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PERFORMANCE EVALUATION OF INTEGRATED SERVICES TELECOMMUNICATIONS NETWORKS

by

WEILIN WANG

A dissertation submitted to the Graduate Faculty in Engineering in partial fulfillment of the requirements for the degree of Doctor of Philosophy, The City University of New York.

1989

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Abstract

Performance Evaluation Of
Integrated Services Telecommunications Networks

by

Weilin Wang

Advisor: Professor Tarek N. Saadawi

This dissertation consists of four parts. First, we summarize the major techniques used in performance evaluation. We then survey the recent research on bandwidth allocation for the integrated services network, particularly the broadband network in the framework of performance evaluation.

Second, we will present the study of trunk congestion control problem arising in multi-service circuit-switched networks operating under non-hierarchical alternate routing.

Third, we will study bandwidth allocation and variation control in the ATM network. The impact on network performance due to the varying-bandwidth services will be analyzed. In addition, a bandwidth allocation algorithm will be proposed and evaluated.

Finally, we will present the design of a performance measurement system for the Ethernet LANs operating in the multimedia environment. Guidelines for implementation are also presented.

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Chapter 1

Introduction

In this chapter, three major techniques used in performance evaluation are briefly discussed. Previous work on network resource management in the framework of performance evaluation is then surveyed. A summary of this dissertation is also included.

1.1 Performance Evaluation Techniques

Telecommunications networks have evolved verv rapidly during the last decade. Recently, several developments have been witnessed in the form of broadcast network, local area network (LAN), integrated services digital network (ISDN) and broadband ISDN (B-ISDN). Performance evaluation of these networks has been, and will remain an important aspect mainly because it helps in 1) assessing a network's performance against overhead introduced; 2) tuning a network's parameters to optimize its performance goals; 3) comparing the performance of two or more networks in quantitative terms.

These are basically three techniques commonly used in network performance evaluation, namely analytical technique, simulation and measurement. In analytical technique performance information is obtained by solving the equations constituting a model, in which queueing theory plays a major role. Quick and yet, in many cases, satisfactorily accurate information can be gathered. In simulation a model is driven with certain inputs and the corresponding outputs are observed and analyzed to determine the performance indexes. Fewer simplifying assumptions are required in deriving a simulation model and hence it may capture more of a hypothetical real system. Model construction and validation are essential and, often, very difficult, since all its relevant behavior and properties should be able to determine in a practical way, given a reasonably limited set of descriptors. The measurement technique, or empirical modeling, on the other hand, does not require a model of the system. It performs measurement on the system itself, either under the real operating conditions or under hypothetically generated conditions. A disadvantage of empirical modelling is the high cost incurred in building the prototype for measurement if the system being modeled is not available.

The performance criteria vary from the user's to the network's viewpoints. For example, a user likes to have a required quality of service with minimal cost, while a network manager concerns more about the maximum utilization of the network resources. Tradeoff must be made between these sometimes conflicting criteria.

1.2 Bandwidth Allocation for Integrated Networks

- Strategies, Techniques and Modelling

In modern telecommunications there is an increasing need to simultaneously transmit heterogenous traffic types (e.g. data, voice, facsimile, video, image) with diverse characteristics and quality of service (QOS) requirements. It has become extremely important to integrate efficiently and equitably these traffic types onto the shared network resources while meeting the QOS requirements.

The network integration may occur in different levels, for example, in access level, transport level, and integrated switching level. The integrated access involves the efficient and equitable sharing, among a set of dissimilar data sources (data terminal, telephone set, LAN, PBX, FAX, TV conference system, etc.) from a single interface to a single link connecting the end-user to a network switching center. ISDN can be viewed as an example of integration at the access level. The integrated transport involves the flexible sharing, among services from many user-interfaces, of transmission link beyond the local access network. It avoids the segregation of different traffic types and media onto different transmission links. The integrated switching involves switching multi-rate, multimedia services within a single switch. It avoids the necessity of adding a new switch type whenever a new service with distinct characteristics is introduced. A network integrated at this level is considered fully integrated since all types of traffic are handled entirely by the same facilities. In the following sections, we will confine our discussion of bandwidth allocation to the integrated access and integrated transport levels. That is, we will describe how the bandwidth allocation functions control the sharing of bandwidth among all services integrated on the local access link and inter-switch link.

Two basic multiplexing techniques have been considered for deploying broadband networks. The first approach assumes a common time reference among the terminals, widely known as the synchronous transfer mode (STM) multiplexing. It is the mechanism employed in most of the modern digital switching machines. An STM link under consideration for broadband applications consists of a set of synchronous time division multiplexed (TDM) channels of various speeds. The other approach assumes no frame reference among the terminals, thus the name asynchronous transfer mode (ATM) multiplexing. An ATM link consists of a single reconfigurable pool of logical channels, each identified by a label called virtual circuit identification (VCI). Bandwidth is allocated dynamically and information is transmitted over these channels in packets (cells). In this way flexibility and efficiency can be achieved. In fact, much focus has been placed on ATM due to its inherent flexibility, service independency [CCITT]. However, it is possible that some continuous bitstream oriented (CBO) services such as switched video in B-ISDN will be STM circuit switched. An alternate approach called hybrid TDM combines STM and ATM by applying ATM to portion of a TDM frame. It emulates circuit switching to guarantee bandwidth with fixed delay while retaining the flexibility of ATM [HUI].

The bandwidth allocation problem has been widely studied in a variety of traffic-handling environments. In the following sections, some of the strategies, techniques and modeling methodologies will be discussed.

1.2.1 Movable Boundary Strategies

ISDN traffic can be assorted into two catagories: real-time traffic, such as voice and video conferencing, and non real-time traffic, such as videotex, file transfer, etc.. The real-time traffic is most adequately transmitted in circuit switching mode while the non real-time is most efficiently transmitted in packet switching form. Most of the studies regarding multiplexing different types of traffic on the same channel in the time division multiplexing (TDM) facility have concentrated on these two types of traffic, specifically voice and data, with voice operating as a loss system and data queued. Both the packet- and call-arrival are mostly modeled by Poisson processes. The objective is typically to control the multiplexer so as to maximize channel utilization or minimize call blocking probability and packet delay.

In a TDM link, different type of traffic may be assigned a separate channel of specified bandwidth (complete partitioning); they might each compete for a group of channels on the FIFO (complete sharing) or priority basis; they might share channels dynamically (e.g. moveable boundary). In the complete sharing scheme, the so-called garbage channel effect may occur [MIYAKE]. This would happen, for example, in a system serving a narrowband (NB) type and a wideband (WB) type traffic, the uncontrolled NB type traffic leaves insufficient bandwidth for accommodating an incoming WB type call, inducing WB call loss. With large bandwidth ratio, this may results in drastic bandwidth inefficiency. On the other hand, the complete partitioning policy may become inefficient under unbalanced load conditions: excessive delay or blocking of some types of traffic may occur while bandwidth allocated to other types is underutilized [KRAI,1].

In the movable boundary scheme, the total bandwidth is partitioned into two compartments of N_C and $N-N_C$ slots each. The compartment of N_C is reserved for circuit-switched traffic, and the other compartment is for packet switched traffic use. The boundary can be partially movable or completely movable. Slotted Envelope Network (SENET) is an example in which the movable boundary scheme is employed.

Leon-Garcia et al. developed and analyzed the performance of data traffic in a SENET-multiplexed link where data traffic is allowed to "borrow" voice traffic capacity that is temporarily idle [LEONGA]. The method is evaluated, using fluid approximation, to be applicable in integrated system where a slow varying traffic class can induce temporary overloads on a packet traffic class by seizing some of the capacity normally available to packet traffic.

Kraimeche and Schwartz [KRAI,1] proposed two strategies for managing a blockable WB traffic and a queueable NB type of traffic. One strategy assigns preemptive priority to the WB traffic over the NB traffic and manages a movable boundary between channel allocation. The other strategy employs a WB to NB bit rate compression mechanism, i.e., deviates the pending WB call request to NB channels. The priority scheme improves the performance of NB traffic for light WB traffic load, while the bit rate compression scheme improves the performance of WB under the condition of light NB traffic. It is also shown that the best combined performance (maximizing WB throughput and minimizing NB delay) is obtained when the two strategies are adaptively combined, i.e. switching from one strategy to another according to the relative offered load of WB and NB traffic. The priority scheme is extended to impose access control on the blockable type traffic in response to the statistical fluctuation of the queue length [KRAI,3]. In this way, a desirable trade-off between the blocking performance, access delay and throughput can be obtained.

It was also proposed in [KRAI,2] that in the all-queueable multi-type traffic system, a sorting policy can be used to attain a good trade off between the channel utilization and fairness. Specifically, the policy maintains a list of waiting customers sorted in descending order of their bandwidth requirements. The first customer on the list that can fit into the available bandwidth slot acquires service. The system has been modeled as a multiserver queueing system with services requiring random number of servers. In a recent study [KRAI,4], a hierarchical structure of channel groups of various speeds accommodating messages on a delay basis at an STM is analyzed. An overflow access control scheme which allows messages arriving at one group to overflow to other groups is shown to improve significantly both the access delay and bandwidth utilization.

[SRIRAM] describes a scheme for integrating packetized voice and data traffic [SRIRAM], which has been implemented in the Integrated Access and Cross-connect System (IACS) at AT&T Bell Lab. In this scheme, voice and data are queued separately to facilitate dynamic bandwidth sharing and mutual overload protection. It guarantees bandwidths to data and voice by reserving T_1 and T_2 time limits for transmitting data and voice packets respectively. When one queue is exhausted, the transmission is immediately moved over to the other queue if it is not empty. A dynamic blocking dropping scheme on voice packets is adopted for congestion control, i.e., during periods of congestion, the less significant blocks of voice packets are dropped. Dropping a small fraction of the less significant bits causes no noticeable degradation of speech but provides significant traffic smoothing advantage [KARANA]. Simulation shows that the scheme enables it to meet the disparate performance requirements for voice and data and to increase the efficiency of transmission bandwidth usage.

