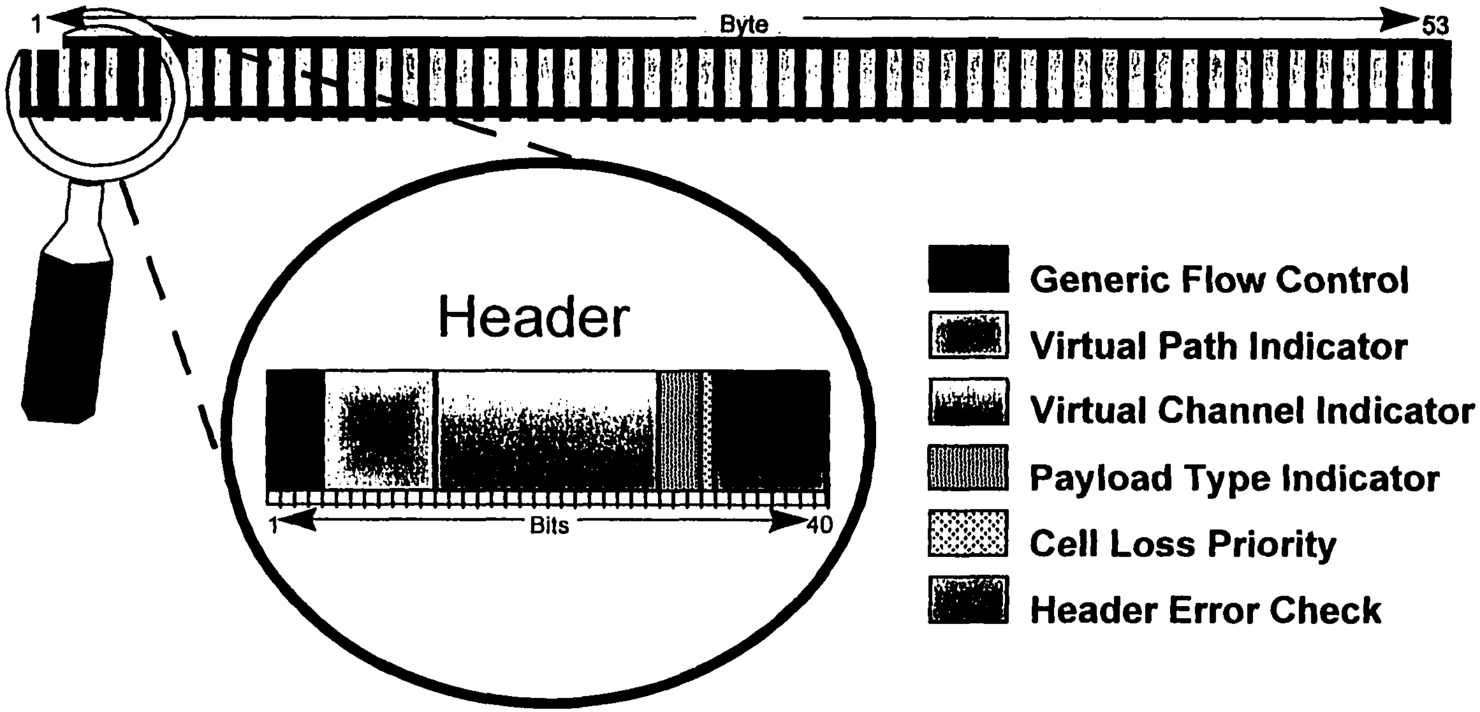
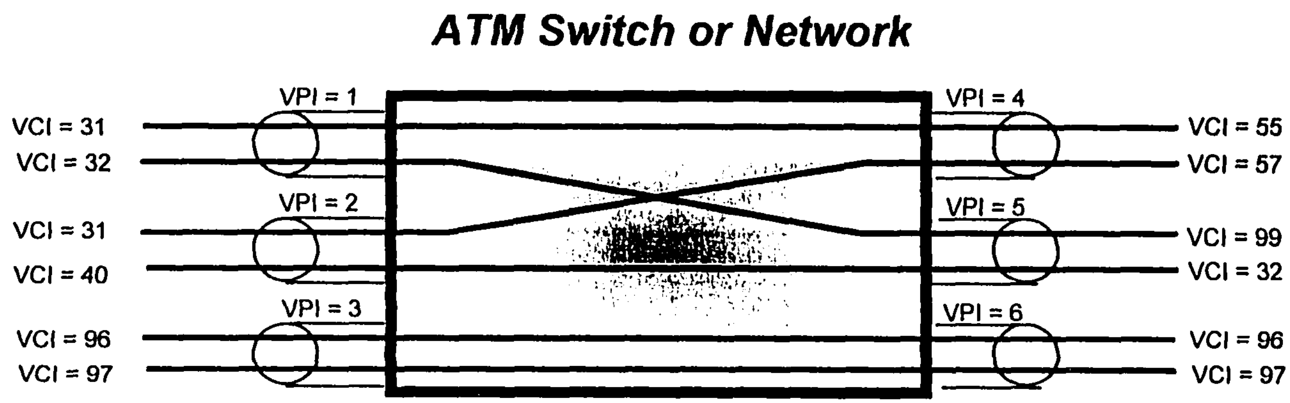


Figure 2.2

Virtual Paths and Virtual Channels



- Bundles of Virtual Channels are switched via Virtual Paths.
- Virtual Path service from a carrier allows reconfiguration of Virtual Channels without service orders to carry.

Figure 2.3

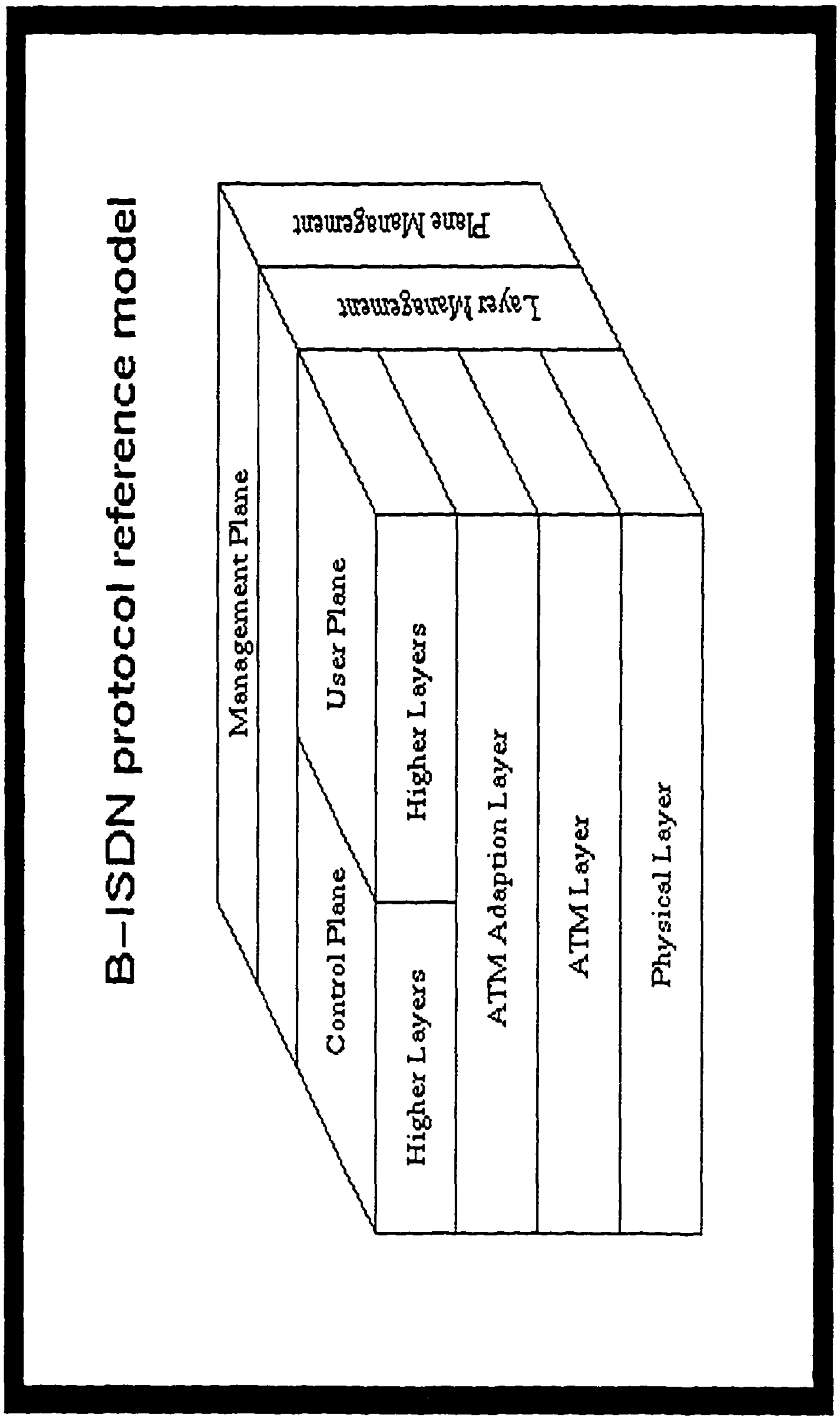
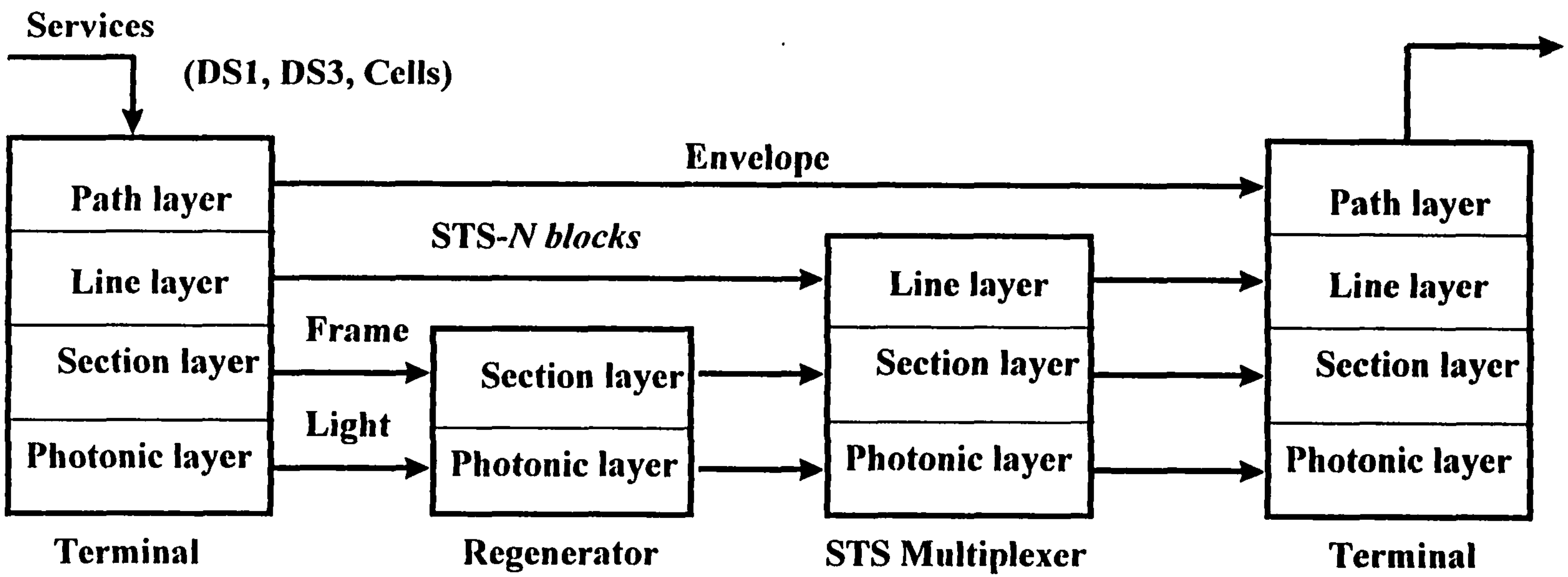
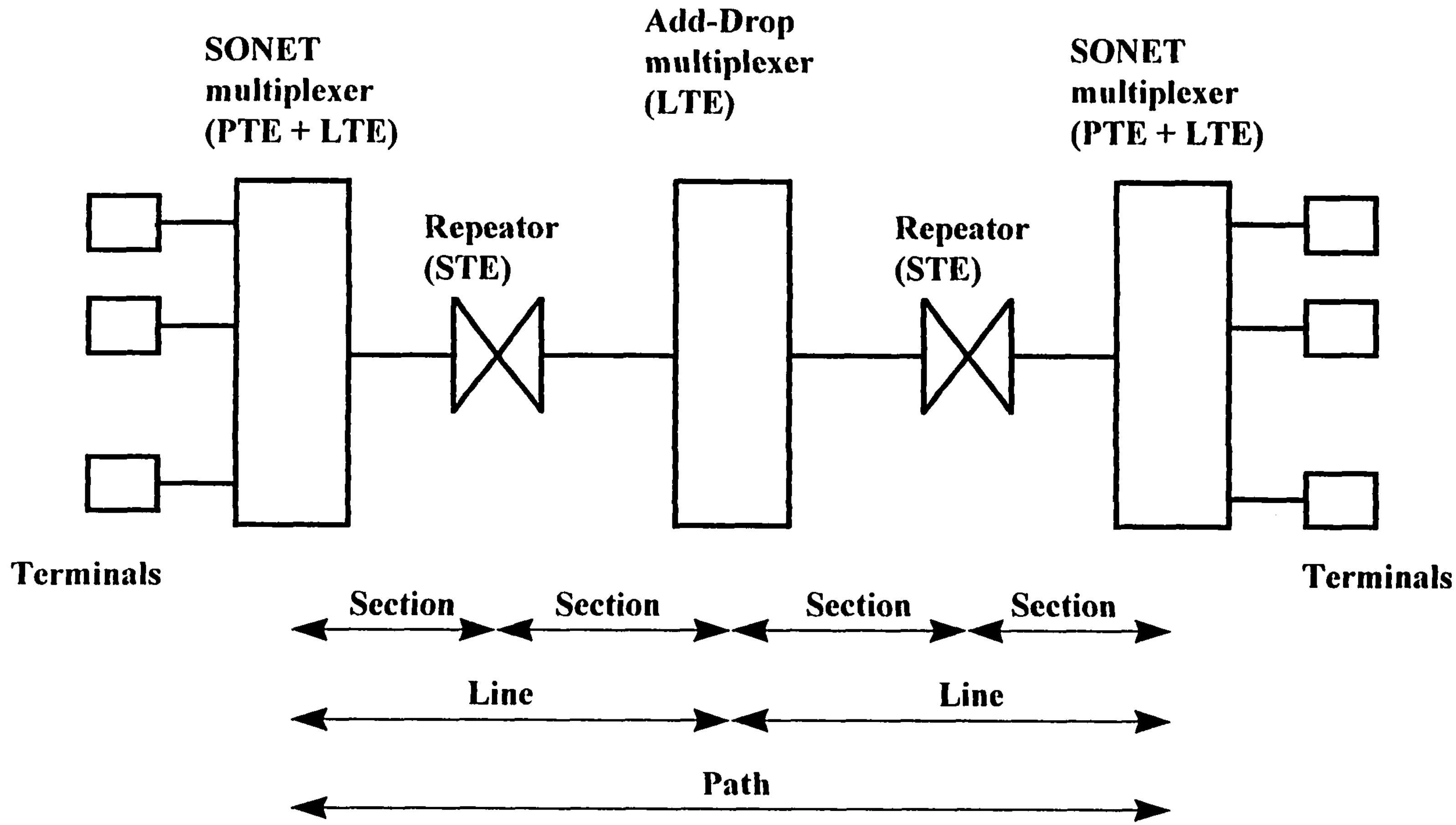

B-ISDN protocol reference model
Plane Management
Layer Management
Management Plane
Control Plane
User Plane
Higher Layers
Higher Layers
ATM Adaption Layer
ATM Layer
Physical Layer