1.2.2 Trunk Reservation and Restricted Access Control

One strategy to avoid the garbage channel effect is to reserve a certain amount of bandwidth (e.g. several trunks in a link) for WB call mere use [YAMAGU] [ROBERT] [KAWASH]. This scheme is refereed to as trunk reservation. It restricts the access of NB or non priority calls to the bandwidth resource. The trunk reservation technique can also be used to protect the first offered traffic from the overflow traffic. It is found to be useful to stabilize circuit switched networks operating under non-hierarchical routing which could otherwise exhibit a bistable mode of operation under certain circumstances [AKINPE] [KRUPP].

Bandwidth allocation within the framework of congestion control in the multi-service circuit switched network operating under non-hierarchical alternate routing is studied in [WANG,1]. It is presented in greater detail in Chapter 2.

Mason et al. proposed a multi-slot model for integrating slow-switching WB calls, such as video teleconferences, with NB calls such as telephone traffic [MASON]. The demand consists of two classes of Poisson traffic, queueable WB and blockable NB, both with exponentially distributed holding time. A cutoff parameter r_0 specifies the maximum number of WB calls which are allowed to be simultaneously connected, which protects NB traffic from overload of WB traffic and prevents bimodal stability with the attendant long blocking periods for NB traffic. A M-server queueing model for link performance was derived using Z-transform and matrix geometric techniques, under the simplifying assumption of independence channel release for WB calls. These evaluation methods involve considerable computation, however. Problems of numerical stability were encountered for system having a large number of servers. Liao and Mason then discovered an approximate performance model, which includes an additional overflow NB traffic type [LIAO,1]. A static trunk reservation control parameter r_1 for first offered NB traffic is specified to avoid possible network instability. Another control parameter r_2 is used to protect the WB traffic from overload of NB traffic. Specifically, the system blocks NB calls when there are some WB calls waiting if the number of the WB calls connected or in service is $\leq r_2$. It is shown that the control parameter r_2 improves the link performance as quantified by a power factor measure. The cutoff priority parameter r_0, which protects NB traffic, is also included in the model. It results in smaller power factor. The feasible range of choice for r_0 is limited by the system stability condition, $\lambda_2<r_0\mu_2$, where λ_2 and μ_2 are the WB arrival and holding rates respectively. However, it is required to prevent long blocking periods and bistable behavior for some values of the system parameters. The trunk reservation parameter r_1 will results in a smaller power factor for a single link, but it significantly improves the overall performance of an integrated services networks employing alternate

routing for NB traffic. The WB traffic protection parameter r_2 protects WB traffic from overload of NB traffic and improves the overall system performance even in the case where NB traffic is not unusually heavy. It is also shown that using a sufficiently large r_2 can remove the undesirable feature of oscillation. The approximation is shown to give very good results throughout the useful range of operation. It is also shown to be efficient in terms of both computation time and memory space.

It is demonstrated in [LIAO,2] that it is more efficient to reserve a certain number of trunks for the WB traffic and that the reservation should be a function of the WB calls in service. This model has been successfully applied in an integrated service system where both NB and WB traffics are routed on the basis of revenue maximization via shadow prices. This model also applies to the case where overflow traffic is present. [MIYAKE] proposes an optimal trunk reservation control method, based on Markov Decision process. It is demonstrated that the optimal reservation pattern changes somewhat smoothly according to the WB and NB traffic ratio.

1.2.3 Optimization of Bandwidth Allocation

Conceptually, bandwidth control can be formulated as a constrained dynamic optimization problem. However, it may become prohibitive for most practical cases from the viewpoints of both performance evaluation and implementation.

Wu and Mark [WU] use a non-fixed priority for data and voice transmission over TDM. During periods of large data queue length growth, voice compression takes effect, utilizing Digital Speech Interpolation (DSI) and embedded coding techniques, to devote more transmission capacity to data. It uses dynamic programming in the capacity allocation which jointly optimizes the voice and data performances in a TDM environment. It is observed that the aggregate system throughput can be improved with slight degradation in voice quality.

Tcha et al. modeled the bandwidth allocation as a multiobjective optimization problem using the fuzzy set approach [TCHA], which has been accepted as a tool for dealing with a certain form of imprecision inherent to multiobjective decision-making environment. Some examples of finding the optimal compromise solution for a moderately sized network were shown, at which solution the blocking probability for voice and the average message delay for data are optimally traded off under the fixed boundary multiplexing system.

1.2.4 Multi-Level Bandwidth Control

To economize on voice transmission cost on expensive transmission links such as transoceanic cables and satellite channels, communication bandwidth can be allocated on a talkspurt basis, as employed in the fast circuit switching or burst switching. In this type of switching, bandwidth is allocated at two levels, i.e., call level and talkspurt level. A call set-up request is blocked when the channel is overly congested. Otherwise the call is admitted. In the duration of a call, the channel continuously senses whether the call is in a talkspurt state. Once a talkspurt is initiated, the channel immediately tries to allocate bandwidth (e.g. vacant TDM slots) for the talkspurt. The talkspurt is blocked if the channel fails to allocate the circuit for it. This blocking results in clipping of speech.

To facilitate flexible sharing of network by end user holding multi-rate or bursty service calls, packet switching technology seems promising for implementing an integrated access and transport network. It is suggested that the multi-level control for bandwidth allocation, based on the congestion measures at the packet level, the burst level and the call level, be used for the broadband integrated packet network [HUI]. The multi-level congestion evaluation and control is motivated by the fact that communication terminals often have traffic states characterized by these levels. The call level control emulates circuit-switching for fixed and high bit-rate services; and denies calls when facility overload may cause excessive burst blocking. The burst level control, on the other hand, emulates fast circuit-switching for individual trunks for avoiding excessive packet blocking within a trunk.

For high but fixed rate full motion video calls, or low bit-rate calls such as voice with no silence removal, only the call and packet levels are of concern. For variable bit-rate (VBR) video calls, which is far less bursty than a graphics station, it takes only a small number of such calls before the burstiness is largely averaged out. Therefore the burst level control may not be necessary if the average bit rate of such terminals is much smaller than that of the channel bit rate. For data transfer with low peak rate, its burstiness is of little significance to merit a burst level allocation. For high bit-rate transfer, however, both the call and burst levels control may be used for bandwidth allocation.

1.2.5 Congestion Control and Bandwidth Management for ATM

In an ATM, the access controls process call requests for network resources to support a particular service need, defined by a set of service descriptors. An intermediate adaptation interface at the access converts the user traffic into the appropriate packet format (ATM cells). Within the transport network, the access controls buffer and interleave packets from multiple users on outgoing links.

Congestion control is required to fairly allocate the shared bandwidth in a manner which satisfies the performance requirements for each service. Overcomplicating and optimizing the transport-level congestion control for specific service types is contrary to the underlying objectives motivating transport integration, however.

Since the ATM can not take the window control mechanism as in conventional packet switching because it must handle high speed real-time communications [JOHNSO], the bandwidth should be allocated at call set-up, and the allocated bandwidth is maintained as long as the call continues. There are few analyses of bandwidth allocation algorithms for ATM links [OHTA] [WANG,2] [JOOS], and most of them are heuristic in nature. Ohta et al. presented a bandwidth control algorithm based on the virtual path concept [OHTA]. Bandwidth is allocated by fixed steps (a set of cells) in response to bandwidth increase requests, bounded by the virtual path bandwidth. It rejects a call and keeps the current bandwidth if the bandwidth increase is not possible. Bandwidth is deallocated if possible, according to the virtual path utilization condition. With the simplifying assumption of uniform bandwidth requirement for every call, it was evaluated to be advantageous over the static scheme with no bandwidth control, in terms of transmission efficiency and node processing work load [OHTA]. A dynamic allocation algorithm appears in [WANG,2]. Details can be found in Chapter 4.

In [NOGUCH], two alternatives of bandwidth management strategies were considered. First, the allocated bandwidth is directly determined to assure the user requesting service quality. The call is accepted if the bandwidth needed is available; rejected otherwise. This approach is favorable in terms of QOS, but is extraordinarily complicated in terms of implementation, since it is difficult to assure quality of each call for every traffic condition in the network. Second, the allocated bandwidth is indirectly determined by the distribution of arriving cells during the some unit time interval, i.e. average and variance. If the x% values of superposed distribution for calls simultaneously connected and a new requesting call is less than network resource, the requesting will be accepted; rejected otherwise. The value x depends on the design of communication quality conditions in the switching node. This approach, conversely, is easy to implement. It requires some approximation, however. [NAKAMA] suggested that the bandwidth request be negotiated based on the anticipated load factor obtained by adding the effective bit rate (the maximum bit rate scaled by a factor between 0 and 1, depending the burstiness of the call and the network traffic volume) to the already being used bandwidth. Simulation shows that a request may be accepted if the resultant load factor does not exceed certain value, say, 0.7 to assure the cell loss probability below 10^{-9}. Joos and Verbiest gave a different bandwidth allocation algorithm based on the load control strategy [JOOS], in which the load is estimated according to an approximating Gaussian distribution. The algorithm features low complexity and optimizes QOS and statistical multiplexing gain for VBR sources.

Greater bandwidth efficiency can be attained through the provision of multiple performance classes. With multiple guaranteed performance classes, it is simplest to segregate the total bandwidth available to each class and control traffic admitted to each independently. The number of guaranteed performance classes must be kept to a minimum to retain efficiency benefit and to avoid tailoring the network to specific services. Another approach is to offer guaranteed (high priority) and non-guaranteed (low priority) performance class [WOODRU]. In addition, discarding non-priority cells prevents its adverse influence to the priority cells during a temporary congestion due to insufficient estimation of traffic level [NAKAMA].

In applications such as video transmissions users may vary their bandwidth during a virtual connection session for varied reception quality. Impact of the bandwidth variation on an ATM has been studied in [WANG,3]. Details of the study is presented in Chapter 3.