Figure 2.4



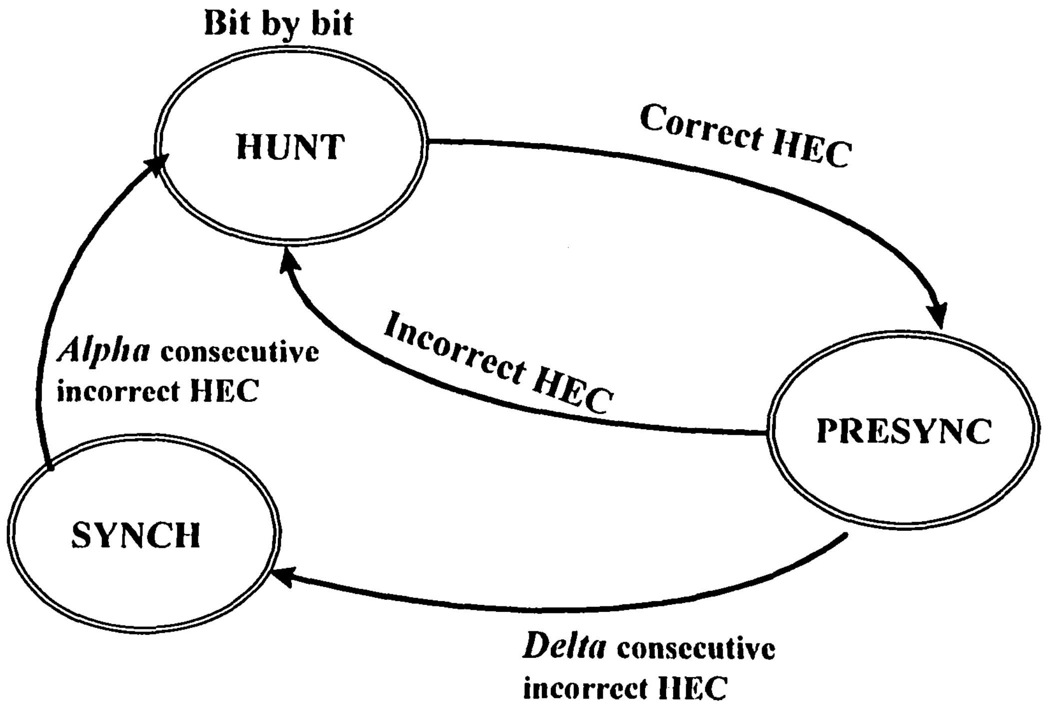
Logical hierarchy

Figure 2.5



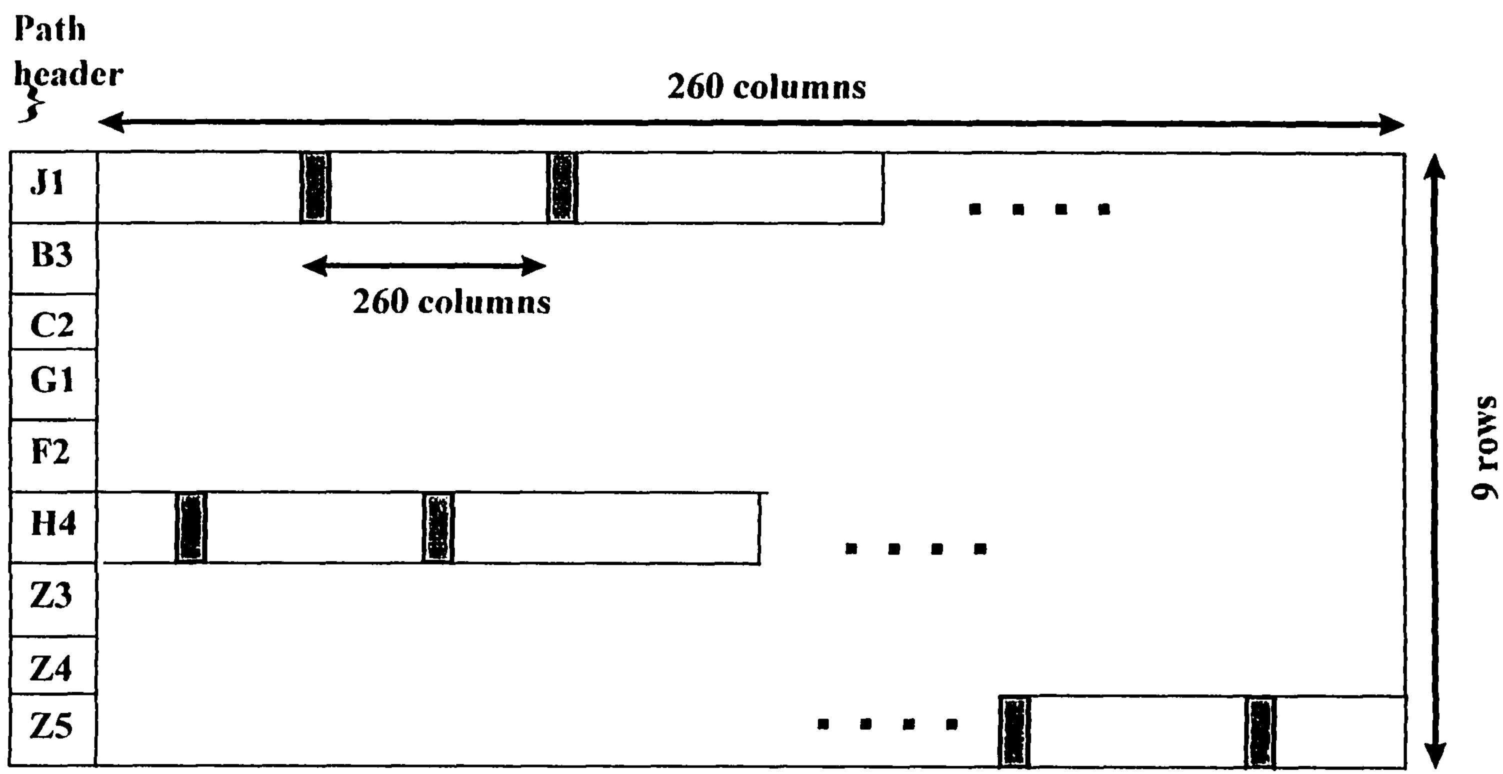
Physical hierarchy

Figure 2.6



Cell Delineation State Diagam

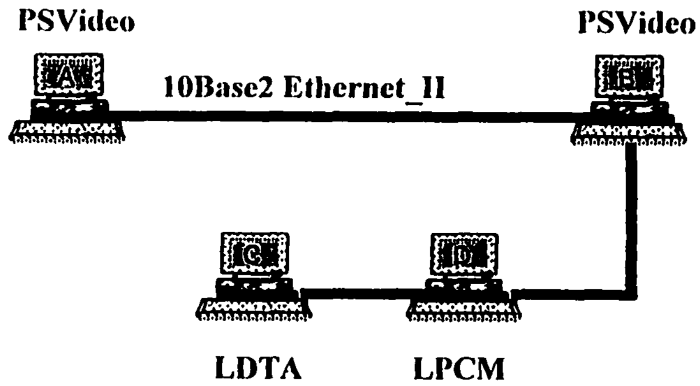
Figure 2.7



STM-1 Payload for SDH-Based ATM Cell Transmission

Figure 2.8

Network Architecture



A & B - Intel's ProShare Personal Conferencing Video System 150 (PSVideo).

C - Intel's LANDesk Traffic Analyst (LDTA)

D - Intel's LANDesk Personal Conferencing Manager (LPCM)

Figure 3.1

Novell Frame Type Designation: Ethernet_II

Ethernet_II Frame over IPX Transport Protocol

DIX Ethernet Header (14 bytes at offset 0)

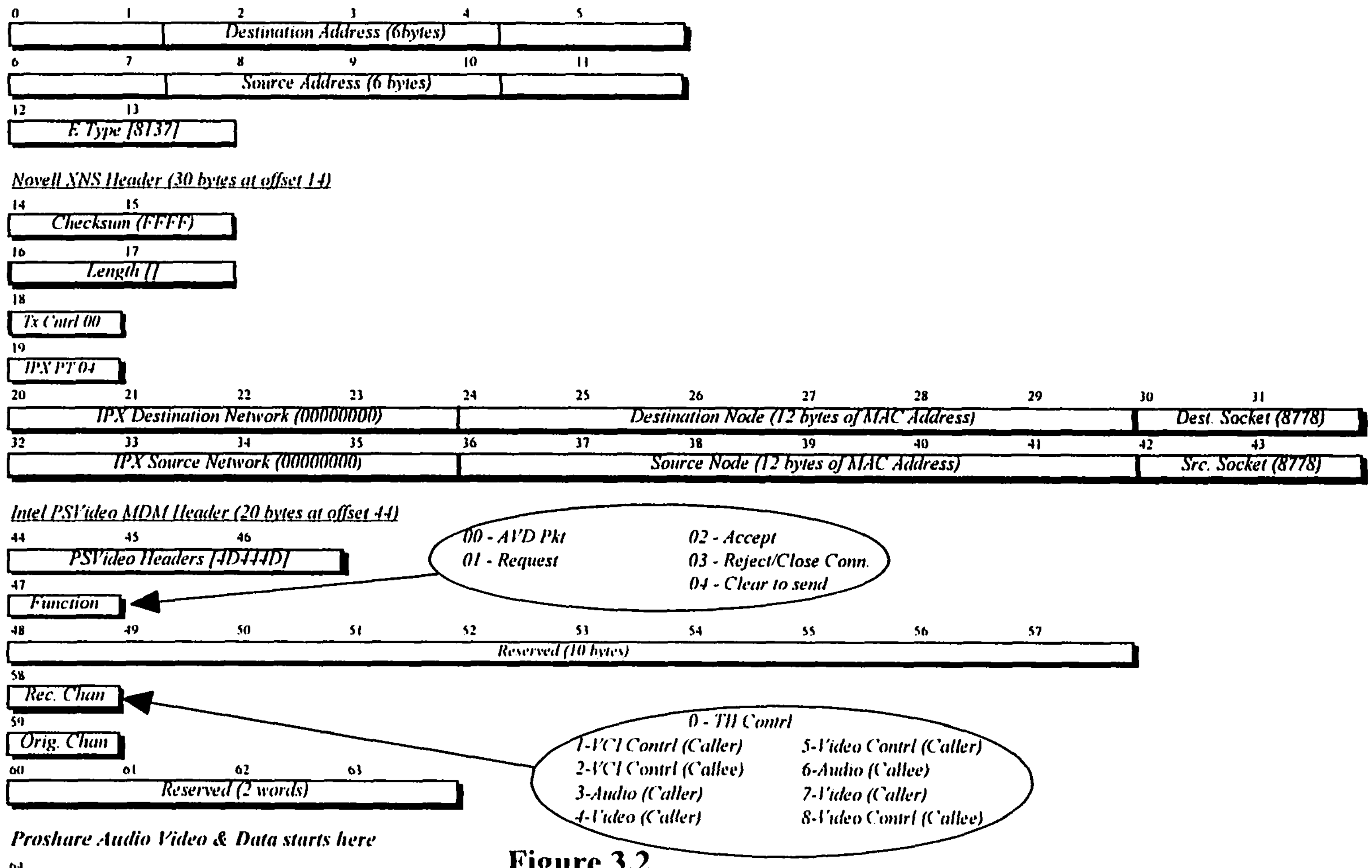


Figure 3.2

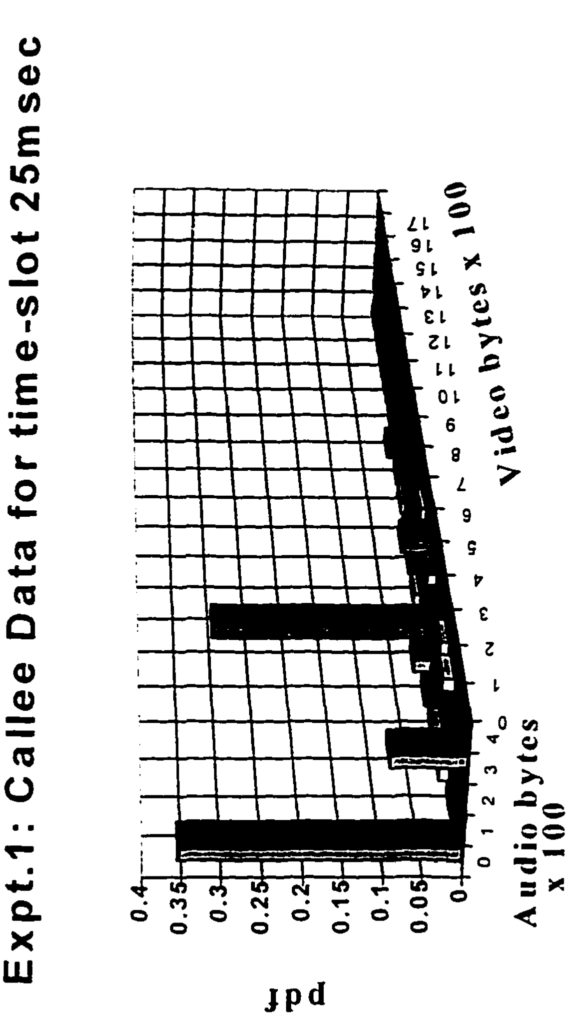

Expt.1: Callee Data for time-slot 25msec
pdf
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
Audio bytes x 100
0 1 2 3 4
Video bytes x 100
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17

Figure 3.3

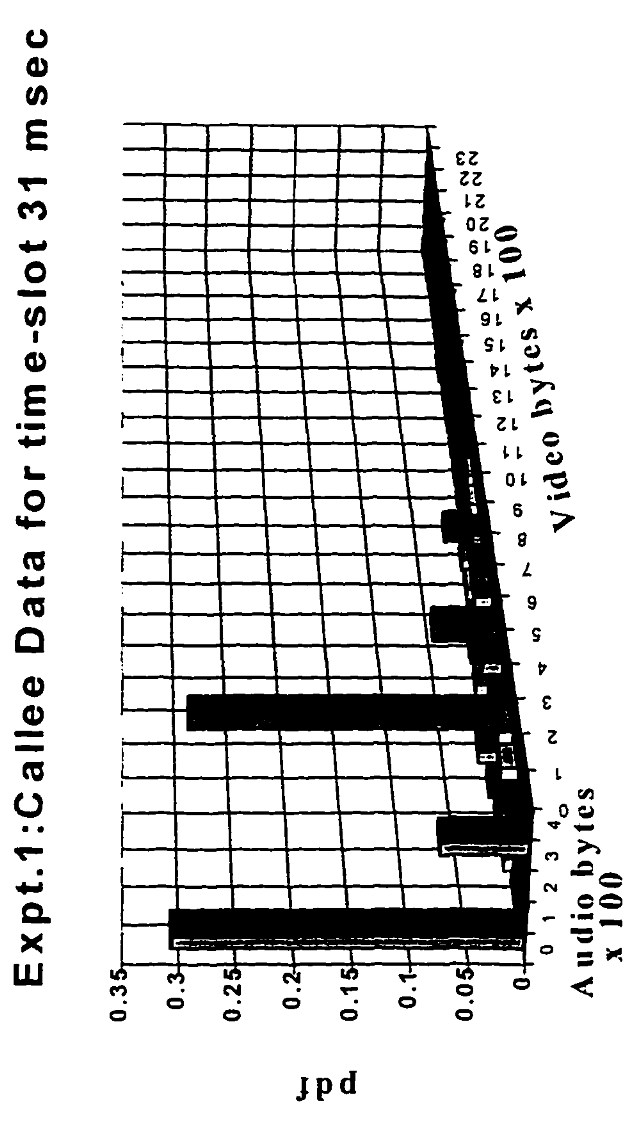

Expt.1:Callee Data for time-slot 31 msec
pdf
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
Audio bytes x 100
Video bytes x 100

Figure 3.4

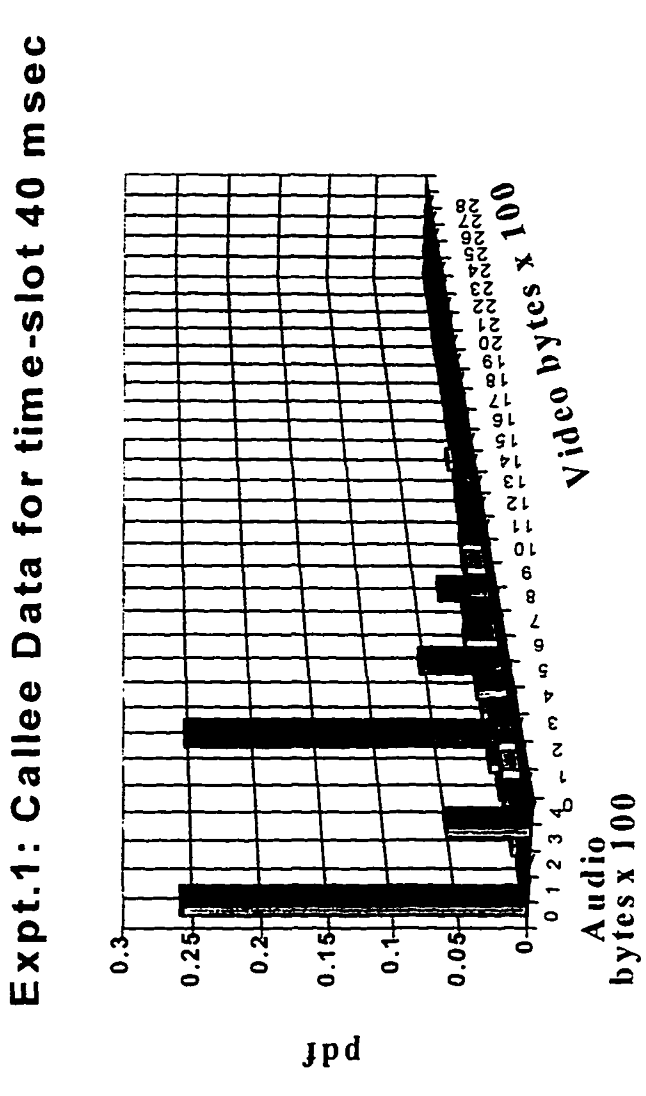

Expt.1: Callee Data for time-slot 40 msec
pdf
0.3
0.25
0.2
0.15
0.1
0.05
0
Audio
bytes x 100
Video bytes x 100

Figure 3.5

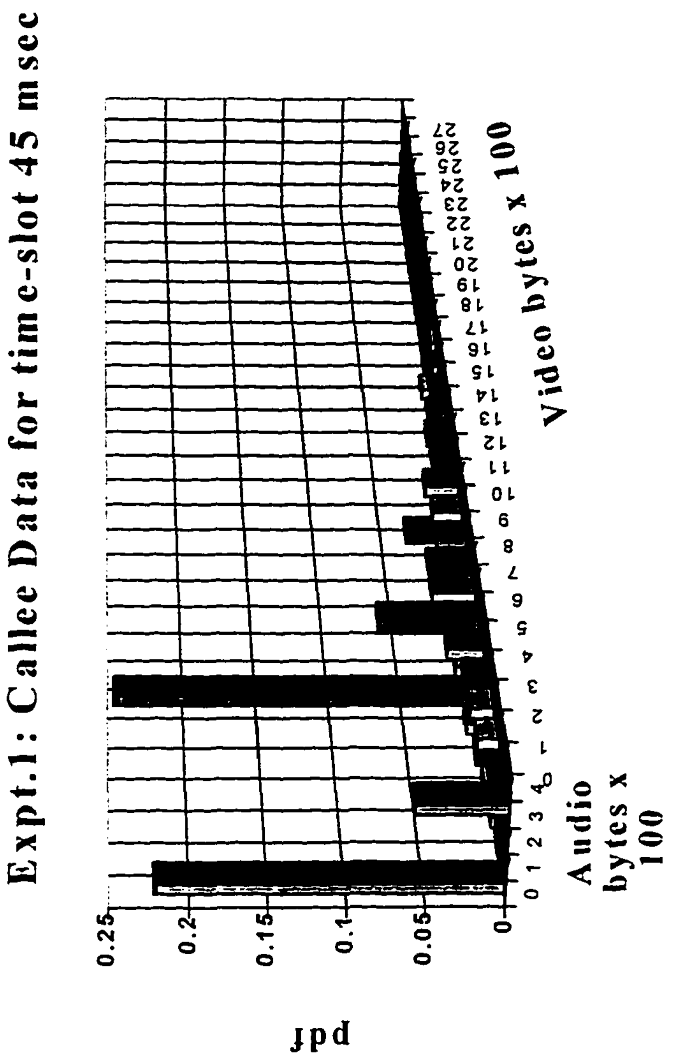

Expt.1: Callee Data for time-slot 45 msec
pdf
0.25
0.2
0.15
0.1
0.05
0
Audio bytes x 100
Video bytes x 100

Figure 3.6

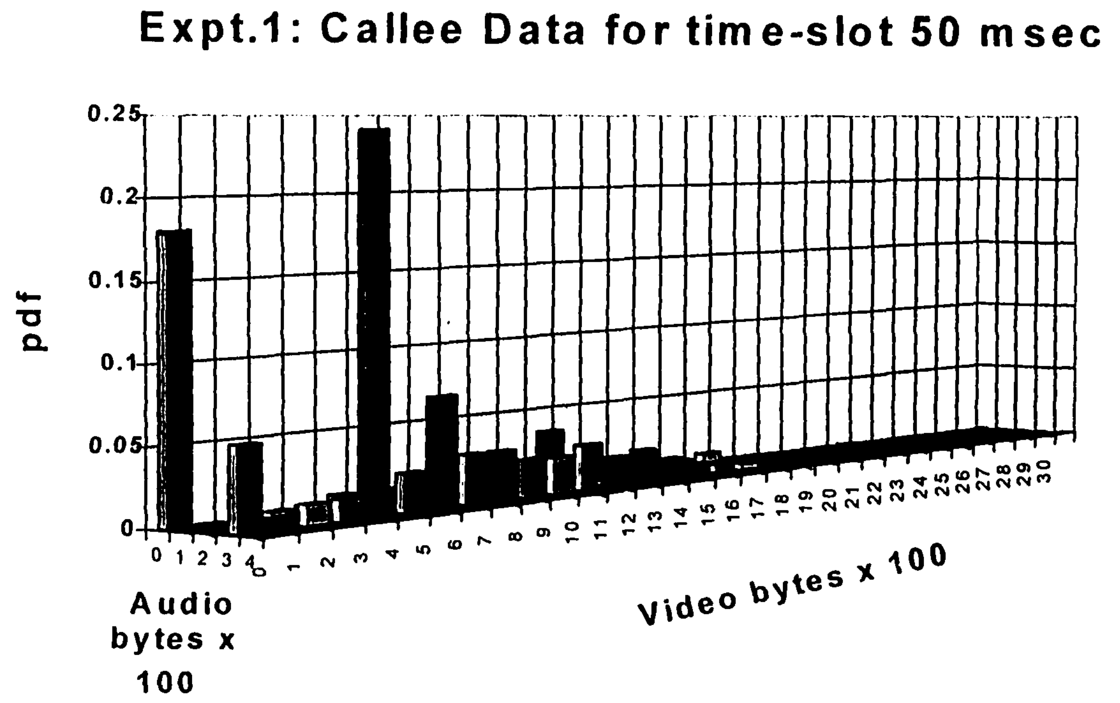

Expt.1: Callee Data for time-slot 50 msec
pdf
0.25
0.2
0.15
0.1
0.05
0
Audio bytes x 100
Video bytes x 100

Figure 3.7

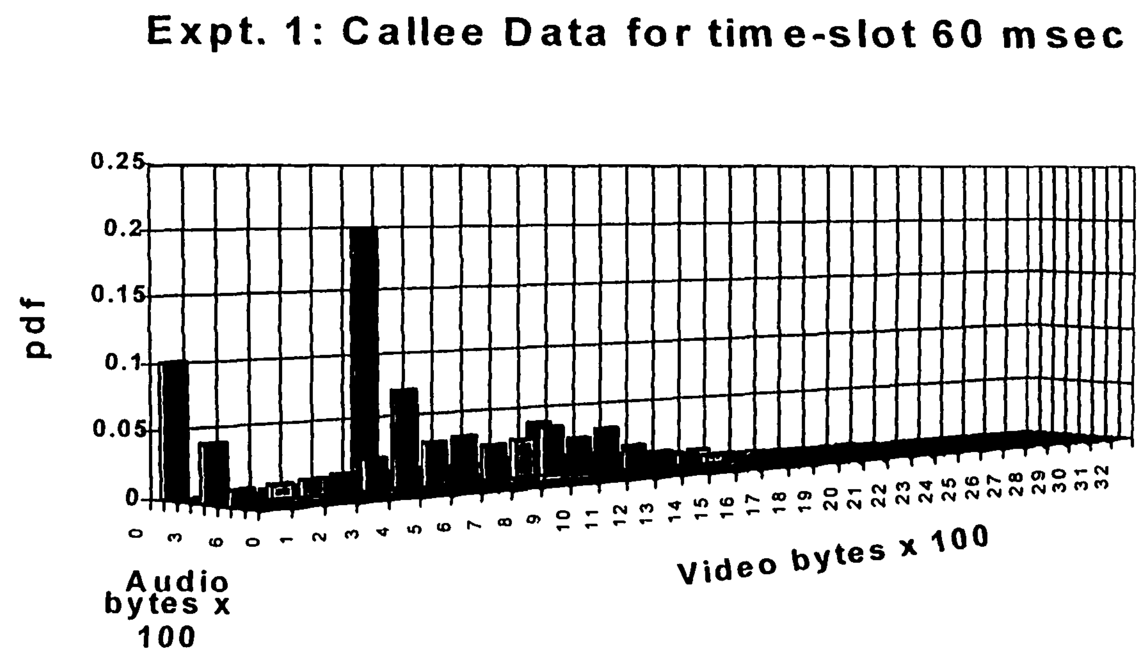

Expt. 1: Callee Data for time-slot 60 msec
pdf
0.25
0.2
0.15
0.1
0.05
0
Audio bytes x 100
0 3 6
Video bytes x 100
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31 32

Figure 3.8

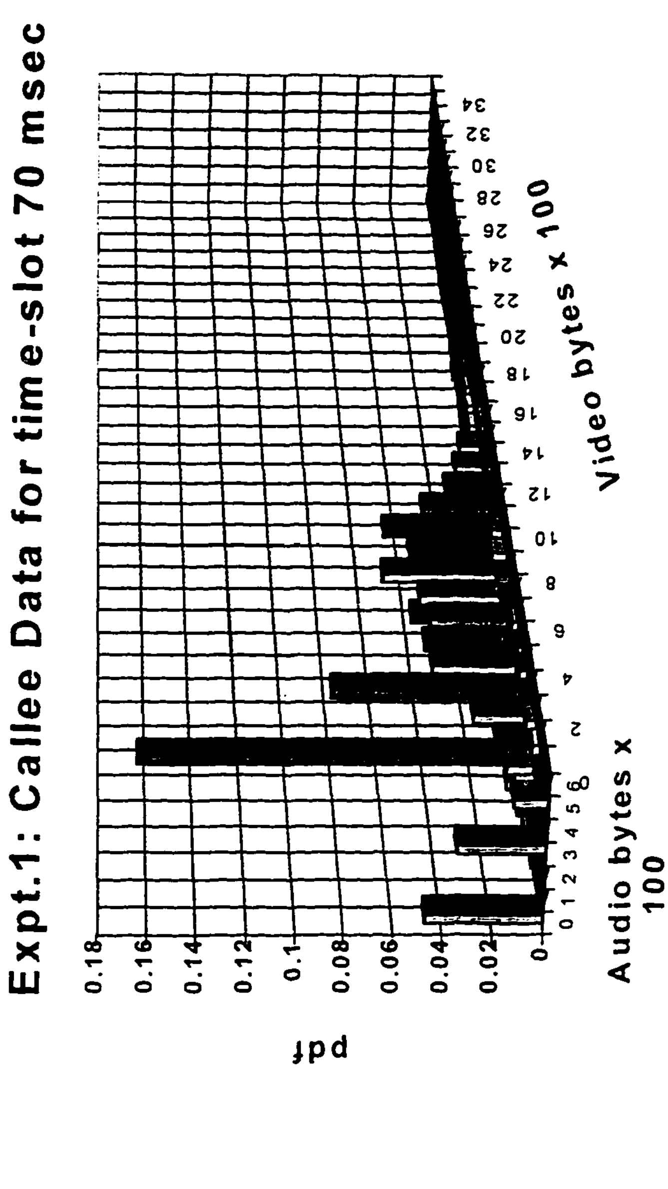

Expt.1: Callee Data for time-slot 70 msec
pdf
0.18
0.16
0.14
0.12
0.1
0.08
0.06
0.04
0.02
0
Audio bytes x 100
0 1 2 3 4 5 6
Video bytes x 100
2 4 6 8 10 12 14 16 18 20 22 24 26 28 30 32 34

Figure 3.9

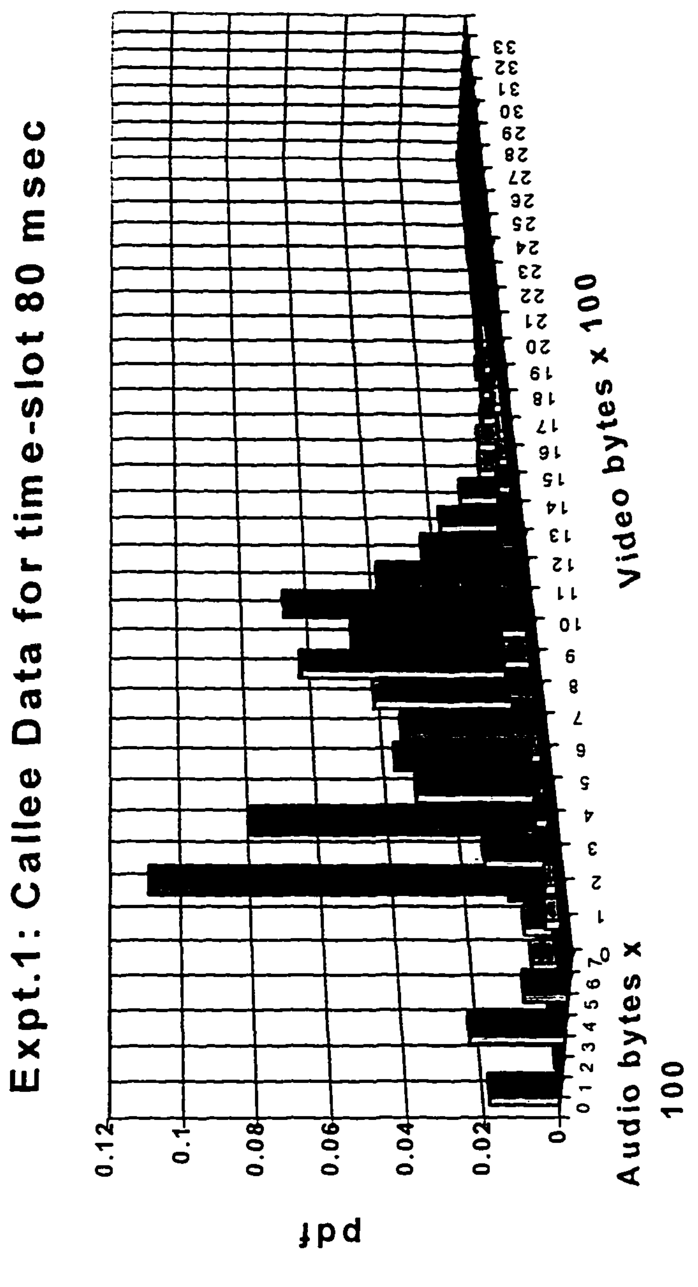

Expt.1: Callee Data for time-slot 80 msec
pdf
0.12
0.1
0.08
0.06
0.04
0.02
0
Audio bytes x 100
Video bytes x 100

Figure 3.10

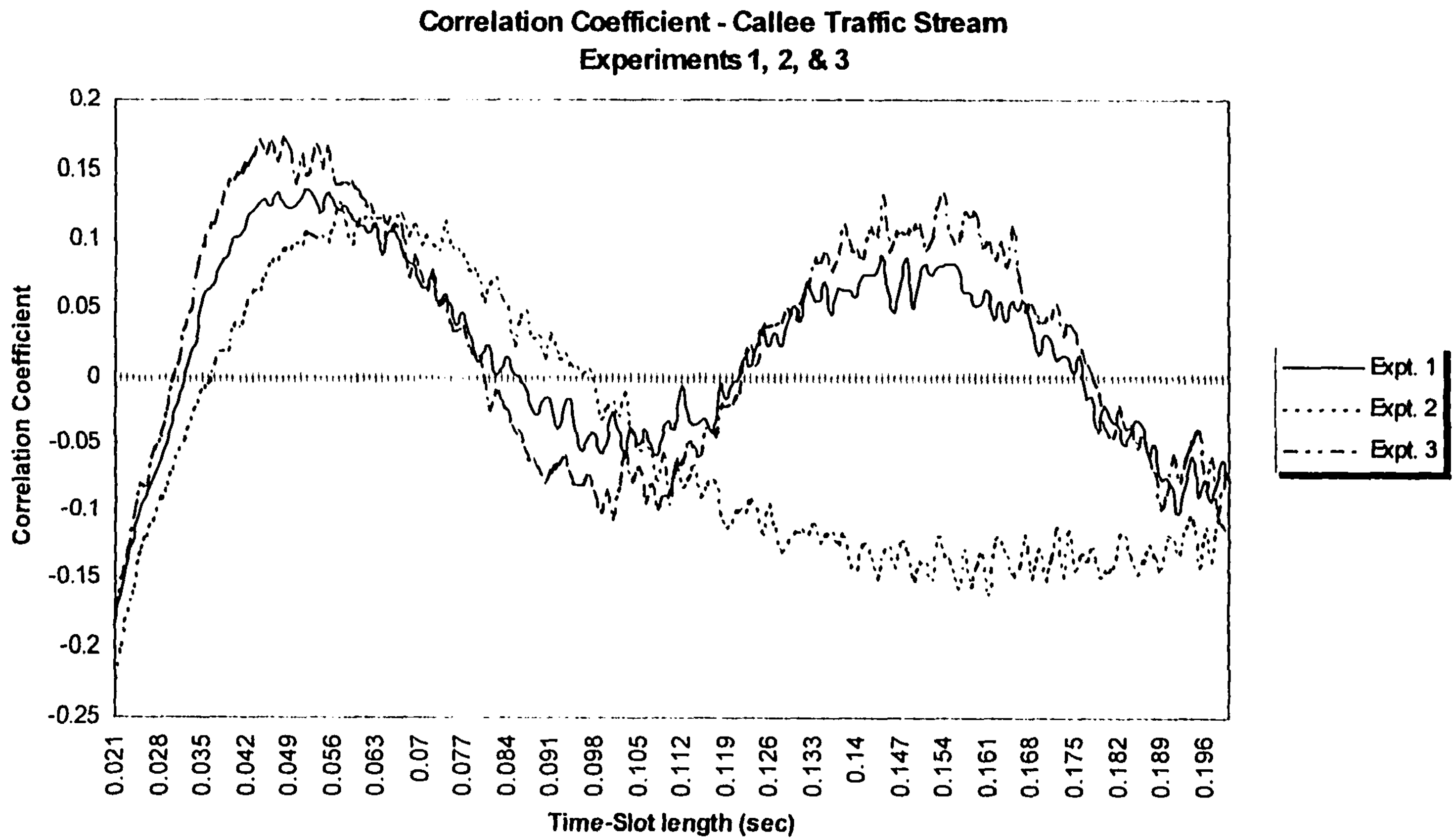

Correlation Coefficient - Callee Traffic Stream
Experiments 1, 2, & 3
Correlation Coefficient
0.2
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
-0.2
-0.25
0.021
0.028
0.035
0.042
0.049
0.056
0.063
0.07
0.077
0.084
0.091
0.098
0.105
0.112
0.119
0.126
0.133
0.14
0.147
0.154
0.161
0.168
0.175
0.182
0.189
0.196
Time-Slot length (sec)
Expt. 1
Expt. 2
Expt. 3