1.3 Summary of Dissertation

The integrated communications services have very different traffic characteristics from those of conventional telephone calls. The video communications require wideband communications channels, while teletex services necessitate a narrower bandwidth than existing telephone services; the average holding time of teleconferences and terminal to computer data transfer is usually rather long, while that of message services short. Some services such as image and graphics may generate large bursts of data in the network. In this dissertation we will present the performance evaluation of various types of integrated services networks, specifically, heterogeneous circuit-switched networks, ATM networks and multimedia LANs.

1.3.1 Trunk Congestion Control

The problem of trunk congestion in circuit-switched networks concerns link blockings of the network under heavy load conditions. This problem in homogeneous traffic environment has been widely studied [AKINPE] [HAENAC] [YUM]. In Chapter 2, we will present the study on trunk congestion in the heterogeneous environment, i.e., a circuit-switched network with two traffic types. We will analyze and compare the network performance under different routing procedures, and with trunk reservation, restricted access and priority control techniques.

1.3.2 Bandwidth Control for ATM

In an ATM, admission control and enforcement functions must be provided to manage access to bandwidth resource and to prevent the network from unwanted congestion. User-initiated bandwidth variation during a virtual connection must be under control to guarantee the level of performance for the connection blockings and to avoid possible performance instability. In Chapter 3, the impact on the network performance due to the varying-bandwidth services is studied. A means for controlling the bandwidth crossover is presented, and its effectiveness evaluated. In addition, we will study admission control and bandwidth management in Chapter 4. A dynamic bandwidth allocation algorithm to achieve equitable bandwidth accessibility and high throughput will be proposed and evaluated.

1.3.3 Multimedia LANs Performance Measurement

The Ethernet is an examples of LANs designed for local communications of narrow band traffic, typically data. Little is known about the performance of these types of LANs when real-time traffic, such as slow motion video (e.g., compressed and properly coded, with 56 Kbps nominal rate) and voice, mixed with conventional data traffic, is carried in the network. In Chapter 5, we will present the analysis and design of a software system for measuring the performance of the Ethernet LAN in such a multimedia environment.

Chapter 2

Trunk Congestion Control in Heterogenous Circuit-Switched Networks

The integrated communications services have very different traffic characteristics from those of conventional telephone calls. In this chapter, a circuit switched network with two-type traffic of unequal arrival rates and holding times is modeled as a finite two-dimensional Markov chain. The problem of trunk congestion under different routing schemes and controls will be analyzed.

2.1 Introduction

The problem of trunk congestion in circuit-switched networks concerns link blockings of the network under heavy traffic load conditions. Trunk congestion control, an issue of network resources management, determines the quality of the network performance users perceive. The problem of congestion in networks with homogeneous traffic has been widely studied [AKINPE] [HAENAC]. As a means of controlling congestion or protecting the traffic stream the trunk reservation scheme has been receiving much attention with regard to improving the blocking performance of circuit-switched networks using alternate routing [KAWASH] [KRUPP]. [KRUPP] shows that in a circuit-switched network carrying homogeneous traffic, i.e. conventional voice calls, if the traffic is light, alternate routing results in better performance than non-alternate routing. A primary reason for the cost effectiveness of the alternate routing in this case is that the traffic patterns vary in different locations of the network. The alternate routing can be adjusted to evening traffic patterns at various switch stations. However, if the network traffic is heavy and the alternate routing is still used instabilities may be encountered if the load increases beyond the design limit. It also points out that, by reserving a few trunks in each link for direct-routed traffic only, the network can be stabilized and the link blocking performance improved.

In this chapter, we will study the problem of trunk congestion in a two-type traffic circuit-switched networks operating under alternate routing. We will study the trunk reservation with regard to controlling trunk congestion under heavy loads. Also, two access control schemes namely restricted access and preemption will be applied and evaluated. The restricted access control enforces transmission priority by restricting access of one class of calls to the network channel resource in favor of calls of the other class. This scheme, if properly applied, may minimize the blocking of one class of calls while keeping that of another at certain level. In the preemptive priority control scheme, calls of lower priority are preempted if new calls of higher priority arrive and find no free path to the destination. These control schemes can also be used in conjunction with optimal dynamic routing algorithms to yield high grade of service. We shall analyze the performance of the network employing these routing procedures and control policies and, at the end, present numerical examples which justify our conclusions. Applications of the results to heterogeneous traffic systems such as ISDNs are also discussed.

2.2 Heterogeneous Traffic Systems

In this section, we describe the circuit switched network model and the traffic statistical assumptions, and then discuss briefly the performance evaluation criteria used in this study.

2.2.1 Model Description

Trunk congestion is the unavailability of one or more trunks necessary to construct a voice path in the network. It is assumed for this study that the network switches are non-blocking and the set-up time required for call completion in the network is negligible compared to the average holding times of either class calls. This assumption confines our study to a performance analysis of inter-switch trunk congestion. To simplify the analysis the following assumptions are employed.

The network under consideration has N nodes (switches). It is fully connected with all the links being full duplex and each having the same number, say n, of trunks with the same transmission capacity. All node-pair traffic loads are assumed to be equal. For such a fully connected and symmetric model, it is sufficient to describe the traffic statistics for one source-destination node pair on a link.

A call, when accepted to a node of the network for transmission, is to be connected through a path, on a call-by-call basis, to another node in the network. Each call is allowed to attempt one or more paths. Calls unable to find an idle path are blocked and cleared.

The arrival call processes for both classes of traffic entering the network (first-offered or direct-routed traffic) are Poisson with aggregate rates $_1$ and $_2$ respectively; this assumption, upon which much of the analysis is based, hypotheses that arrivals in the network occur independently and the time between successive arrivals of originating calls is negative exponentially distributed.

The alternate-routed (overflow) traffic of the two classes of direct-routed traffic is also assumed to be Poisson. In fact, the overflow traffic is generally peaked. The Poisson approximation is necessary for computational tractability, and is generally acceptable because the total network blocking appears to be rather insensitive to the actual model used, especially with non-hierarchical routing [GIRARD].

The network is in statistical equilibrium. This is required of any traffic systems for which steady-state is desired. Under this stationary assumption, the link blocking probabilities are time independent during the period in which node-node offered loads are valid.

The call holding times for each class of the traffic is exponentially distributed with unequal means $1/\mu_1$ and

$1/\mu_2$, for erlang rates of $A_1=\lambda_1/\mu_1$ and $A_2=\lambda_2/\mu_2$, respectively; this is essential for application to an ISDN.

The alternate paths consist of m disjoint two-link paths, assuming $N \geq m+2$, where N is the number of nodes in the network. Traffic conditions of the successive links are assumed to be independent, for simplicity. As implied by the full connectivity assumption all direct paths consist of one link only.

We also assume that a call blocked on a link of a path can always be cranked back to the originating node so that the call can access the next path in its route.

With such assumptions the network can be represented by a finite two-dimensional Markov chain. We will show in the following sections how to construct this Markov chain and derive difference equations associated with the average occupancy of trunk groups for the network routing considered.

2.2.2 Performance Measures

A very important measure for a circuit-switched network is the call blocking probability, which defines the grade of service of the network. The call blocking probability is defined as the blocking probability of a call originating from a source node and destined for its destination node. This index fixes the probability of being able to establish a communication between any two nodes in the network. The blockings, in fact, depend on a number of factors such as the loads, i.e. the node to node traffic demands, the link capacities and the routing used. Other important measures include circuit throughput, call set-up time. Evaluating the performance of a circuit switched network can be performed using simulation programs. Such programs are generally huge and highly time consuming when dealing with realistic networks. In this case quick, economical and easy to deal with techniques such as queueing analysis are needed. In this study we approach our analysis analytically, namely by establishing and solving difference equations associated with the average occupancy of trunk groups. The link blockings can then be calculated. Based on the assumptions made in the previous subsection, the call blocking probability for each traffic class i, i=1 and 2, is $z_i=p_i(1-q_i^2)^m$ where p_i is the link blocking probability for class i traffic, q_i is the availability of a link for class i alternate routed traffic, and m the number of allowable two-link alternate paths.

2.3 Routing in Heterogeneous Traffic Systems

In this section we assume that the traffic of both classes shares the network trunk resource on an equal basis.

2.3.1 Direct Routing

In this simple form a call can attempt only one (direct) path. If busy, the call is blocked and deemed lost.

The product form solution for the state probability exists. For the external offered loads $a_1=\lambda_1/\mu_1$ and $a_2=\lambda_2/\mu_2$ the link blocking for both classes of traffic, P_b, occurred when all the n trunks of the link are busy, is readily found to be [COOPER]

$$P_b = \frac{(a_1+a_2)^n /n!}{\sum_{k=0}^{n} (a_1+a_2)^k /k!} \qquad (2.1a)$$

And the call blocking probability CBP, is equal to the link blocking probability in this case, i.e., $CBP = P_b$.

The direct routing scheme is characterized by its simplicity and stable blocking performance. The later is essential to networks with high traffic fluctuation rate.

2.3.2 Alternate Routing

In this scheme a call attempts a path (direct path) first. If busy, it will attempt up to m more alternate paths. If all m paths are busy the call is blocked and lost. In network with uniform traffic, it has been shown that the routing scheme can reduce network blocking rate at light load, since it makes use of trunks not currently being used. However, the efficiency deteriorates in overloads. This is due to excessive alternate routing. It causes an increase in the average amount of trunk resource per call, since each alternate-routed call uses two trunks, thereby reducing network utilization. We shall see in the numerical examples that similar observations can be made in the network with two classes of traffic.