Figure 3.11

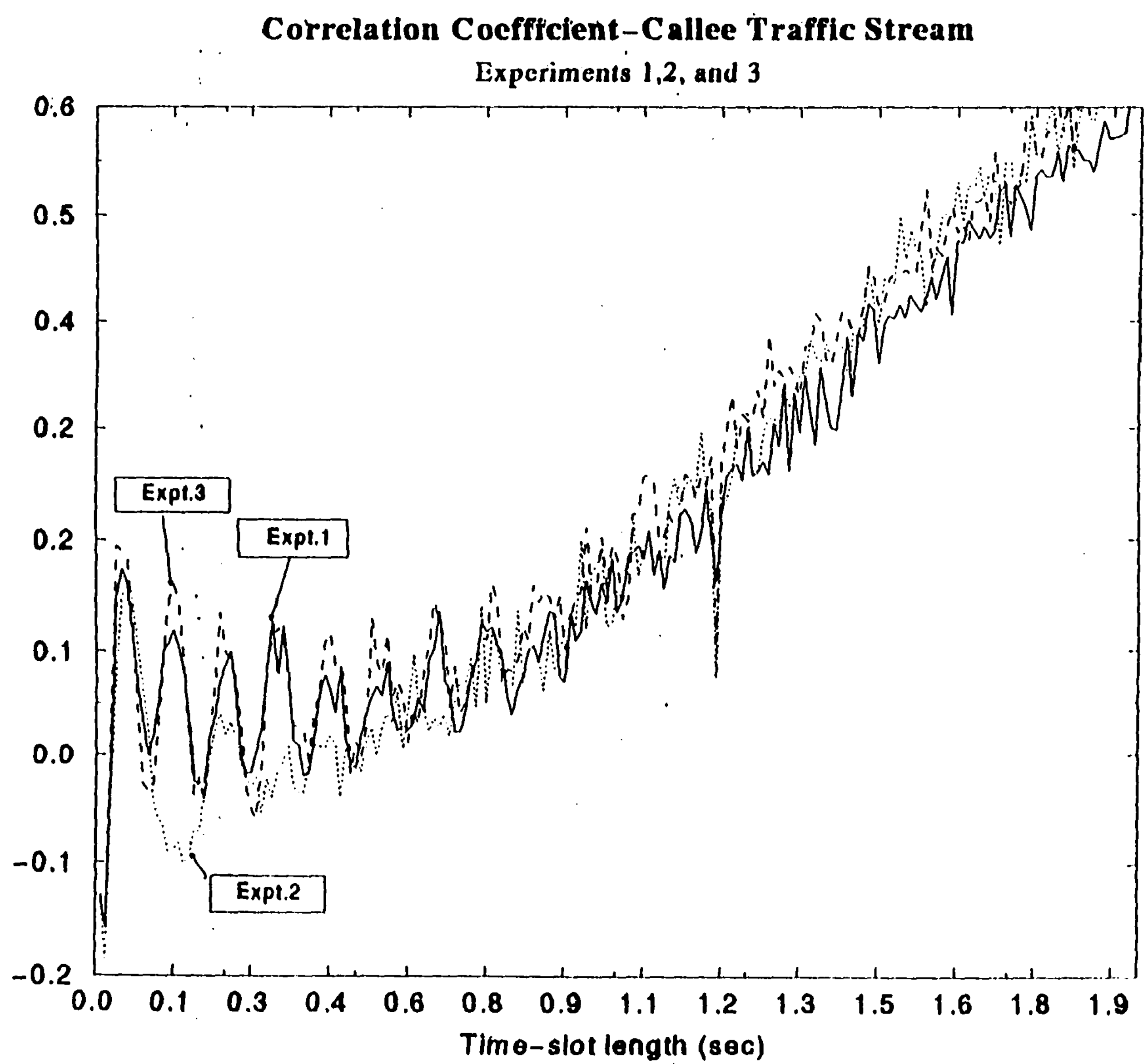

Correlation Coefficient-Callee Traffic Stream
Experiments 1,2, and 3
0.6
0.5
0.4
0.2
0.2
0.1
0.0
-0.1
-0.2
0.0 0.1 0.3 0.4 0.6 0.8 0.9 1.1 1.2 1.3 1.5 1.6 1.8 1.9
Time-slot length (sec)
Expt.3
Expt.1
Expt.2

Figure 3.12

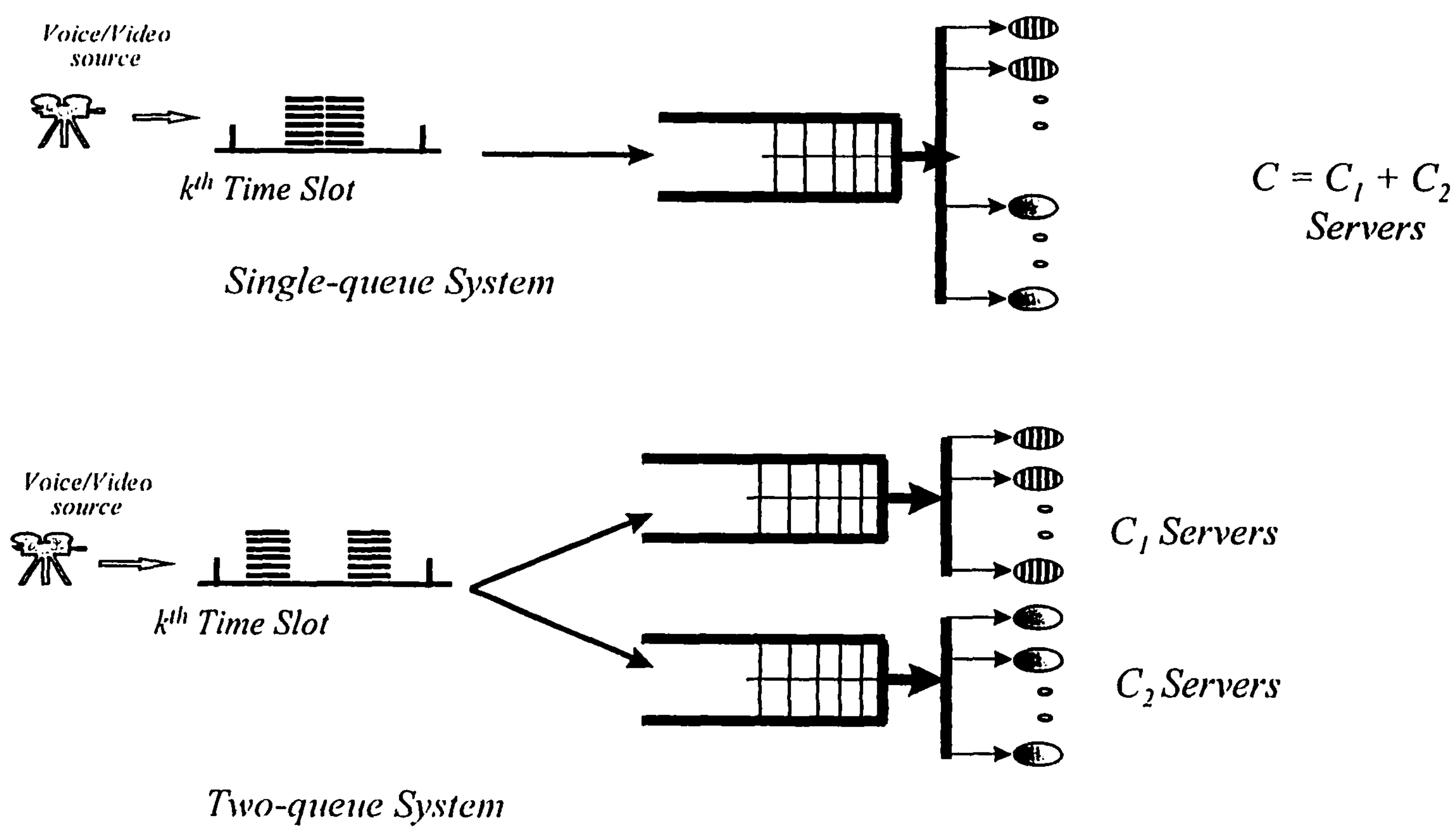

Voice/Video source
k^{th} Time Slot
Single-queue System
$C = C_1 + C_2$ Servers
Voice/Video source
k^{th} Time Slot
Two-queue System
C_1 Servers
C_2 Servers

Figure 4.1

Union of zeros

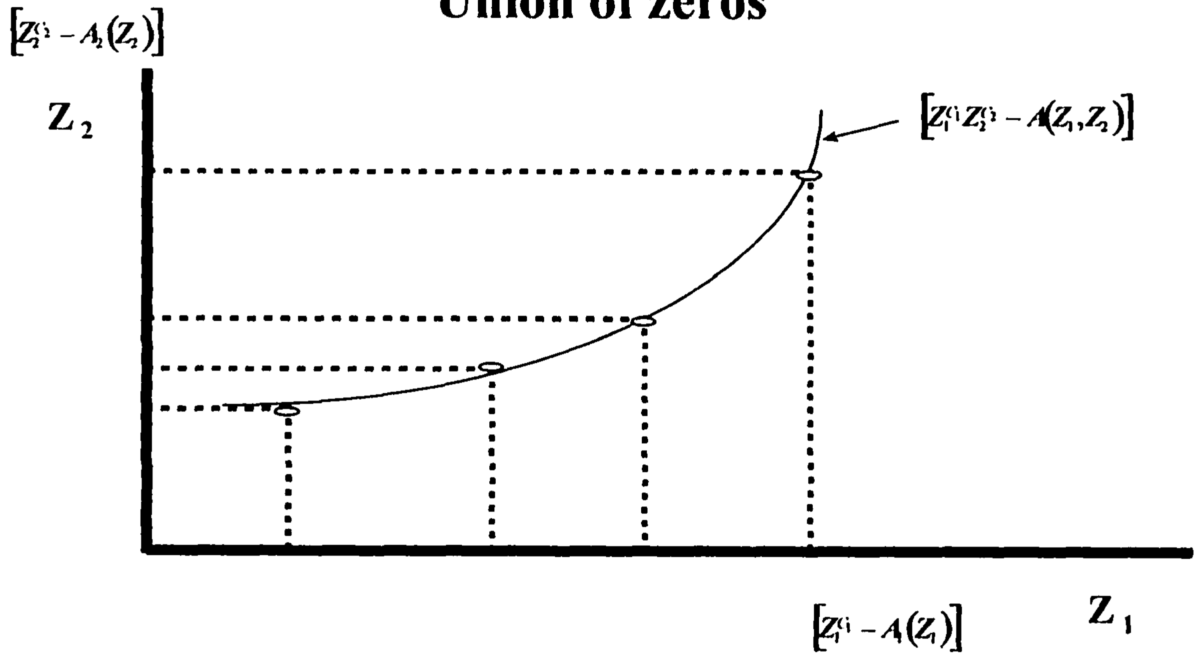


Figure 4.2

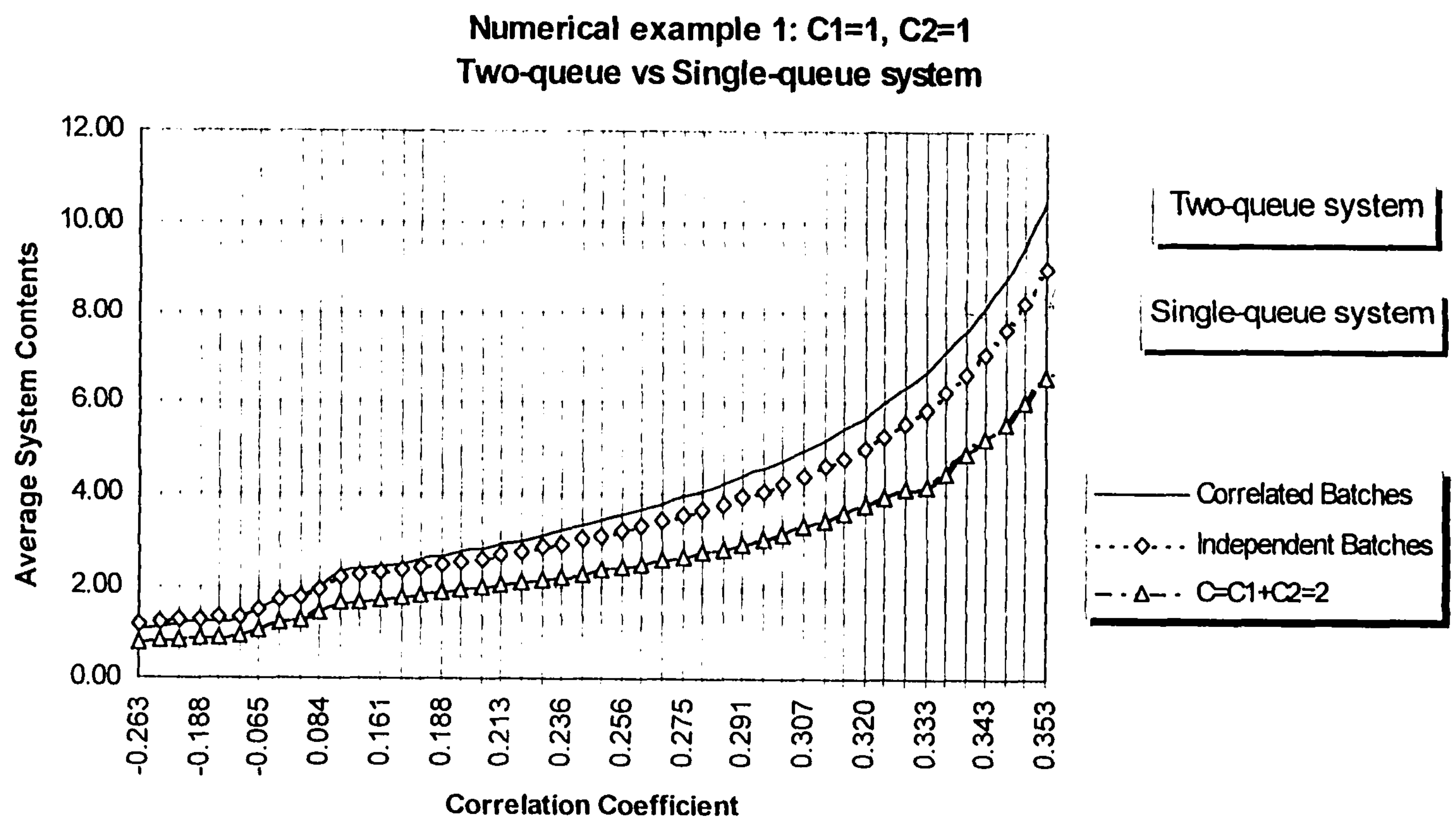
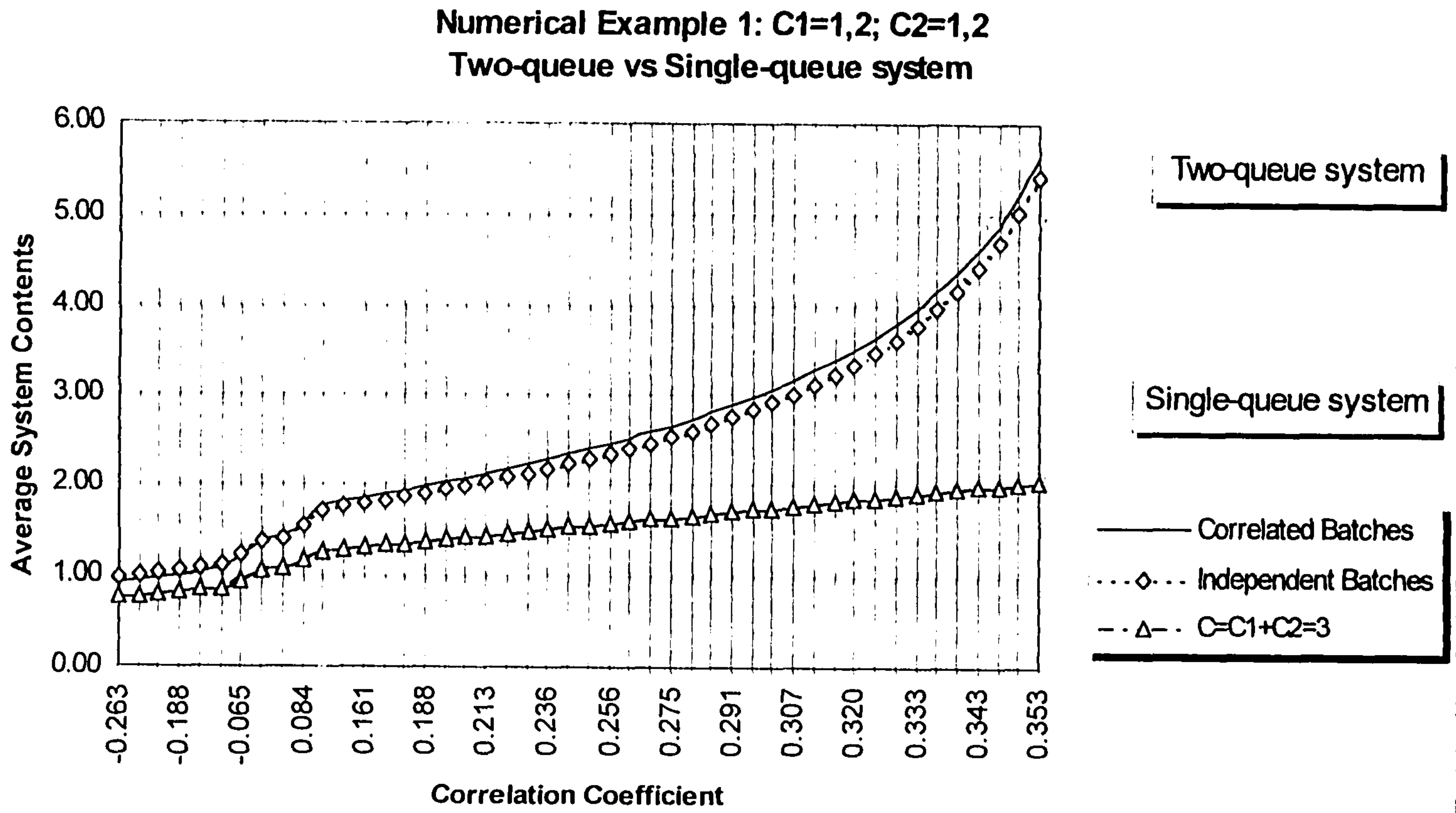