We denote the link blocking probability and link availability, respectively, P_b and P_a. Then $(1-P_a^2)$ is the blocking probability of a two-link alternate path. By the symmetry of the network model, each link trunk group is offered an equal amount of the class 1 or class 2 alternate-routed traffic that it overflows to the rest of the network. The total offered load of a link is thus the two classes of direct-routed traffic plus all the class 1 and class 2 alternate-routed traffic, i.e.

$$A_i=a_i+2a_iP_bP_a+2a_iP_b(1-P_a^2)P_a+\ldots+2a_iP_b(1-P_a^2)^{m-1}P_a$$

$$=a_i[1+\frac{2P_b}{P_a}(1-(1-P_a)^m)], \qquad (i=1,2) \qquad (2.3.2a)$$

where a_i, $2a_iP_bP_a$, ..., and $2a_i(1-P_a^2)^mP_a$ are, respectively, the class i direct-routed traffic, the class i alternate-routed traffic overflowed to the first alternate path prescribed when the direct path is busy, ..., and the class i alternate-routed traffic overflowed to the m-th alternate path when the direct path and all the first to (m-1)-th alternate paths are busy. The factor 2 in these terms reflects the fact that the alternate paths consist of two links and therefore use twice as much the trunk resource. With the Poisson assumption of the overflow traffic, product form solution for the state probability exists, and the link blocking probabilities are obtained by summing the state probabilities over the set of all blocking states.

$$P_b = \frac{(A_1+A_2)^n/n!}{\sum_{k=0}^{n}(A_1+A_2)^k/k!} \qquad (2.3.2b)$$

The link availability is simply $P_a=1-P_b$ in this case. Eqs. (2.3.2a) and (2.3.2b) can be solved simultaneously by iterative methods, given the convergence range, the external offered load a_1 and a_2, and the initial arbitrary values of the total offered load A_1 and A_2 as the first estimate. The algorithm successively improves the estimate of the total offered loads and the link blocking. Traffic is successively redistributed through the network based on network routing and previous link blocking estimates, which result in new total offered loads and corresponding link blockings. The iterative process concludes when subsequent iterations produce no substantial alternation in link blockings.

The call blocking probability for either class of traffic corresponds to the case that the direct path and the alternate paths are all busy, and is thus given by

$$CBP = P_b(1-P_a{}^2)^m \qquad (2.3.2c)$$

2.3.3 Application of Trunk Reservation

The effect of excessive alternate routing can be mitigated by the trunk reservation technique in which the flow of overflow traffic is restricted. In this scheme, an overflow call is denied access to a trunk group if a minimum number, say r, of idle trunks are not available. For links with more than r idle trunks, overflow traffic is not blocked, however. The offered loads are given by

$$A_i=a_i[1+2P_b(1-(1-P_{a,r}{}^2)^m)/P_b] \quad (i=1,2) \qquad (2.3.3a)$$

where $P_{a,r}$ is the link availability for class i overflow traffic; otherwise overflow traffic is blocked; the offered load is then simply the external load a_i. The state equations are as follows.

$$(j_1\mu_1+j_2\mu_2+A_1\mu_1+A_2\mu_2)P(j_1,j_2)=A_1\mu_1P(j_1-1,j_2)+$$
$$A_2\mu_2P(j_1,j_2-1)+(j_1+1)\mu_1P(j_1+1,j_2)+(j_2+1)\mu_2P(j_1,j_2+1)$$
$$(j_1+j_2<n-r+1) \qquad (2.3.3b)$$

$$(j_1\mu_1+j_2\mu_2+a_1\mu_1+a_2\mu_2)P(j_1,j_2)=a_1\mu_1P(j_1-1,j_2)+$$
$$a_2\mu_2P(j_1,j_2-1)+(j_1+1)\mu_1P(j_1+1,j_2)+(j_2+1)\mu_2P(j_1,j_2+1)$$
$$(n-r+1<j_1+j_2<n) \qquad (2.3.3c)$$

with the boundary conditions

$$(j_1\mu_1+j_2\mu_2+a_1\mu_1+a_2\mu_2)P(j_1,j_2)=A_1\mu_1P(j_1-1,j_2)+$$
$$A_2\mu_2P(j_1,j_2-1)+(j_1+1)\mu_1P(j_1-1,j_2)+(j_2+1)\mu_2P(j_1,j_2+1)$$
$$(j_1+j_2=n-r+1) \qquad (2.3.3d)$$

and

$$(j_1\mu_1+j_2\mu_2)P(j_1,j_2)=a_1\mu_1P(j_1-1,j_2)+a_2\mu_2P(j_1,j_2-1)$$
$$(j_1+j_2=n) \qquad (2.3.3e)$$

where $P(j_1,j_2)$ must satisfy $\sum P(j_1,j_2)=1$. $P(j_1,j_2)$ can be solved for by using an iterative method called successive overrelasation (SOR) [COOPER]. Then

$$P_b = \sum_{j_1+j_2=n} P(j_1,j_2) \qquad (2.3.3f)$$

$$P_{a,r} = \sum_{j_1+j_2<n-r+1} P(j_1,j_2) \qquad (2.3.3g)$$

and

$$CBP = P_b(1-P_{a,r}{}^2)^m \qquad (2.3.3h)$$

2.4 Access Control and Preemptive Priority Enforcement

Two control policies, namely access control and preemption [KRAI,1], can be applied to regulate between the two classes of traffic the access to the trunk resource. In systems employing these access controls a call is either blocked (all allowable routes contain at least one trunk group where all trunks are busy) or rejected (at least one allowable route has one free trunk in all its trunk groups, but yet the call attempt is rejected by the control policy to protect future call attempts of higher priority), or routed. Practically, priority can be given to the class of traffic with longer average holding time and thus reduce trunk idle time and total call set up time. However, in applications where the network traffic consists of human-human communications (e.g. voice calls) and machine-machine communications (e.g. data transfer), it is desirable to give priority to human-human communications services. These control schemes can also be used in conjunction with optimal dynamic routing algorithms to yield high grade of service.

2.4.1 Restricted Access Control

This is equivalent to reserving certain amount of the trunk resource for one class of the calls which assumes higher transmission priority. Specifically, we reserve $n_1<r$ trunks for class 1 direct traffic. A class 2 call will be blocked if the link has n_1 or fewer than n_1 free trunks.

$$A_i=a_i[1+2P_{bi}(1-(1-P_{ai,r}{}^2)^m)/P_{bi}]$$
$$(i=1,2) \qquad (2.4.1a)$$

for links with more than r free trunks, and $A_i=a_i$, for links with r or fewer free trunks. The state equations can be found as

$$(j_1\mu_1+j_2\mu_2+A_1\mu_1+A_2\mu_2)P(j_1,j_2)=A_1\mu_1P(j_1-1,j_2)+$$
$$A_2\mu_2P(j_1,j_2-1)+(j_1+1)\mu_1P(j_1+1,j_2)+(j_2+1)\mu_2P(j_1,j_2+1)$$
$$(j_1+j_2<n-r+1, j_2<n-n_1+1) \qquad (2.4.1b)$$

$$(j_1\mu_1+j_2\mu_2+a_1\mu_1+a_2\mu_2)P(j_1,j_2)=a_1\mu_1P(j_1-1,j_2)+$$
$$a_2\mu_2P(j_1,j_2-1)+(j_1+1)\mu_1P(j_1+1,j_2)+(j_2+1)\mu_2P(j_1,j_2+1)$$
$$(n-r+1<j_1+j_2 \text{ or } n-r+1<j_2,\ j_1+j_2<n,\ j_2<n-n_1) \qquad (2.4.1c)$$

with the boundary conditions

$$(j_1\mu_1+j_2\mu_2+a_1\mu_1+a_2\mu_2)P(j_1,j_2)=A_1\mu_1P(j_1-1,j_2)+$$
$$A_2\mu_2P(j_1,j_2-1)+(j_1+1)\mu_1P(j_1-1,j_2)+(j_2+1)\mu_2P(j_1,j_2+1)$$
$$(j_1+j_2=n-r+1,\ j_2<=n-r-n_1+1) \qquad (2.4.1d)$$

$$(j_1\mu_1+j_2\mu_2+a_1\mu_1+a_2\mu_2)P(j_1,j_2)=a_1\mu_1P(j_1-1,j_2)+$$

$$A_2\mu_2P(j_1,j_2-1)+(j_1+1)\mu_1P(j_1-1,j_2)+(j_2+1)\mu_2P(j_1,j_2+1)$$

$$(j_2=n-r-n_1+1,\ j_1<n_1) \qquad (2.4.1e)$$

$$(j_1\mu_1+j_2\mu_2)P(j_1,j_2)=a_1\mu_1P(j_1-1,j_2)+a_2\mu_2P(j_1,j_2-1)$$

$$(j_1+j_2=n,\ j_2<=n-n_1) \qquad (2.4.1f)$$

and

$$(j_1\mu_1+j_2\mu_2+a_1\mu_1)P(j_1,j_2)=a_1\mu_1P(j_1-1,j_2)+a_2\mu_2P(j_1,j_2-1)+$$

$$\mu_1(j_1+1)P(j_1+1,j_2)$$

$$(j_2=n-n_1,j_1<n1) \qquad (2.4.1g)$$

where $P(j_1,j_2)$ satisfies $\sum P(j_1,j_2)=1$. The blocking rates

$$P_{b1} = \sum_{j_1+j_2=n} P(j_1,j_2) \qquad (2.4.1h)$$

and

$$P_{b2} = P_{b1} + \sum_{\substack{j_2=n-n_1\\ j_1<n_1}} P(j_1,j_2) \qquad (2.4.1i)$$

The availability for the alternate path, $P_{a,r}$ is then

$$P_{a,r} = \sum_{\substack{j_1+j_2<n-r+1\\ j_2<n-r-n_1+1}} P(j_1,j_2) \qquad (2.4.1j)$$

And the call blocking probabilities

$$CBP_1 = P_{a,r}(1 - P_{b1}{}^2)^m \qquad (2.4.1k)$$

and

$$CBP_2 = P_{a,r}(1 - P_{b2}{}^2)^m \qquad (2.4.11)$$

As can be seen in the numerical examples this scheme may minimize the blocking probability of one class of calls while keeping that of the other at certain level; the average blockings are improved under certain traffic load conditions.

2.4.2 Preemptive Priority Control

The restricted access control may be inefficient for calls of lower priority as they may be frequently rejected while certain amount of trunk resource is idle in the portion allotted to the higher priority traffic. One way to improve the situation is to allow the lower priority calls to access the trunk resource with the risk of being preempted should a higher priority call arrive and find no free trunks to its destination. Preempted calls are assumed lost for simplicity. This scheme may also apply in the situation when one class of the calls is emergent and requires prompt service.