Numerical example 1: C1=1, C2=1
Two-queue vs Single-queue system
Average System Contents
12.00
10.00
8.00
6.00
4.00
2.00
0.00
-0.263
-0.188
-0.065
0.084
0.161
0.188
0.213
0.236
0.256
0.275
0.291
0.307
0.320
0.333
0.343
0.353
Correlation Coefficient
Two-queue system
Single-queue system
Correlated Batches
Independent Batches
C=C1+C2=2

Figure 5.1



Numerical Example 1: C1=1,2; C2=1,2
Two-queue vs Single-queue system
Average System Contents
6.00
5.00
4.00
3.00
2.00
1.00
0.00
-0.263
-0.188
-0.065
0.084
0.161
0.188
0.213
0.236
0.256
0.275
0.291
0.307
0.320
0.333
0.343
0.353
Correlation Coefficient
Two-queue system
Single-queue system
Correlated Batches
Independent Batches
C=C1+C2=3

Figure 5.2

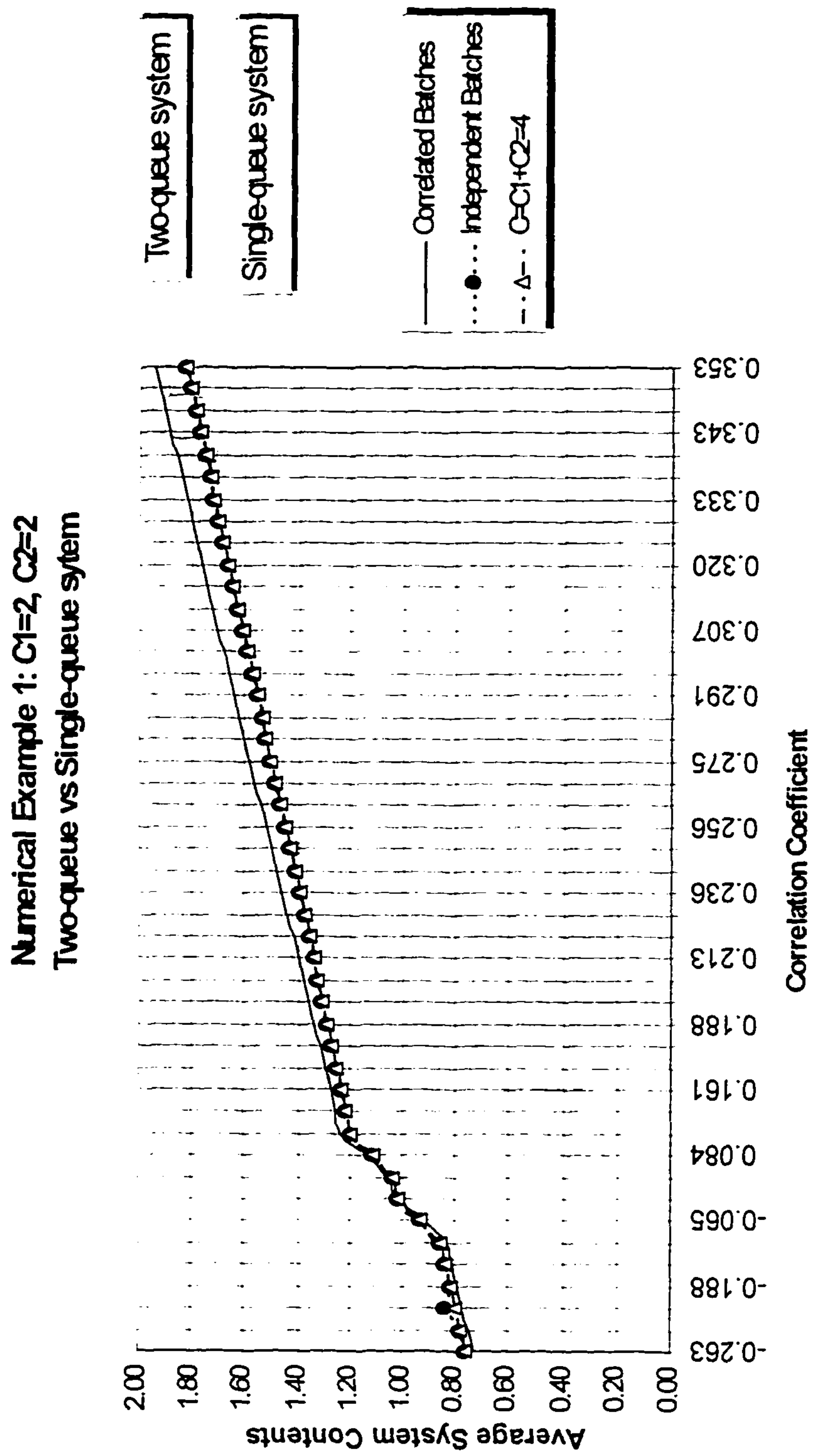

Numerical Example 1: C1=2, C2=2
Two-queue vs Single-queue sytem
Two-queue system
Single-queue system
Correlated Batches
Independent Batches
C=C1+C2=4
Average System Contents
2.00
1.80
1.60
1.40
1.20
1.00
0.80
0.60
0.40
0.20
0.00
-0.263
-0.188
-0.065
0.084
0.161
0.188
0.213
0.236
0.256
0.275
0.291
0.307
0.320
0.333
0.343
0.353
Correlation Coefficient

Figure 5.3

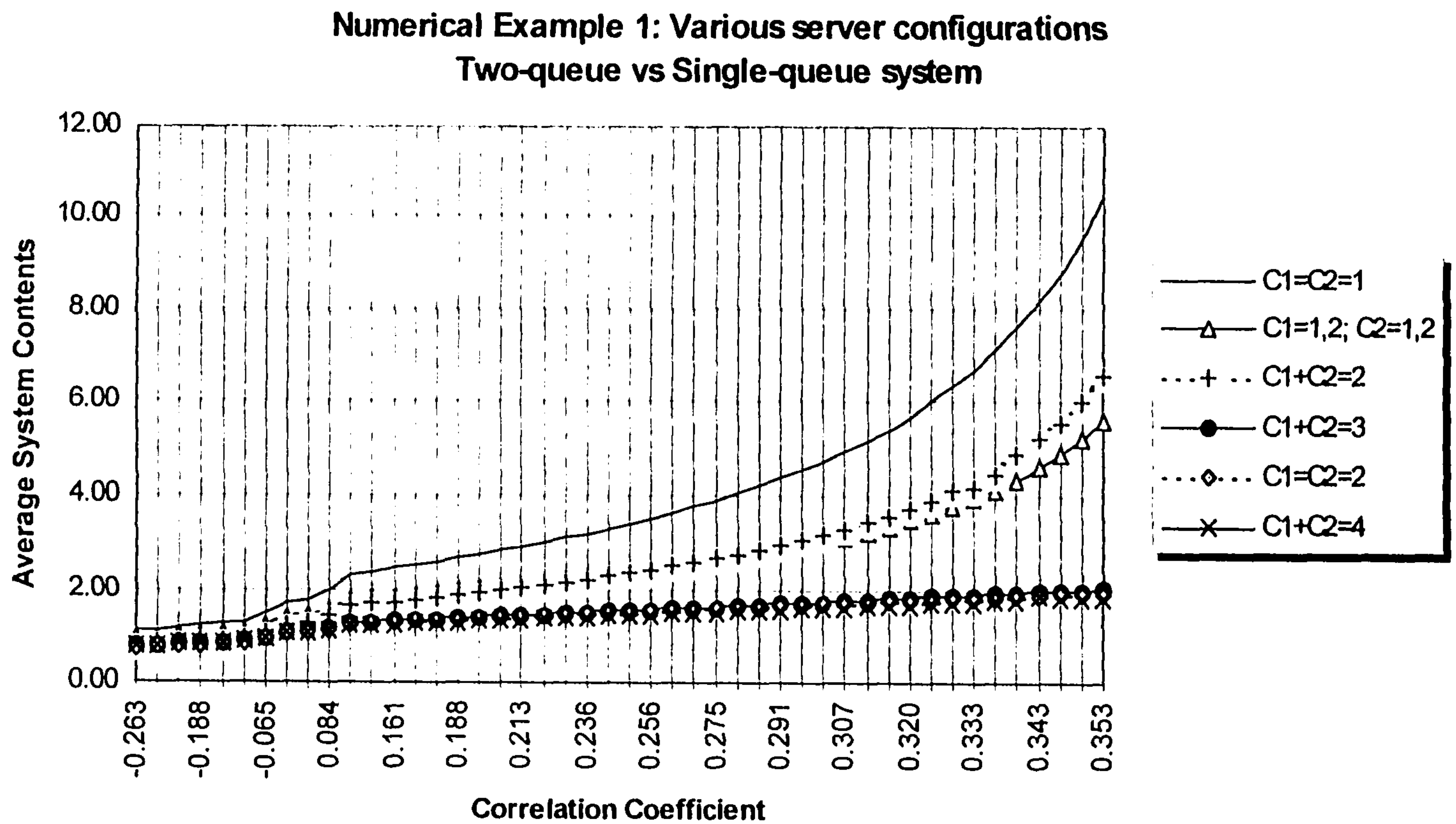

Numerical Example 1: Various server configurations
Two-queue vs Single-queue system
Average System Contents
12.00
10.00
8.00
6.00
4.00
2.00
0.00
-0.263
-0.188
-0.065
0.084
0.161
0.188
0.213
0.236
0.256
0.275
0.291
0.307
0.320
0.333
0.343
0.353
Correlation Coefficient
C1=C2=1
C1=1,2; C2=1,2
C1+C2=2
C1+C2=3
C1=C2=2
C1+C2=4

Figure 5.4

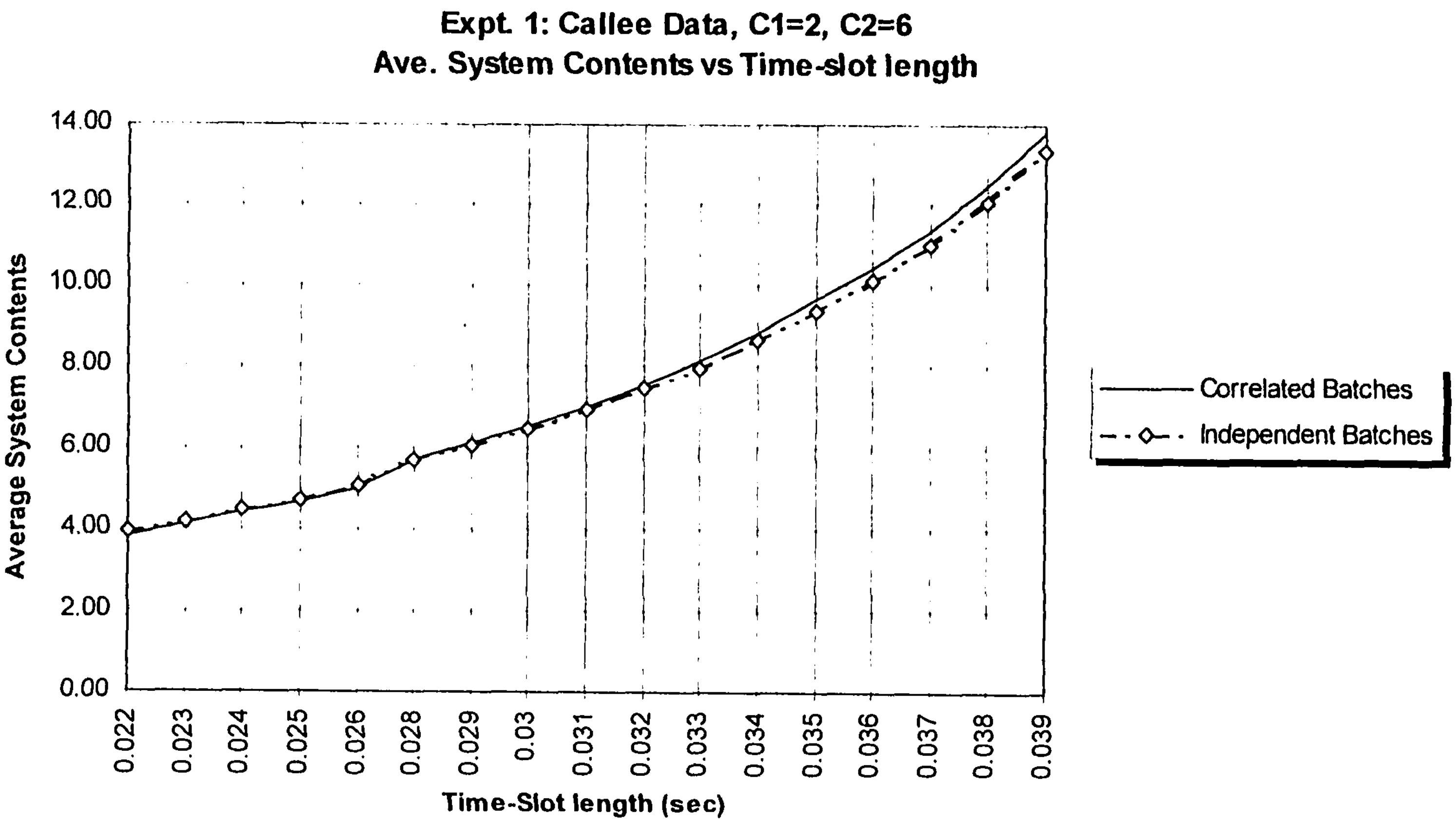

Expt. 1: Callee Data, C1=2, C2=6
Ave. System Contents vs Time-slot length
Average System Contents
14.00
12.00
10.00
8.00
6.00
4.00
2.00
0.00
0.022
0.023
0.024
0.025
0.026
0.028
0.029
0.03
0.031
0.032
0.033
0.034
0.035
0.036
0.037
0.038
0.039
Time-Slot length (sec)
Correlated Batches
Independent Batches

Figure 5.5

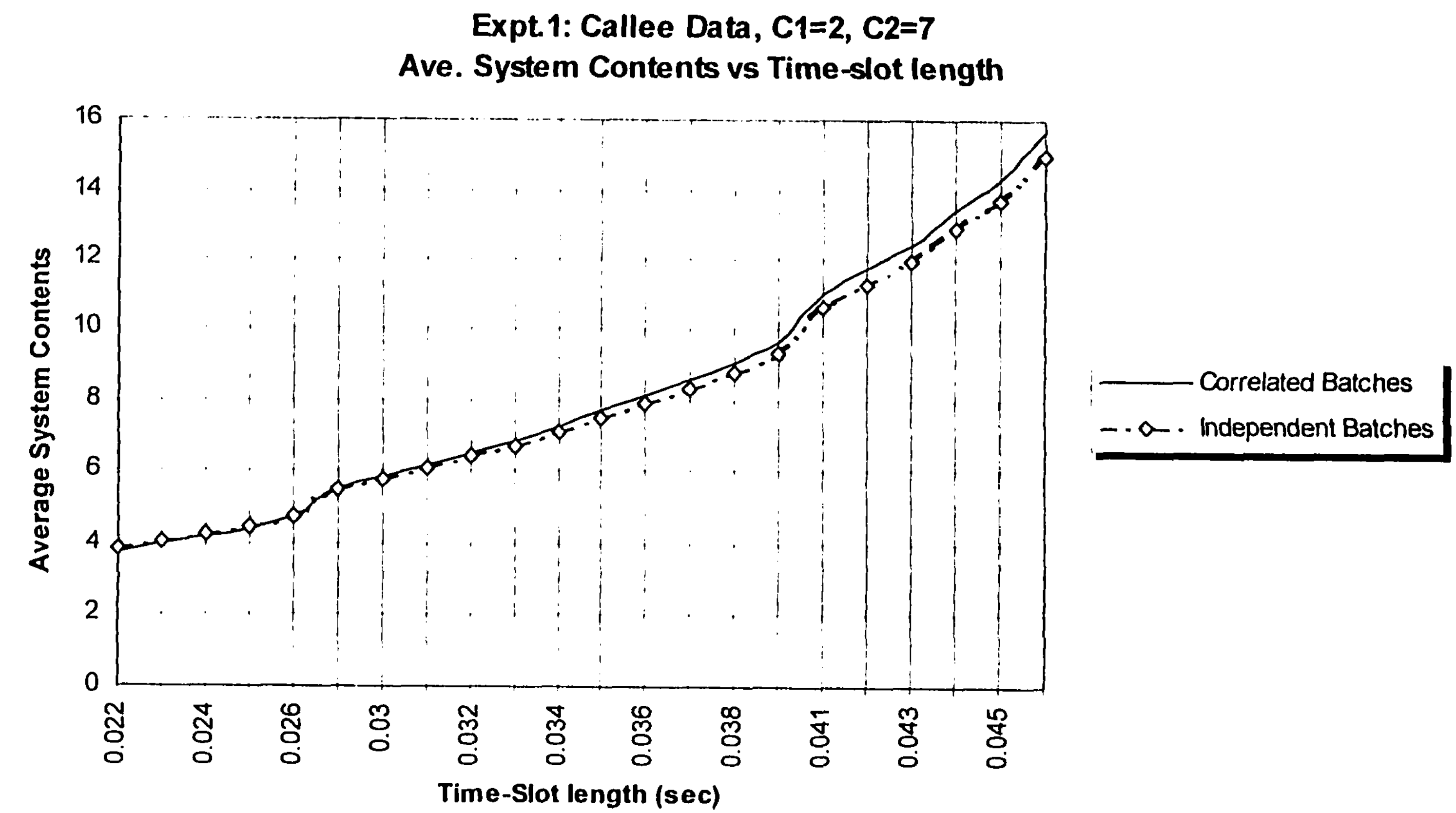

Expt.1: Callee Data, C1=2, C2=7
Ave. System Contents vs Time-slot length
Average System Contents
0
2
4
6
8
10
12
14
16
0.022
0.024
0.026
0.03
0.032
0.034
0.036
0.038
0.041
0.043
0.045
Time-Slot length (sec)
Correlated Batches
Independent Batches

Figure 5.6

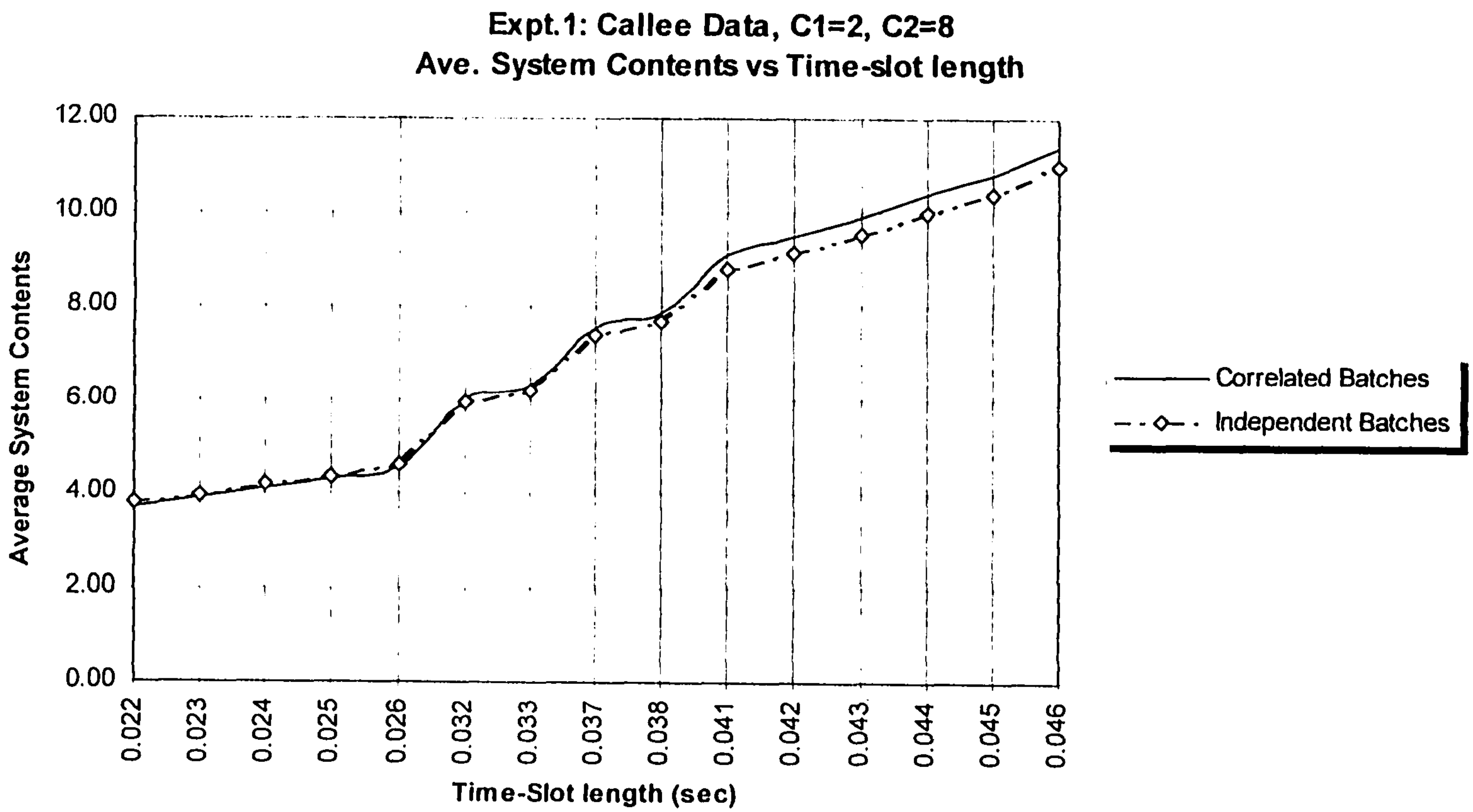

Expt.1: Callee Data, C1=2, C2=8
Ave. System Contents vs Time-slot length
12.00
10.00
8.00
6.00
4.00
2.00
0.00
Average System Contents
0.022
0.023
0.024
0.025
0.026
0.032
0.033
0.037
0.038
0.041
0.042
0.043
0.044
0.045
0.046
Time-Slot length (sec)
Correlated Batches
Independent Batches

Figure 5.7

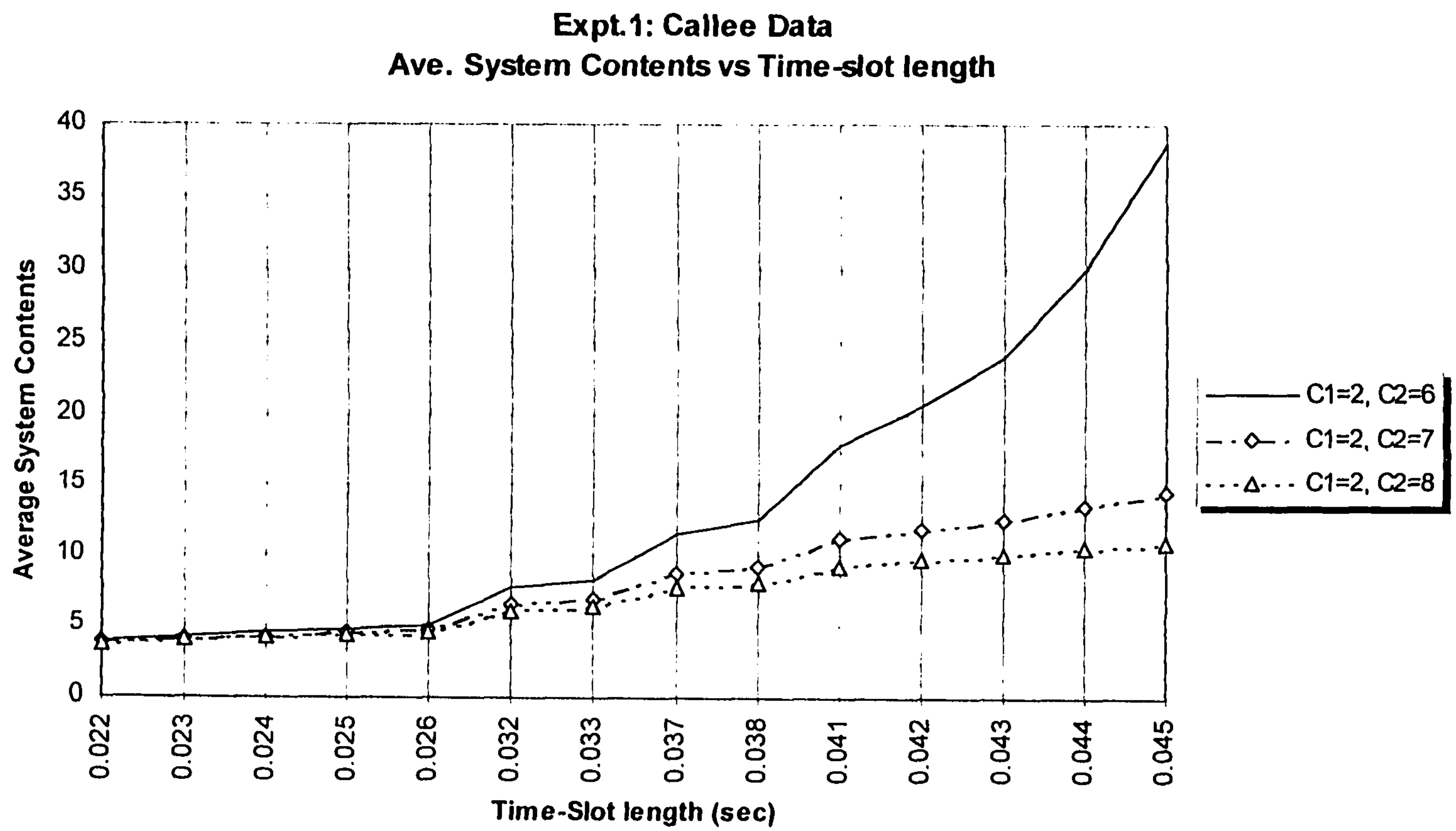

Expt.1: Callee Data
Ave. System Contents vs Time-slot length
Average System Contents
0
5
10
15
20
25
30
35
40
0.022
0.023
0.024
0.025
0.026
0.032
0.033
0.037
0.038
0.041
0.042
0.043
0.044
0.045
Time-Slot length (sec)
C1=2, C2=6
C1=2, C2=7
C1=2, C2=8

Figure 5.8

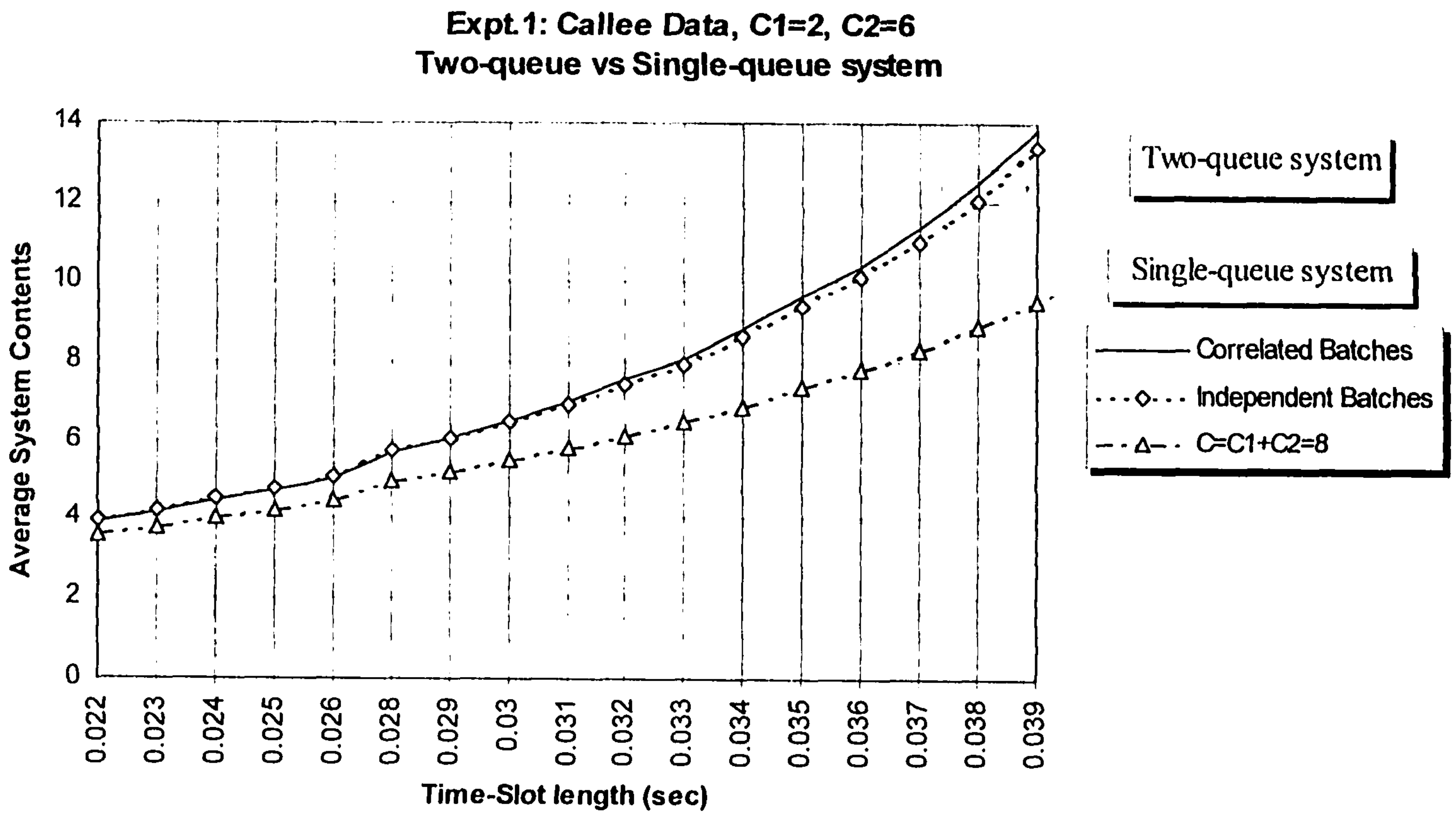

Expt.1: Callee Data, C1=2, C2=6
Two-queue vs Single-queue system
Average System Contents
0
2
4
6
8
10
12
14
0.022
0.023
0.024
0.025
0.026
0.028
0.029
0.03
0.031
0.032
0.033
0.034
0.035
0.036
0.037
0.038
0.039
Time-Slot length (sec)
Two-queue system
Single-queue system
Correlated Batches
Independent Batches
C=C1+C2=8