Similar to the case with access control the offered loads are given by (2.4.1a), the state equations by (2.4.1b) and (2.4.1c), and the interior boundary conditions by (2.4.1d) and (2.4.1e). The outer boundary conditions are updated to reflect the preemptive priority policy as follows.

$$(j_1\mu_1+j_2\mu_2+a_1\mu_1(1-\text{delta}(j_2,n_1)))P(j_1,j_2)=$$
$$a_1\mu_1P(j_1-1,j_2)+a_2\mu_2P(j_1,j_2-1)+a_1\mu_1P(j_1-1,j_2+1)$$
$$(j_1+j_2=n,\ j_2<=n-n_1) \qquad (2.4.2a)$$

where delta() is the delta function, and

$$(j_1\mu_1+j_2\mu_2+a_2\mu_2)P(j_1,j_2)=a_1\mu_1P(j_1-1,j_2)+a_2\mu_2P(j_1,j_2-1)$$
$$(j_2=n-n_1,\ j_1<n_1) \qquad (2.4.2b)$$

The link blocking probabilities P_{b1} and P_{b2}, alternate path availability, $P_{a,r,p}$, call blocking probabilities CBP$_1$ and CBP$_2$ are, respectively, given by (2.4.1h)-(2.4.1l). Another performance measure, i.e. the probability of class calls being preempted, is

$$P_{pe} = \sum_{\substack{j_1+j_2=n \\ j_2>0}} P(j_1,j_2) \qquad (2.4.2c)$$

2.5 Comparative Performance

First we compare the call blocking performance of alternate routing and direct routing. Consider a network with trunk capacity of n=10 for each trunk group, handling two classes of calls. The two curves labeled m=0 and m=1 correspond to the blocking probability of direct routing and alternate routing with one alternate path per source-destination node pair. It is shown that the call blocking probability of alternate routing is much more sensitive to the increase of the external offered loads than that of direct routing. This is similar to the instability observed in the homogeneous network [AKINPE]. The curve labeled m=1 in Fig. 2.5 (a) shows how using alternate routing improves the blocking performance at low traffic load.

Now consider alternate routing with trunk reservation. We again take n=10 and m=1. In Fig. 2.5b, three curves are shown where r=0, r=3 and r=10 trunks, respectively, are reserved for direct-routed traffic. The two special cases, r=0 and r=10, correspond to alternate and direct routing respectively are plotted as reference. It is shown that the trunk reservation scheme improves the blocking performance at the heavy traffic load over the alternate routing with no trunk reservation while maintains the blockings at the level of direct routing at light load. This assembles the observation in homogeneous networks [AKINPE].

Next we compare the blocking performances for the two classes of traffic under the access control. In Fig. 2.5c, the two curves labeled CBP_1 and CBP_2, respectively, are the call blocking probabilities for class 1 and class 2 traffic under the control policy where $n_1=3$ trunks for each link are reserved for class 1 traffic and class 2 traffic will be rejected if the link has three or fewer than three free trunks. These two curves are compared with the one labeled CBP, the call blocking probability for both classes of traffic under no access control. The performance of class 1 traffic improves while that of class 2 degrades, as the consequence of the biasing control policy. The average blocking performance under the access control policy also improves over that with no access control.

Finally, we compare the blocking performances under the restricted access and preemptive priority control. In Fig. 2.5d, the curves shows how using the preemptive priority control upgrades the blocking performance of class 2, with probability P_{pe} being preempted.

2.6 Summary and Conclusion

In this chapter, several types of routing and controls for heterogeneous circuit switched networks with two classes of traffic of unequal holding times were introduced. A performance evaluation was made, based on a fully connected symmetrical network model. It was shown that the trunk reservation technique can be applied to alleviate trunk congestion caused by excessive alternate routing at heavy loads. It was further shown that the restricted access and preemptive priority control can be used to enforce certain routing priority ordering over two classes of traffic, keeping call blocking probability of one class of traffic at certain level while minimizing that of the other class. These facts are significant in applications of integrated communications services where various classes of services have different traffic characteristics and service requirements.

Chapter 3

Bandwidth Variation and Control for ATM Networks

The technological advances and market demand have been driving the evolution towards B-ISDN supporting multimedia broadband services such as high speed data, image and video services with bit rates up to 600 Mbps. ATM is considered as candidate for transport multiplexing technique across the User Network Interface (UNI) based on its inherent flexibility, service independency, and high efficiency [CCITT].

During a virtual connection session, users could vary the bandwidth of their applications, such as video or image, to achieve different quality of reception. They could also reshape their traffic just to avoid certain undesired enforcement actions by the network. In this and the following chapters, we will study the admission control, bandwidth allocation and the bandwidth variation problems. In this chapter, the impact on the ATM network performance due to varying-bandwidth services is analyzed based on the simplified virtual network model detailed in Section 3.2. A dynamic bandwidth allocation and control algorithm is proposed and evaluated in Chapter 4. Numerical examples are also provided to illustrate the comparative performance.

3.1 Introduction

The Asynchronous Transfer Mode (ATM) is considered a viable candidate as the transport multiplexing technique for the broadband ISDN. In an ATM admission control and enforcement functions must be provided to manage access to bandwidth resource and to prevent the network from unwanted congestion. User-initiated bandwidth variation during a virtual connection must be under control to guarantee the level of overall performance for the connection blockings and bandwidth efficiency, and to avoid possible performance instability. In this chapter, a virtual network model of ATM is described, and the impact on the network performance due to the varying-bandwidth services is studied. A means for controlling the bandwidth crossover is presented, and its effectiveness is evaluated.

3.2 ATM Networks and Virtual Paths

ATM is a packet based transfer mode using asynchronous time division multiplexing technique. In an ATM network, hierarchical structure of channels/paths (bundles of circuits) and the resultant complicated TDM frame structure of interface can be eliminated [OKANO]. Moreover, it is capable of multiplexing/demultiplexing services of various bit rates while employing simple network hardware and software. These are made possible by transporting the packets (cells) in the network independent of specific channel rates and by the store and forward process of transportation.

In an ATM, the user information is carried in cells which are fixed sized blocks and consist of a header and an information field. The header contains a label which uniquely identifies a channel and is used for multiplexing and routing. The information field is transported transparently by the ATM layer. Since the interface structure contains a set of labeled channels (logically bound rather than physically positioned), an ATM can dynamically adapt to the changes in the service demand and allocate the resources such as channel bandwidth accordingly.

An ATM network model based on the concept called "Virtual Path" (VP) is described in [OHTA] [TOKIZA]. In this model two nodes are connected by a logical link called virtual path. A virtual path consists of a number of virtual circuits and has its own bandwidth which gives the upper limit of bandwidth of the virtual circuits enclosed. Virtual paths are multiplexed in a physical link on a cell multiplexing basis. They are established or released dynamically based on 1) long term service provision; 2) demand for alternate routing in case of network failure; and 3) short term demand. The capacity of the virtual paths can be dynamically allocated in the same fashion. The concept of virtual path is expected to play a significant role in constructing a simplified network architecture [SATO].

The cell header contains two subfields, i.e., Virtual Path Identifier (VPI) and Virtual Channel Identifier (VCI). The separation of a VPI from a VCI eliminates call-by-call basic processing at each transit node during call establishment. During the call set-up process, the access node identifies the appropriate path, or creates one if necessary, and makes the decision whether to accept or reject the call based on the current path bandwidth utilization conditions. It may be advantageous to allocate virtual paths along the same route to group together traffic of similar characteristics in order to be able to optimize the resources and policing characteristics [FISHER].

The Virtual Path scheme has the advantages of 1) simplified network architecture attained by the separation of network transport functions; 2) easier to implement dynamic path routing and dynamic path bandwidth allocation, thus increases the adaptability to varying network traffic; 3) flexible network reconfiguration capability, which provides the network with increased reliability; 4) elimination of call-by-call based call set-up processing at transit nodes.

3.3 Impact of Varying-Bandwidth Services and Performance Measures

In applications such as video transmissions users may vary their bandwidth during the session for varied reception quality. For example, a user may request connection for a video conference call at a low bit rate, say, 56 Kbps. Then, during the video session, the user may desire to increase the video transmission rate to, say, 384 Kbps for better quality. Another scenario is a user voluntarily reduces the transmission bandwidth to avoid certain undesired policing actions by the network. The probability of a user's request for bandwidth change being honored is an important performance measure in the network where multi-bit-rates are supported. We will refer to this probability as the crossover availability (P_{ca}), since it fixes the rate at which a call can actually crossover from one bandwidth destination to another. Two other related measures can also be defined- the probability of a bandwidth change request being denied, referred to as crossover blocking probability (P_{cb}), and the probability of no bandwidth change requests, referred to as non-crossover probability (P_{nc}). Notice that $P_{ca}=1-(P_{cb}+P_{nc})$. It may be desireable in some situations that excessive crossing over be avoided. This can be achieved by limiting the amount of bandwidth that can be occupied by calls crossed over. It is possible that a user intends to switch back and forth among different bit-rates during the session. Some network enforcement functions may be necessary to avoid possible instability of network performance due to such user behavior.