Figure 5.9

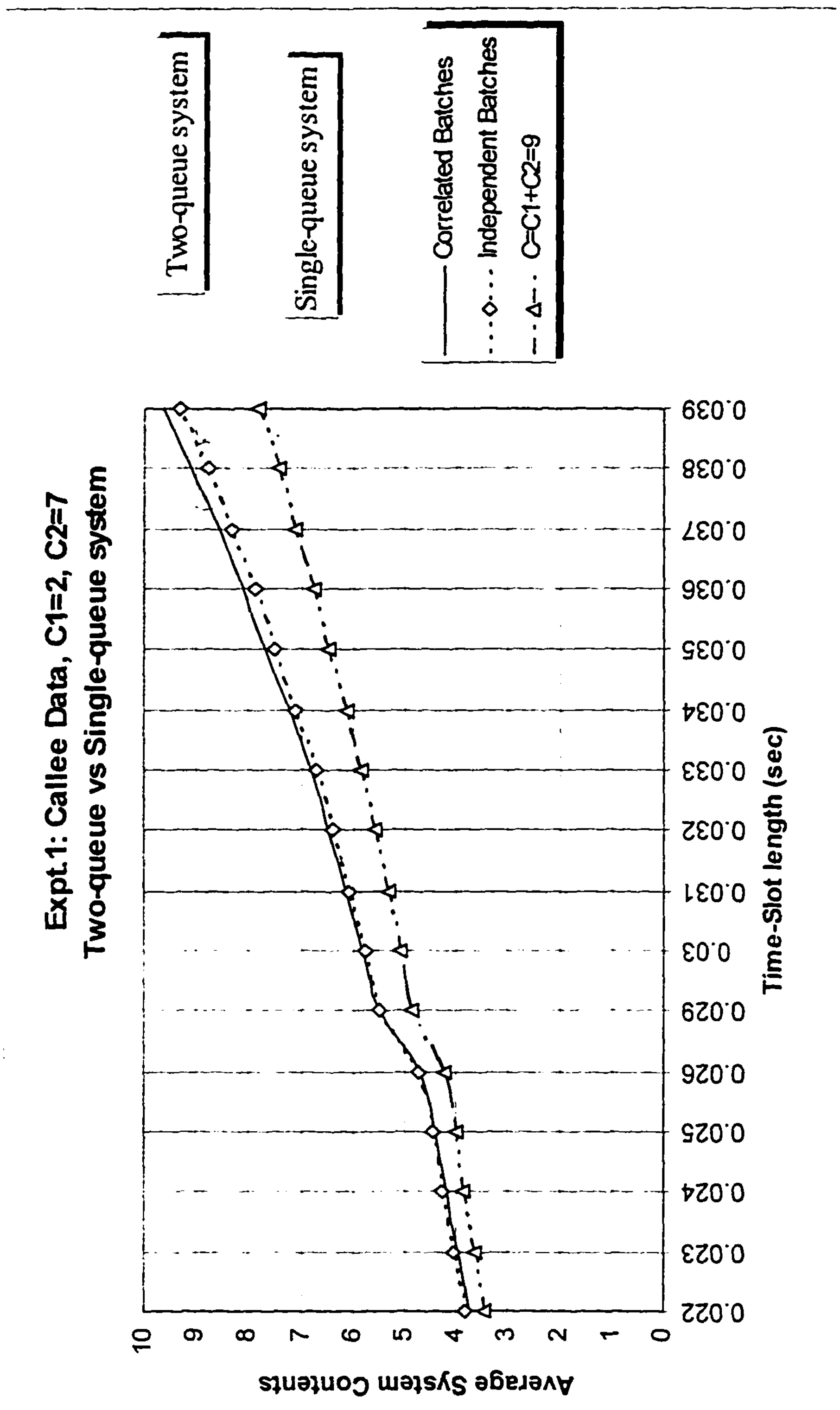

Expt.1: Callee Data, C1=2, C2=7
Two-queue vs Single-queue system
Two-queue system
Single-queue system
Correlated Batches
Independent Batches
C=C1+C2=9
Average System Contents
Time-Slot length (sec)
0
1
2
3
4
5
6
7
8
9
10
0.022
0.023
0.024
0.025
0.026
0.029
0.03
0.031
0.032
0.033
0.034
0.035
0.036
0.037
0.038
0.039

Figure 5.10

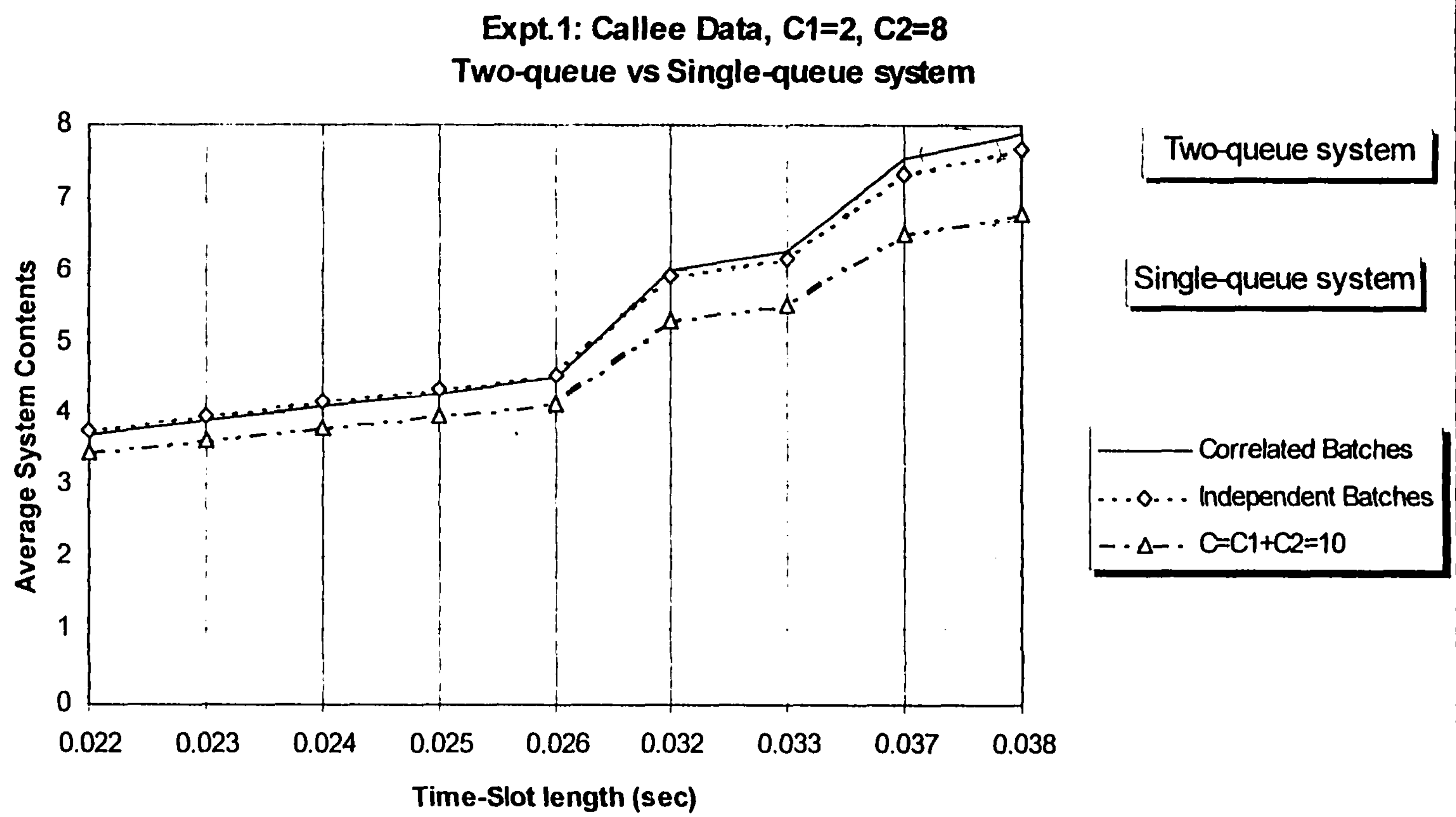

Expt.1: Callee Data, C1=2, C2=8
Two-queue vs Single-queue system
Two-queue system
Single-queue system
Average System Contents
0
1
2
3
4
5
6
7
8
0.022
0.023
0.024
0.025
0.026
0.032
0.033
0.037
0.038
Time-Slot length (sec)
Correlated Batches
Independent Batches
C=C1+C2=10

Figure 5.11

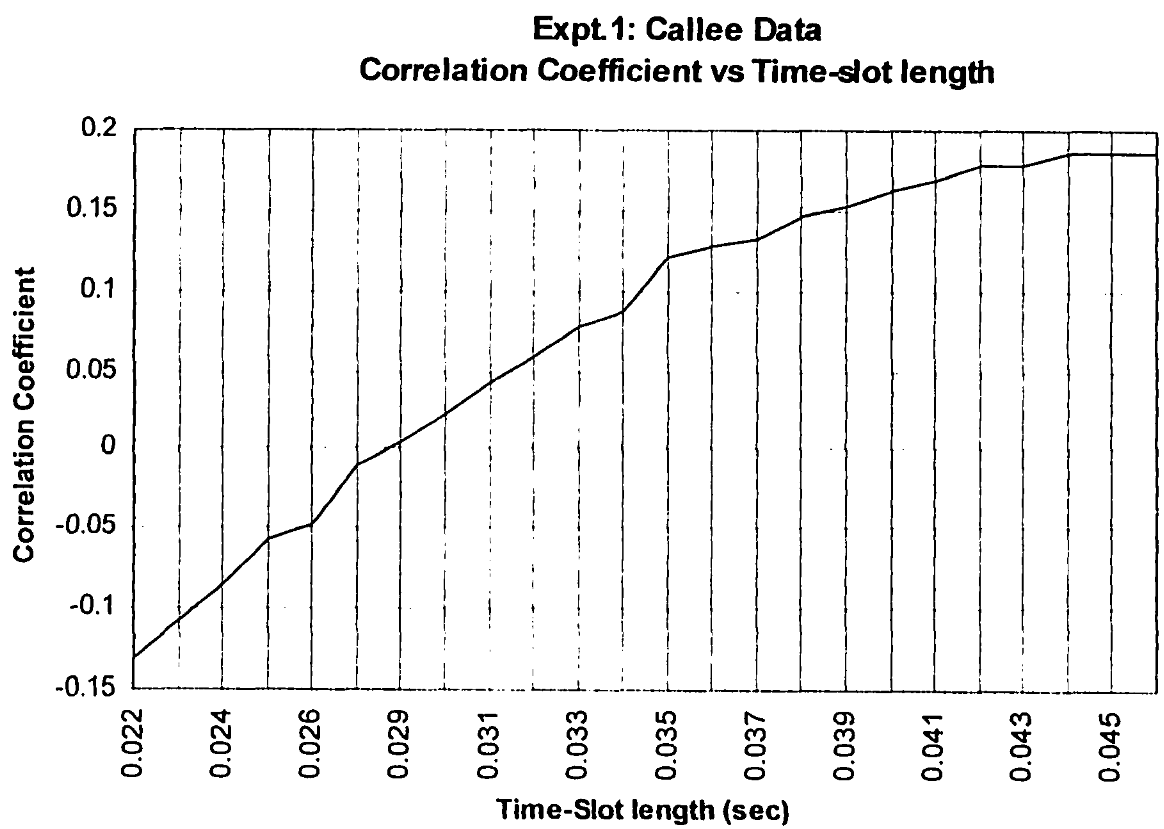

Expt.1: Callee Data
Correlation Coefficient vs Time-slot length
Correlation Coefficient
0.2
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
0.022
0.024
0.026
0.029
0.031
0.033
0.035
0.037
0.039
0.041
0.043
0.045
Time-Slot length (sec)

Figure 5.12

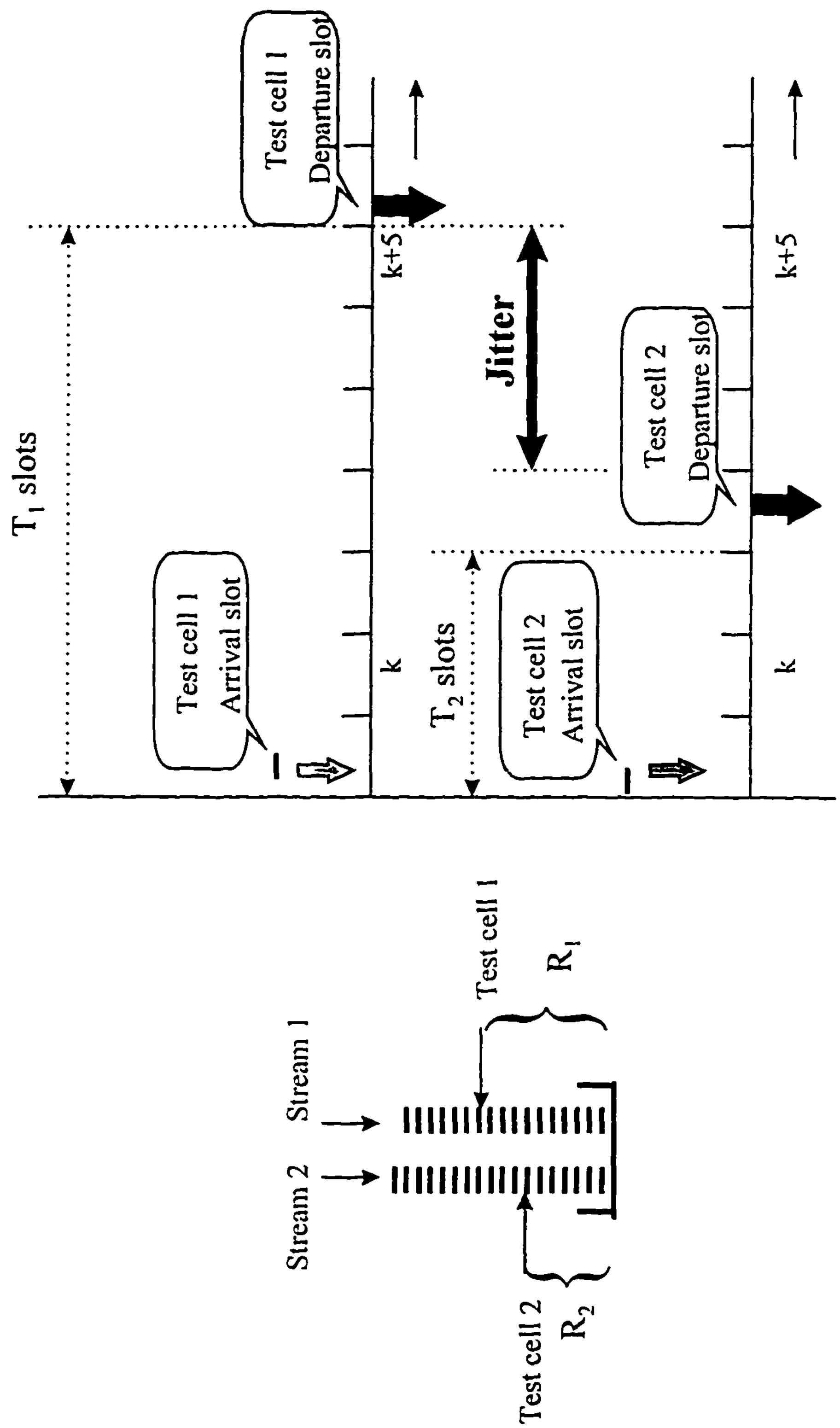

T1 slots
Test cell 1 Arrival slot
Test cell 1 Departure slot
k
k+5
Jitter
T2 slots
Test cell 2 Arrival slot
Test cell 2 Departure slot
k
k+5
Stream 1
Stream 2
Test cell 1
R1
Test cell 2
R2

Figure 6.1

Table 2.1: Optical Transmission Standards

SONET ELECTRIC	SONET OPTICAL	CCITT (SDH)	BW [MBPS]
STS-1	OC-1		51.84
STS-3	OC-3	STM-1	155.52
STS-9	OC-9	STM-3	466.56
STS-12	OC-12	STM-4	622.08
STS-18	OC-18	STM-6	933.12
STS-24	OC-24	STM-8	1244.16
STS-48	OC-48	STM-16	2488.32
STS-192	OC-192		~9600

Table 2.2: Electrical transmission Standards

North America	Europe	BW [MBPS]
DS1=T1		1.544
	E1	2.048
DS2		6.312
	E2	8.448
	E3	34.368
DS3		44.736
DS4		?

DS0 = 64-kbps

Table (3.1) - Number of Control, Audio, and Video Packets

Expt. No	Duration (Mn.)	Total no. of pkts, Control & Info	Workstation	Video Packets, No. of bytes	Audio Packets, No. of bytes
1	30	Info.Pkts-141,702	Caller	Pkts. 48577 Bytes 23171276	Pkts. 19705 Bytes 4610970
		Ctrl.Pkts - 2458	Callee	Pkts. 51257 Bytes 23257484	Pkts. 19705 Bytes 4610970
2	30	Info.Pkts-141,702	Caller	Pkts. 37507 Bytes 18692866	Pkts. 15923 Bytes 3725982
		Ctrl.Pkts - 2458	Callee	Pkts. 39972 Bytes 18733120	Pkts. 15922 Bytes 3725748
3	30	Info.Pkts-141,702	Caller	Pkts. 45112 Bytes 21464372	Pkts. 18238 Bytes 4267692
		Ctrl.Pkts - 2458	Callee	Pkts. 47942 Bytes 21586248	Pkts. 18238 Bytes 4267692

Table 5.1

p	E[z1]	E[z2]	Var[z1]	Var[z2]	Corr[z1,z2]	S'[1]corr c1=c2=1	S'[1]ind c1=c2=1	S'[1]corr c1=1,c2=2 c1=2,c2=1	S'[1]ind c1=1,c2=2 c1=2,c2=1	S'[1]corr c1=c2=2	S'[1]ind c1=c2=2
0.065	0.915	0.915	0.6877	0.6877	0.3538	10.4346	9.0064	5.6257	5.4182	1.94188	1.83
0.07	0.87	0.87	0.6931	0.6931	0.3363	7.098	6.2015	4.1557	3.9707	1.8431	1.74
0.075	0.825	0.825	0.6943	0.6943	0.3163	5.4196	4.7928	3.3839	3.2214	1.7433	1.65
0.08	0.78	0.78	0.6916	0.6916	0.2914	4.3818	3.9236	2.8818	2.7418	1.6426	1.56
0.085	0.735	0.735	0.6847	0.6847	0.2625	3.6582	3.319	2.512	2.3945	1.541	1.47
0.09	0.69	0.69	0.6739	0.6739	0.2283	3.1121	2.8638	2.2169	2.1219	1.4387	1.38
0.95	0.645	0.645	0.6589	0.6589	0.1881	2.6758	2.5012	1.9681	1.8956	1.3357	1.29
0.1	0.6	0.6	0.64	0.64	0.1406	2.3125	2.2	1.75	1.7	1.2321	1.2
0.105	0.555	0.555	0.6169	0.6169	0.0842	1.9998	1.9414	1.5532	1.5257	1.1279	1.11
0.11	0.51	0.51	0.5899	0.5899	0.01678	1.7239	1.7138	1.3719	1.3669	1.0233	1.02
0.115	0.465	0.465	0.5587	0.5587	-0.0648	1.4755	1.5094	1.2022	1.2197	0.9182	0.93
0.12	0.42	0.42	0.5236	0.5236	-0.16501	1.2482	1.3227	1.0413	1.0813	0.8126	0.84
0.121	0.411	0.411	0.51607	0.51607	-0.1878	1.2049	1.3227	1.0101	1.0546	0.7915	0.822
0.122	0.402	0.402	0.5083	0.5083	-0.2366	1.12	1.2176	0.979	1.0281	0.7703	0.804
0.123	0.393	0.393	0.5005	0.5005	-0.2366	1.12	1.2176	0.9483	1.0018	0.74914	0.786
0.124	0.384	0.384	0.4925	0.4925	-0.2628	1.0785	1.1835	0.9177	0.9757	0.7279	0.768

Bibliography

[1] Gerold Blakowski, Ralf Steinmetz, "A Media Synchronization Survey: Reference Model, Specification, and Case Studies, " *IEEE J. Select. Areas Commun.*, Vol. 14, No. 1, January 1996.

[2] Sreenan, Cormac John, "Synchronization Services for Digital Continuous Media," *Ph.D. Dissertation*, Christ's College, University of Cambridge, October 1992.

[3] Ralf Steinmetz, "Human Perception of Jitter and Media Synchronization," *IEEE J. Select Areas Commun.*, Vol. 14, No. 1, January 1996.

[4] H. Zhang and S. Keshav, "Comparison of rate-based service disciplines," in *Proc. ACM SIGCOMM'91*, Zürich, Switzerland, Sept. 1991.

[5] Ian F. Akyildiz, Wei Yen, "Multi-media Group Synchronization Protocols for Integrated Services Networks," *IEEE J. Select Areas Commun.*, Vol. 14, No. 1, January 1996.

[6] "ProShare Personal Conferencing Video System, User's Guide. Part 624276-002," *Intel Corporation*, 5200 N.E. Elam Young Park way, Hillsboro, Oregon 97124-6497.

[7] CCITT Study Group XVIII Draft Recommendation I.121. *Broadband Aspects of ISDN*, February 1988. (pp 2, 35)

[8] J. L. Hui and E. Arthurs, "A broadband packet switch for integrated transport." IEEE J. Select. Areas Commun., vol. SAC-5, no. 8, pp. 1264-1273, 1987.

[9] I. Imagawa, S. Urushidani, and K. Hagishima, "A new self-routing switch driven with input-output address difference," in Proc. GLOBECOM'88, 1988, pp.1607-1611.

[10] P. E. White, J. Y. Hui, M. Decina, and R. Tatsuboshi, "Switching for broadband communications," IEEE J. Select. Areas Commun., vol. SAC-5, no. 8, pp. 1217-1221, 1987.

[11] J. Filipiak, "Structured systems analysis methodology for design of an ATM network architecture," IEEE J. Select. Areas Commun., vol. 7, no. 8, pp. 1263-1273, 1989.

[12] J. P. Vorstermans and A. P. De Vleeschouwer, "Layered ATM systems and architectural concepts for subscribers' premises networks," IEEE J. Select. Areas Commun., vol. 6, no. 9, pp. 1545-1555, 1988.

[13] H. Ohnishi, T. Okada, and K. Noguchi, "Flow control schemes and delay/loss tradeoff in ATM networks," IEEE J. Select. Areas Commun., vol. 6, no. 9, pp. 1609-1616, 1988.

[14] C. Blondia and T. Theimer, "A Discrete-Time Model for ATM Traffic," *RACE Document, PRLB_123_0015_CC_CD*, August 1989.

[15] C. Blondia, "A Discrete-Time Batch Markovian Arrival Process as B-ISDN Traffic Model," *Submitted for publication* 1991.

[16] H. Bruneel, "On Buffers with Periodical Input Traffic," Europ. J. Op. Re., 21, 379-386

[17] H. Heffes and D. M. Lucantoni, "A Markov Modulated Characterization of Packetized Voice and Data Traffic and Related Statistical Multiplexer Performance," *IEEE J. Select. Areas in Commun.*, SAC-4, No. 6, Sept., 1986.

[18] D. M. Lucantoni, "New Results on the Single Server Queue with a Batch Markovian Arrival Process," *Commun. in Statistics - Stochastic Models.* 1990.

[19] D. Anick, D. Mitra and M. M. Sondhi, "Stochastic Theory of a Data-Handling System with Multiple Sources," B.S.T.J., 61, No. 8, pp. 1871-1894,Oct., 1982

[20] F. Hubner, "Analysis of a Finite-Capacity Asynchronous Multiplexer with deterministic Traffic Sources," 7th ITC Seminar, Morristown , NJ, Oct. 1990.

[21] H. Kroner, "Statistical Multiplexing of Sporadic Sources - Exact and Approximate performance Analysis," Proc. 13th ITC, Copenhagen, June 1991.

[22] B. Maglaris, D. Anastassiou, P. Sen, G. Karlsson, and J. D. Robbins, "Performance Models of Statistical Multiplexing in Packet Video Communications," IEEE Trans. Commun., 36, No. 7, pp. 834-844, July 1988.

[23] R. Nagarajan, J. F. Korose, and D. Towsley, "Approximation Techniques for computing Packet Loss in Finite-Buffered Voice Multiplexers," IEEE J. Select. Areas Commun., vol. 9, no. 3, pp. 368-377, April 1991.

[24] I. Norros, J. W. Roberts, A. Simonian, and J. T. Virtamo, " The Superposition of Variable Bit Rate Sources in an ATM Multiplexer," IEEE J. Select. Areas Commun., vol. 9, no. 3, pp. 378-387, April 1991.

[25] G. Ramamurthy and B. Sengupta, "Delay Analysis of a Packet Voice Multiplexer by the $\sum D_i / D / 1$ Queue," IEEE Trans. Commun., 39, no. 7 pp. 1107-1114, July 1991.

[26] J. W. Roberts and J. T. Virtamo, " The Superposition of Periodic Cell Arrival Streams in an ATM Multiplexer," IEEE Trans. Commun., 39, no. 2, pp. 298-303, Feb. 1991.

[27] J. Y. Hui, "Resource allocation for broadband networks," IEEE J. Select. Areas Commun., vol. 6, no. 9, pp. 1598-1608, 1988.

[28] H. Saito, "New dimensioning concept for ATM networks," Proc. ITC Spec. Sem., 1990 paper 15.3

[29] H. Uose, K. Shioda, and K. Mase, "Fast cell loss rate evaluation methods and their application to ATM network control," Proc. ITC Spec. Sem., 1990 paper 13.3

[30] J. Burgin, "Broadband ISDN resource management," Proc. ITC Special Seminar, Adelaide, Australia, 1989, paper 12.2

[31] J. Bergin, "Routing and resource control in the broadband ISDN," A.T.R., vol. 22, no. 1, pp. 3-19, 1988.

[32] A. Gersht and K. J. Lee, "A congestion control framework for ATM networks," Proc. INFOCOM'89, 1989, pp. 701-710.

[33] R. Hunter, P. Kaijser, and F. Nielsen, "ODA: A document architecture for open systems," Computer Commun., vol. 12, pp. 69-79, Apr. 1989.

[34] Meyer Dwass, "First Steps in Probability," McGraw-Hill Book Company, 1967.

[35] "LANDesk Personal Conferencing Manager, User's Guide. Part 628247-002," *Intel Corporation*, 5200 N.E. Elam Young Park way, Hillsboro, Oregon 97124-6497.

[36] "LANDesk Traffic Analyst for Windows DOS, User's Guide. Part 400589-001 and Part 400592-001," *Intel Corporation*, 5200 N.E. Elam Young Park way, Hillsboro, Oregon 97124-6497.

[37] "LANDesk Protocol Compiler, User's Guide". *Intel Corporation*, 5200 N.E. Elam Young Park way, Hillsboro, Oregon 97124-6497.

[38] "Document 8159," Available on the Internet at: http://www-cs.intel.com in the directory /cgi/vdkw-cgi.exe/xad6e4e7c-953/search/6177774/2HLO.

[39] M. L. Liou, "Overview of the px64 Kbit/s Video Coding Standard," COMMUNICATION of the ACM, Vol. 34, No. 4, pp. 59-63, April 1991.

[40] "Introduction to the Indeo Video Driver, Indeo1.txt," Available on the Internet at: ftp.intel.com in the directory /pub/IAL/multi-media.

[41] "Indeo, The new (Iteration) Codec On the Block," CD-ROM Professional, November/December 1994.

[42] "How to Create Great-Looking Indeo Video for the CD-ROM Publication." Available on the Internet at: http://www.intel.com/pc-supp/multimed/indeo/greatv.html.

[43] H. Bruneel, B. G. Kim, "Discrete-Time Models for Communication Systems Including ATM," *Kluwer Academic Publishers*, 1993.

[44] C. Blondia and O. Casals, "Performance Analysis of Statistical Multiplexing of VBR Sources," *IEEE InfoCom*, 1992, pp. 828-838.

[45] A. La Corte, A. Lombardo, G. Schembra, "Modeling Superposition of ON-OFF Correlated Traffic Sources in Multi-media Applications," *IEEE InfoCom* 1995, pp. 993-1000.

[46] Leonid M. Roytman, M. N. S. Swamy, George Eichman, "BIBO Stability in the Presence of Nonessential Singularities of the Second Kind in 2-D Digital Filters," *IEEE Transactions on Circuits and Systems*, Vol. CAS-34, No. 1, January 1987.

[47] Phillip Griffiths, Joseph Harris, "Principles of Algebraic Geometry," Chap 5, John Wiley & Sons, 1978.

[48] I. A. Aïzenberg, A. P. Yuzhakov, "Integral Representations and Residues in Multidimensional Complex Analysis," Translations of mathematical Monographs, American Mathematical Society, Volume 58,1983.

[49] A. G. Vitushkin (ed.), "Several Complex Variables I", Springer-Verlag, Berlin, 1990

[50] Avgoust K. Tsikh, "Multidimensional Residues and Their Applications," Translations of mathematical Monographs, American Mathematical Society, Volume 103,1992.

[51] L. Li, A. Karmouch, and N. D. Georganas, "Synchronization in Real-Time Multi-media Data Delivery," *IEEE ICC Proceedings*, 1992, pp. 587-591.

[52] C. Nicolaou, "An Architecture for Real Time Multi-media Communication Systems," *IEEE J. Select Areas Commun.*, Vol. 8, No. 3, April 1990.

[53] D. Ferrari, "Delay Jitter Control Scheme for Packet-Switching Internetworks," *Comp. Commun.*, Vol. 15, No. 6, pp. 367-373, July/August 1992.

[54] J. Escobar, D. Deutsch, and C. Partridge, "Flow Synchronization Protocol," *Proc. IEEE ICC*, 1993.

[55] T. C. P. Little and A. Ghafoor, "Multi-media Synchronization Protocols for Broadband Integrated Services," *IEEE J. Select Areas Commun.*, Vol. 9, No. 9, December 1991.

[56] Alberto Leon-Garcia, "Probability and Random Processes for Electrical Engineering," Addison-Wesley Publishing Company, 1989.

[57] Athanasios Papoulis, "Probability, Random Variables, and Stochastic Processes," 3rd Edition, McGraw-Hill Book Company, 1991.

[58] W. Allen Smith, "Elementary Complex Variables," Bell & Howell Co., 1974.

[59] H. R. Chillingworth, "Complex Variables," Pergamon Press Ltd., 1973.

[60] Lars V. Ahlfors, "Complex Analysis: An Introduction to the Theory of Analytic Functions of One Complex Variable," McGraw-Hill Book Co., 1953.

[61] Peter Colwell, Jerold C. Mathews, "Introduction to Complex Variables," Bell & Howell Co., 1973.

[62] William Fogg Osgood, "Topics in the Theory of Functions of Several Complex Variables," The Madison Colloquium, 1913, Part II, Dover Publications, Inc., 1966.

[63] "Fifteen Papers in Complex Analysis," American Mathematical Society Transactions, Series 2, Vol. 146, 1990.

[64] Lars Hörmander, "An Introduction to Complex Analysis in Several Variables," North-Holland Mathematical Library, Vol. 7, Elsevier Science Publishers B. V. 1989.

[65] R. Narasimhan, "Analysis on Real and Complex Manifolds," North-Holland Mathematical Library, Vol. 35, Elsevier Science Publishers B. V. 1985.

[66] Walter Rudin, "Function Theory in Polydiscs," W. A. Benjamin, Inc., 1969.

[67] James M. Hyslop, "Infinite Series," 5th Edition, Oliver and Boyd, 1965.

[68] Konrad Knopp, "Infinite Sequences and Series," Dover Publications, Inc. 1956.

[69] Herbert S. Wilf, "Generatingfunctionology," 2nd Edition, Academic Press, Inc. 1994.

[70] J. J. Hunter, "Mathematical Techniques of Applied Probability, Discrete Time Models," Volumes I, II, Academic Press, 1983.

[71] Samuel Karlin, Howard M. Taylor, "A First Course in Stochastic Processes," 2nd Edition, Academic Press, Inc. 1975.

[72] Samuel Karlin, Howard M. Taylor, "A Second Course in Stochastic Processes," 2nd Edition, Academic Press, Inc. 1981.

[73] Stephen Wolfram, "THE MATHEMATICA BOOK", Third Edition, Cambridge University Press, 1996.

[74] William H. Press, etc. all, "Numerical Recipes in C The Art of Scientific Computing 2nd ed." 1992, pp. 636

[75] Ross L. Finny, George B. Thomas, Jr. "Calculus, " Addison-Wesley Publishing Company, 1989.

[76] Bellamy, J. "Digital Telephony," Second Edition, Wiley, New York, 1991

[77] Minzer S., and D. Spears, "New Directions in Signalling for Broadband ISDN," IEEE Communications Magazine, February 1989.

[78] Minzer S. "Broadband ISDN and Asynchronous Transfer Mode (ATM)," IEEE Communications Magazine, September 1989.