To be tractable for the performance evaluation of a network in which users may vary their bandwidth during the session, we assume that there are two virtual paths in an ATM network link, each serving one class of users with similar requirements. A service class i, i=1,2, is identified by its Poisson arrival rate λ_i, its mean service time $1/\mu_i$, and bandwidth requirement in terms of the number of cells required t_i. A class 1 call which requires t_1 cells may, with probability e (referred to as crossover rate), request to increase its bandwidth to t_3 cells after an exponentially distributed holding time period with mean being $1/\mu_3$ time units, and continue the call with the increased bandwidth (classified as class 3 call, holding for another exponentially distributed duration with mean $1/\mu_4$) till the session ends. If the bandwidth increase request is denied, due to either lack of bandwidth or network control policy, the call continues with the current bandwidth t_1. A class 2 call, however, maintains its bandwidth t_2 for the entire session. Class 1 and 2 calls are allowed to occupy the bandwidth resource up to $(n-r_1)$ and $(n-r_2)$ cells, respectively, for an ATM link with capacity of n cells. Parameters r_1 and r_2 are introduced for bandwidth allocation and control, which is detailed in Section 3.4; r_3, on the other hand, specifies the amount of bandwidth reserved for the constant-bandwidth calls. Equivalently, $(n-r_3)$ is the upper limit of bandwidth for crossed over calls. Let $P(j_1,j_2,j_3)$ be the equilibrium state probability that there are in progress j_1 class 1 calls, j_2 class 2 calls, and j_3 calls which are spawned from the portion of class 1 calls which have changed their bandwidth during the session. A sample state space for $n=4$, $t_1=1$, $t_2=t_3=2$ is depicted in Fig. 3.1. Steady state equations associated with a typical state (cf Fig. 3.2) can be found as follows.

$$\begin{aligned}
&(\lambda_1+\lambda_2+e\mu_3 j_1+(1-eI(j_1-1,j_2,j_3+1))\mu_1 j_1+\\
&\quad \mu_4 j_3+\mu_2 j_2)P(j_1,j_2,j_3)\\
&= (1-eI(j_1,j_2,j_3+1))\mu_1(j_1+1)P(j_1+1,j_2,j_3) +\\
&\quad \mu_2(j_2+1)P(j_1,j_2+1,j_3)+e\mu_3(j_1+1)P(j_1+1,j_2,j_3-1) +\\
&\quad \lambda_1 P(j_1-1,j_2,j_3)+\mu_4(j_3+1)P(j_1,j_2,j_3+1) +\\
&\quad \lambda_2 P(j_1,j_2-1,j_3)\\
&(0\le t_i j_i\le n-r_i\ ,\ \overline{\sum} t_i j_i\le n-\max(t_1,t_2,t_3),\ i=1,2,3) \qquad (3.1)
\end{aligned}$$

where $t_1\le t_2$, $t_1\le t_3$, $1/\mu_3\le 1/\mu_1$, and $1/\mu_4\le 1/\mu_1$); function $I(j_1,j_2,j_3)=1$ if (j_1,j_2,j_3) is inside the state space, 0 otherwise. $P(j_1,j_2,j_3)$ is subject to the normalization condition

$$\sum_{j_i} P(j_1,j_2,j_3)=1 \qquad (i=1,2,3), \qquad (3.2)$$

The probability of a VP_i user virtual connection request being rejected, or simply the connection blocking probability, is given as follows.

$$SRP_i = \sum_{\substack{(t_1 j_1+t_2 j_2+t_3 j_3+t_i>n)\\ \vee(t_i j_i+t_i>n-r_i)}} P(j_1,j_2,j_3)$$

$$(i=1,2), \qquad (3.3)$$

The crossover blocking probability can be evaluated as

$$P_{cb} = \sum_{\substack{((t_1 j_1+t_2 j_2+t_3 j_3+t_3-t_1>n)\\ \vee(t_3 j_3+t_3>n-r_3))\wedge(j_1>0))}} P(j_1,j_2,j_3) \qquad (3.4)$$

And the crossover availability

$$P_{ca} = \sum_{\substack{(t_1 j_1+t_2 j_2+t_3 j_3+t_3-t_1\le n)\\ \wedge(t_3 j_3+t_3\le n-r_3)\wedge(j_1>0))}} P(j_1,j_2,j_3) \qquad (3.5)$$

Notice that the condition $(j_1>0)$ on the summations above excludes those states involving no bandwidth crossing over (i.e., $j_1=0$, or no class 1 calls in the network).

Iterative numerical methods may be used to obtain state probability distribution, as in Chapter 2, based on which the above listed parameters can be evaluated. The storage requirement for the coefficient matrix associated with the state space is of the order $O(((n/t_1)(n/t_2)(n/t_3))^2)$. Significant storage saving in the numerical computation can be achieved by taking into account the fact that the coefficient matrix is banded with bandwidth (number of non-zero elements centered around the diagonal for a typical row) much smaller than its dimension k for large k.

3.4 Numerical Examples

In this section the impact of the varying-bandwidth services on the network performance will then be evaluated, based on the model described in the previous sections. The connection blocking probability, crossover availability and throughput are used as evaluation criteria.

In Fig. 3.4(a) the crossover availability P_{ca} is plotted versus the crossover rate e. It shows that as e increases the crossover is less possible. The two curves in Fig. 3.4(b) indicate that the service connection blocking probabilities SRP_1 and SRP_2 both increase as e. This is due to the fact that calls crossed over from narrowband to wideband use more network bandwidth, resulting in high blockings for new connection requests. The network overall throughput (weighted with respect to bandwidth usage) may increase as the network carries more calls switched from narrowband to wideband. This is exampled by Fig. 3.4(c). To limit the crossover a non-zero r_3 can be specified. Figs. 3.4(d) and 3.4(e) show the effect on the crossover blocking probability and service connection blocking probability - the crossover blocking increases while connection blockings decrease slightly. This is because the network carries fewer crossover calls of increased bandwidth when bandwidth crossover is limited.

3.5 Summary

In this chapter the impact on the ATM network performance due to the bandwidth variation during a session is analyzed, based on a simplified model described in Section 3.2. It is shown that as more calls intend to cross-over from a lower bandwidth destination to a higher one the network suffers from a higher connection blocking. The crossover availability declines as more calls cross-over and increase the bandwidth used. Slight increase of overall throughput is observed, owing to the fact that the network carries more crossover calls with the increased bandwidth. It is also shown that limitation can be placed on the amount of bandwidth that the crossover calls can occupy to protect the constant-bandwidth calls and to keep the network connection blockings below certain level.

Chapter 4

Bandwidth Allocation for ATM Networks

The integrated broadband network is to provide multimedia services such as digitized voice, high speed data, image and video, and therefore must offer high throughput and sufficient flexibility to support the wide spectrum of services. The Asynchronous Transfer Mode (ATM) network is considered a viable candidate as the transport multiplexing technique across the User Network Interface (UNI) of the broadband ISDN (B-ISDN) based on its inherent flexibility, service independency and high performance [CCITT]. Admission control and enforcement functions must be provided at the access points to the ATM network, however, to avoid unwanted network congestion and achieve a guaranteed level of transport performance. The admission control functions determine whether to accept or deny a virtual connection bandwidth request based on the traffic flow descriptors and the current bandwidth usage conditions. The enforcement functions, on the other hand, ensure an application's offered traffic remains within the bounds specified in call set-up request. In Chapter 3, the impact on the ATM network performance due to varying-bandwidth services is analyzed based on the simplified virtual network model detailed in Section 3.2. A dynamic bandwidth allocation and control algorithm will be proposed and evaluated in this chapter. Numerical examples will also be provided to illustrate the comparative performance.

4.1 Introduction

In an ATM, the access controls process call requests for network resources to support a particular service need, defined by a set of service descriptors. An intermediate adaptation interface at the access converts the user traffic into the appropriate packet format (ATM cells). Within the transport network, the access controls buffer and interleave packets from multiple users on outgoing links. Congestion control is required to fairly allocate the shared bandwidth in a manner which satisfies the performance requirements for each service. Overcomplicating and optimizing the transport-level congestion control for specific service types is contrary to the underlying objectives motivating transport integration, however.

Since the ATM can not take the window control mechanism as in conventional packet switching, the bandwidth should be allocated at call set-up, and the allocated bandwidth is maintained as long as the call continues. There have been few analyses of bandwidth allocation algorithms for ATM links, and most of them are heuristic in nature. Due to the statistical nature of ATM bandwidth moving across the UNI, the network must be able to characterize the subscriber's traffic and determine if the service request will be granted and the call admitted to the network to ensure equitable access to network bandwidth for all contending services. And, when a call is admitted, network resources such as link bandwidth must be allocated and controlled so as to ensure high transmission efficiency.

The congestion control in an ATM can be divided into three levels [WOODRU]. The first level, termed route control, assigns routes through the network for each access node pair. The objective of route control is to make efficient use of the network resources. These route could be dynamically assigned for each connection request according to the state of the network, or could simply be statically assigned for a longer duration, depending on the traffic environment and the degree of efficient resource utilization required. The second level, termed admission control, decides whether to accept or reject a connection request, based on knowledge of current network loading, the new connection's anticipated traffic characteristics, and its performance requirements. The third level, termed bandwidth enforcement, polices individual connections to ensure that their traffic flow admitted to the network conforms to that specified at call set-up time by throwing away or buffering violating packets at the network inlet. The objectives of admission control and bandwidth enforcement are to ensure that a satisfactory level of transport performance is provided to all users.

A framework for ATM network resource management is described in by the Consultative Committee for International Telephone and Telegraphy (CCITT) Recommendation I.121 [CCITT]:

1) At call set-up, a user declares the burstiness characteristics of the call to the network with call control parameters.

2) Based on these characteristics, the network allocates the necessary resources to assure the quality required for cell transfer delay and cell loss. If there are insufficient network resources available, the call will be rejected.

3) The network monitors the cell stream coming from the user to verify that the stream conforms to the parameter values declared by the user.

In this chapter, the problem of network admission control is studied in greater detail. A dynamic bandwidth allocation algorithm is proposed. It is evaluated and compared with other existing schemes.

4.2 Bandwidth Allocation Algorithms

The bandwidth requirements vary with changes in concurrent connections and types of services. In the algorithm based on static allocation, a virtual path is allocated a fixed amount of bandwidth. The static-allocated bandwidth must be large enough to cope with the pre-defined maximum concurrent connections for a particular type of services. This simple allocation scheme suffers from a potentially dramatic bandwidth inefficiency - connection requests are rejected in some virtual paths of a link due to excessively heavy bandwidth usage while the bandwidth of other paths in the same link may be underutilized, and, due to its static nature, can not cope with bursty traffic such as image.

A bandwidth control algorithm based on the virtual path concept was proposed in [OHTA], and is listed as follows.

1) Request bandwidth increase if it is insufficient for the new call arriving at the end node.

2) If the increase is allowed, then increase the bandwidth and set up a virtual circuit for the call; else, keep the current bandwidth and refuse the call.

3) Decrease the bandwidth if possible, according to the virtual path utilization conditions.

The algorithm allows some variations by the unit of bandwidth changes or the range of control. It has been analyzed for a simplified model. In this model, the bandwidth of each virtual path accommodated in a link is incremented or decremented in steps according to the amount of calls offered to each path, where every call is assumed to have the identical bandwidth requirement (i.e. one bandwidth unit or cell per call). The bandwidth of each call and path are assumed deterministic (i.e., no statistical effect on bandwidth is considered). Also, average traffic offered to each virtual path is assumed identical and modeled as Poisson process. With these assumption, one of the networks links was analyzed. It is demonstrated that the transmission efficiency is considerably improved, compared to the static scheme with no bandwidth control. We propose here another algorithm as follows which allocates and controls the bandwidth dynamically and avoids bandwidth fragmentation intrinsic to the "fixed step" allocation scheme adopted in the algorithm above.

1) Each virtual path (VP) in a link is allocated a number of cells (pre-allocated, uncontended bandwidth) which satisfies the minimal bandwidth requirements for the path.

2) Reserve a number of cells (reserved, contending bandwidth) for dynamic allocation to meet the need of bandwidth increase among virtual paths of the same link.

3) Request for bandwidth increase is honored if sufficient reserved bandwidth is available. When honored, the virtual path receives a bandwidth increase as requested; otherwise it keeps the current bandwidth and reject the call.

4) Bandwidth beyond the pre-allocated amount is deallocated upon completion of the calls occupying it.

The bandwidth increase is by calls (variable number of cells) rather than by some fixed steps (fixed number of cells), and thus avoids bandwidth waste. Pre-allocation of a number of cells to a path is necessary to ensure equitable access to bandwidth for all contending services and to guarantee the level of performance for the users on that path. A call is admitted to the network only when the VP has large enough bandwidth to accommodate or when the bandwidth increase contention succeeds; it is rejected otherwise.

4.3 Algorithm Analysis for a Simplified Model

To simplify the formulation and evaluation of the bandwidth allocation algorithm at the admission control, we assume that there are two virtual paths in an ATM link, each serves one class of users with similar requirements. The call arrival process for both paths are assumed Poisson with mean λ_1 and λ_2. Also we assume that the bandwidth of each call and path are deterministic, i.e., no statistical effect on the bandwidth will be considered. The bandwidth resources (cells) are allocated and released dynamically as described in the algorithm proposed in section 4.2.

Let $P(j_1,j_2)$ be the statistical-equilibrium probability that there are j_i VP_i users in service $(i=1,2)$, a_i be the external offered load to victual path VP_i $(a_i=\lambda_i/\mu_i)$; n the link capacity (cells); u_i the VP_i users holding rate; t_i the VP_i users bandwidth requirement (cells); $r_{i'}$ the bandwidth pre-allocated to VP_i users (where i' is the complement of i, i.e., $i'=1$ when $i=2$ and $i'=2$ when $i=1$), or, equivalently, $(n-r_1)$ and $(n-r_2)$ are the upper limits of bandwidth for VP_1 and VP_2 calls, respectively; and, without loss of generality, $t_1 \leq t_2$. Also assume that $P(j_1,j_2)=0$ if (j_1,j_2) is outside the state space. Statistical-equilibrium probability equations associated with a typical state are given as follows.

$$(a_1\mu_1+a_2\mu_2+j_1\mu_1+j_2\mu_2)P(j_1,j_2) =$$
$$(j_1+1)\mu_1P(j_1+1,j_2)+(j_2+1)\mu_1P(j_1,j_2+1)+$$
$$a_1\mu_1P(j_1-1,j_2)+a_2\mu_2P(j_1,j_2-1)$$

$$(0\le t_ij_i\le n-r_i-t_i, 0\le\sum t_ij_i\le n-t_2)) \qquad (4.1)$$

The normalization equation is

$$\sum_{j_i} P(j_1,j_2) = 1 \qquad (i=1,2), \qquad (4.2)$$

The product form solution exists, i.e.

$$P(j_1,j_2) = \frac{a_1{}^{j_1}a_2{}^{j_2}}{j_1!j_2!} P(0,0) \qquad (4.3)$$

where P(0,0) can be determined numerically using the normalization condition Eq.(4.2).

Notice that Eq.(4.1) is simply a special case of Eq.(3.1) with $j_3=0$, i.e., with no bandwidth crossover allowed. Rejection of a VP_i user connection request occurs when the number of free cells in the pre-allocated bandwidth is used up and that in the reserved bandwidth is smaller than the requested bandwidth increase for the VP_i user. The probability of a VP_i user's connection request being rejected is given by

$$SRP_i = \sum_{\substack{(t_1 j_1 + t_2 j_2 + t_i > n) \\ \vee (t_i j_i + t_i > n - r_i)}} P(j_1, j_2) \quad (i=1,2), \tag{4.4}$$

And, the cell throughput is

$$C_i = a_i t_i (1 - SRP_i) \quad (i=1,2), \tag{4.5}$$

4.3 Numerical Examples

In this section the bandwidth control algorithm proposed in the last section is evaluated in terms of virtual connection request blocking rate and the cell throughput. Some numerical examples are presented to illustrate the comparative performance.

Fig. 4.1 shows that the bandwidth control algorithm with parameters as specified brings close the blocking rates of the connection requests from users of different virtual paths, in comparison to those with no bandwidth control. This is essential in achieving equitable accessibility to bandwidth resource for all contending users with diverse requirements. Plots in Fig. 4.2 shows, on the other hand, that the control algorithm improves the overall cell throughput significantly for users with vastly different bandwidth requirements.

4.6 Summary

In this chapter, a dynamic bandwidth allocation algorithm is presented, and evaluated to be advantageous in terms of equitable bandwidth accessability and overall cell throughput improvement.

Chapter 5

A Performance Measurement System
For TCP/IP Based Multimedia Ethernet LANs

5.1 Introduction

The Ethernet is an examples of local area networks (LANs) designed for local communications of narrow band traffic, typically data. Little is known about the performance of these types of LANs when real time traffic, such as slow motion video (compressed and variable-rate coded, with nominal rate, say 56 Kbps) and voice, mixed with conventional data traffic, is carried in the network.

Due to the increasing demand for video services, such as teleconference and picture phone, and the need for integration of data, voice and video services in LANs, we will also investigate the performance evaluation tools for LANs in such a multimedia environment in our last part of the study. A performance measurement system will be analyzed and designed for operation on the TCP/IP based PC LAN (Ethernet) in the Local Area Networks Laboratory.

5.1.1 Performance Measurement

Performance measurement, or empirical modeling, is one of the three major techniques (i.e. analysis, simulation and measurement) for performance evaluation. It does not require a model of the system. Rather, it performs measurement on the system itself, either under the real operating conditions, which is referred to as dynamic empirical modelling, or under hypothetically generated conditions, which is referred to as static empirical modelling. The latter approach allows the evaluator to have direct control of the network work load, and to avoid certain disturbance on the network real operating conditions, which may occur in the former approach, to ensure more accurate measurement. A disadvantage of empirical modelling is the high cost that may incur in building the prototype for measurement if the system being modeled is not available.

5.1.2 Ethernet Local Area Networks

As a popular example of local area networks (LANs), an Ethernet comprises a number of stations connected by a coaxial cable called channel [UNGARO]. A station connected at the cable with a cable tap and transceiver cable to the station's controller (i.e. the Ethernet controller). In general, stations may be a combination of various types of PCs, workstations, minicomputers, file servers, printers, and other devices. Cable segments greater than 500 meters may be connected by signal repeaters.

Data transmission is bit-serial at a rate up to 10 Mbps. Data is transmitted in packets called frames. A frame comprises header, data and CRC (cyclic redundancy check) fields. The header contains the addresses of the source and destination stations, and the data length. The data field is followed by a CRC code for detection of transmission errors.

Ethernet uses a distributed access protocol called Carrier Sense Multiple Access with Collision Detection or CSMA/CD. A station with a message to transmit listens to the channel and, if the channel is free, initiates its transmission. During transmission, the station continues to listen to the channel; if another station is transmitting, a collision occurs. When a station recognizes that a collision has occurred, it immediately abandons its transmission of data, transmit a jam message to ensure that all other stations will recognize the collision, and schedules its retransmissions attempt after a random delay called a backoff delay. An Ethernet LAN has been installed and running in the LAN Lab., and is available for experiment.

5.1.3 TCP/IP Internet Protocols

TCP/IP (Transmission Control Protocol/Internet Protocol) is a collection of protocols which support host-to-host communication for hosts connected to any of a number of heterogeneous networks and is thus referred to as an Internet Protocol Suite [COMER] [DAVIDS]. TCP provides services at the Transport Layer and IP provides services at the Network Layer. These services are augmented by application-like services provided in the higher layers. The unit of transfer is called a segment. Segments are exchanged to establish connections, to transfer data, to send acknowledgements, to advertise window size, and to close connections. Since TCP makes very few assumptions about the underlying network, it is possible to use it over a single network like an Ethernet, as well as over a complex network. TCP/IP internet protocol allows intercommunication of hosts, PCs, terminal servers and other computing facilities on one or more types of LANs or WANs. It also allows users with widely varying computer equipment to choose from among a number of hardware and software components from several networking vendors and roll their own integrated system.

5.2 Modeling of Packet Data, Voice and Video

5.2.1 Probabilistic Models

A voice source alternates randomly between talkspurt and silence. The active time for voice is on the order of 30%-50%, the average active-inactive cycle on the order of a second, and an average call being about 3 minutes. The message stream from a single voice source is digitized by sampling at, say, 8000 samples per second. Then it is quantized, typically at 8 bits per sample, to produce a PCM bit stream. A voice encoder operating at V, say, 1,600 bits/second combines bits from several samples, after possible data compression, to produce a segment of length L_V, say, 400 bits. Thus, every $T_V=L_V/V$ second the voice encoder outputs a voice segment during talkspurts (e.g. T_V=20ms). The number of talkspurts in a session is assumed as geometrically distributed with mean N. We also assume that successive talkspurts and silences form an alternating renewal process, i.e., all these time intervals are independent with talkspurts being exponentially distributed with mean L_t, and silences being exponentially distributed with mean L_s. The values of the mean talkspurt and silence durations are experimentally determined as L_t=1.366 seconds, and L_s=1.802 [BRADY]. The mean number of talkspurts per session is N=5. With 32 kbps ADPCM coding and output rate

18.2 segments/sec. a voice segment (fraction of a talkspurt) has the size 1760 bits.

A data source generates messages in a Poisson manner with arrival rate λ_d, and the length of these messages is geometrically distributed with mean typically L_d=1 Kbits.

The modelling of the burstiness of video source is important and also complicated for performance evaluation of a network carrying video traffic. A discrete time, first-order Markov process is commonly employed in current literature [FUJII] [HUANG] for modelling a video source. It approximates a single video source by an autoregressive (AR) process and characterizes time domain behavior of video signal by autocorelation. AR process is one of the simplest models that takes into consideration correlation of the sequence. The bit-rate of the kth frame x(k) can be written as follows.

$$x(k) = x'(k) + E[x(k)]$$

with

$$x'(k) = cx'(k-1) + e(k)$$

where E[x(k)] is the average bit-rate, c ($0<c<1$) is the model parameter which is one measure of how fast the bit-rate x(k) goes back to the nominal rate x bar), and e(k) is a gaussian random process. It is understood that video sources without scene changes or large motion can be modeled well by the AP process. If the entire video is built up by several typical scenes, the video source can be modeled by multiple AP processes [FUJII] [HUANG]. Amount of information in bytes for a video frame can be derived from the above formulas scaled by the frame sampling rate. As an approximation a zeroth order AP process can be employed to model a single video source [HUANG], i.e.,

$$x(k+1) = cx(k) + e(k)$$

where $e(k)$ is either *gaussian* or double exponential.

5.2.2 Performance Measures

In summary, the performance measures from the network point of view include link utilization, overall average delay, throughput, delay distribution, packet loss rate, etc. [SPANOL] [IBE] [LIU] [JOSEPH]. For individual type of traffic different measures may apply. We are interested in the following for our system.

Video: Line clipping probability, average delay.

Voice: Average delay, packet discard rate, call set-up time.

Data: Average delay, maximum delay, error rate.

The line clipping probability can be determined by identifying lines that are delayed beyond the maximum that can be compensated by the network.

5.3 Performance Measurement System

The system consists of two asymmetric subsystems. One subsystem consists of a traffic generator, a dispatcher, a dummy traffic generator and a secondary (voice) receiver, while the other consists of a receiver, a data analyzer, and a secondary voice generator, etc.. Each subsystem runs on one network station. The system configuration diagram appears in Fig. 5.1.

Segments of data, voice or video are generated at a rate and with the characteristics specified and controlled by the user via the user interface. When a segment is generated it is time-stamped (event time), primitives of TCP/IP Socket Library are invoked to send the segment across the network. Another time quantity may be identified, i.e. the time the segment actually enters the network link (transmission time). The traffic receiver on the other host receives the segment sent from the sender, if not lost, time-stamps it (receipt time), and extracts such information as traffic type and time tags and forwards it to the data analyzer on the same host. Network performance parameters, such as delay, variance, packet loss rate, throughput and link utilization for various types of traffic, can be calculated in the data analyzer, and the results are displayed at real time and saved in DOS files as well.

In measurement real network traffic is replaced with the dummy traffic from the dummy traffic generator, which is integrated with the traffic receiver and data analyzer on the same host. It puts out traffic at a rate specified and controlled by the user. This static modelling strategy gives user direct control of traffic volume on the network, making the experiment reproducible for validation and comparison of perforation measures under different traffic conditions.

5.3.1 System Design Issues

The measurement system is designed to run on, and measure the performance of an Ethernet (EXCELAN PC LAN) [EXCELA], with portability as one of the design goals.

5.3.1.1 System Development Tools and Environment

The Ethernet in the LAN Lab is a TCP/IP based LAN (IEEE 802.3 standard) with three ims-286 AT compatibles (Intel 80286 8 Mhz CPU and EXOS 205T Intelligent Ethernet Control Card) and a Leading Edge XT compatible running under the MS-DOS operating system, and a traffic monitor called LANalyzer (EX 2000) which displays basic information about the network traffic conditions and network resources utilization. The network scheme is depicted in Fig. 5.2. The TCP/IP software provides for an application programming interface called Socket Library

Application Programming Interface (SLAPI) which consists of a set of C-based routines. Neither the PC SimscriptII.5 nor the WorkPlace TCP/IP SLAPI supports mix language programming, the measurement system software will thus be coded in language C (Microsoft C version 5.1).

5.3.1.2 Traffic Generator and Dispatcher

The traffic generator comprises one module for generating each of the three types of messages (i.e., a data generator, a voice generator and a video generator), and a time tag module for recording the times the message segment (packet) is generated and subsequently sent, all driven by a dispatcher module which determines the type of message generated, parameters to associate with the message and proper Socket Library routines to invoke.

The data and voice sources are modeled as Poisson processes and video as an autoregressive process as described in section 5.2.

The system operation flowchart is given in Fig. 5.3. The algorithm of message generation in the sender (traffic generator and dispatcher) can be summarized as follows.

1. Input and validate traffic model parameters.

2. Generate file, voice, and video packets and store them in the queues. These packets are stamped with the packet generation time called "gen_time", at which they should be dispatched.

3. Establish a logical connection with the receiver through the Ethernet bus.

4. Repeat steps 5 to 7 below if the measure session time is less than that required.

5. Dispatch a file packet if current system time "now" equals the gen_time of a packet in the file queue. Discard those packets in the file queue whose gen_time are well past the present time. Generate new packets so that there are always so many packets stored in the file queue.

6. Dispatch a voice packet if now equals the gen_time of a packet in the voice queue. Discard those packets in the voice queue whose gen_time are well past the present time. Generate new packets so that there are always so many packets stored in the voice queue.

7. Dispatch a video packet if now equals the gen_time of a packet in the video queue. Discard those packets in the video queue whose gen_time are well past the present time. Generate new packets so that there are always so many packets stored in the video queue.

8. Disconnect with the receiver after the session has completed.

As seen from statements 5 to 7 above, the sending station has to sequentially check the file queue, the voice queue, and the video queue to see if any packet needs to be sent, and to discard the out-of-date packets. The queues will be implemented as circular, and can accommodate 10 elements each. This type of implementation allows for quick dispatch of data when the time comes. However, new packets have to be generated and stored in the queues while the measurement is proceeding. This overhead can cause significant additional packet loss/delays if the Ethernet bus is under heavy load. Note that the algorithm has additional problems if the 10th element in the queue is also out of date. One can avoid this situation by increasing the size of the queues, but it will introduce further time delays into our measurement results. A remedy would be to let the program alternate between packet generation and measurement sessions.

5.3.1.3 Traffic Receiver and Data Analyzer

On receiving a packet the receiver time-stamps it, records received message sizes and sequence numbers, extracts the useful header information such as packet generation time, original size and sequence number, and packet send time, and passes them to the data analyzer.

The analyzer contains a module for calculating each type of the performance indexes listed in section 3.2. Results are stored as DOS files and displayed at real time in graphical form as well.

5.3.1.4 Dummy traffic generator

The LANalyzer is capable of generating multi-stream dummy traffic at fixed rates, and will be used for this purpose in addition to serving as network traffic monitor.

5.3.1.5 User interface (I/O)

The interface comprises a terminal input routine, and an output display routine for graphical presentation and plotter.

5.3.1.6 Secondary Voice Generator

The secondary generator acts as a callee with respect to the primary voice station (the caller). It generates a voice message (a talkspurt) with geometrically distributed size whenever an ending talkspurt from the primary station is received. It stays silent otherwise. Note that the two stations may both be talking at some instances.

5.4 Remarks

The system was implemented using the skeleton system approach. Evolution is underway. Measurement sessions design, data collection and reduction, and performance assessment based on gathered data need to be excised in using the system. Further information can be found in the system design and implementation documents [WANG,4] [WANGSA] and [YANG].

CONCLUSION

The integrated communications services have very different traffic characteristics from those of conventional telephone calls. The video communications require wideband communications channels, while teletex services necessitate a narrower bandwidth than existing telephone services; the average holding time of teleconferences and terminal to computer data transfer is usually rather long, while that of message services short. Some services such as image and graphics may generate large bursts of data in the network. In this dissertation we present the performance evaluation of various types of integrated services networks, specifically, heterogeneous circuit-switched networks, ATM packet switched broadband networks and multimedia LANs.

The problem of trunk congestion in circuit-switched networks concerns the network link blockings under heavy load conditions. In Chapter 2, several types of routing and controls for heterogeneous circuit switched networks with two classes of traffic of unequal holding times were introduced. A performance evaluation was made, based on a fully connected symmetrical network model. It was shown that the trunk reservation technique can be applied to alleviate trunk congestion caused by excessive alternate routing at heavy loads. It was further shown that the restricted access and preemptive priority control can be used to enforce certain routing priority ordering over two classes of traffic, keeping call blocking probability of one class of traffic at certain level while minimizing that of the other class.

In an ATM, admission control and enforcement functions must be provided to manage access to bandwidth resource and to prevent the network from unwanted congestion. User-initiated bandwidth variation during a virtual connection must be under control to guarantee the level of performance for the connection blockings and to avoid possible performance instability. In Chapter 3, the impact on the ATM network performance due to the bandwidth variation during a session is analyzed, based on a simplified model described in Section 3.2. It is shown that as more calls intend to cross-over from a lower bandwidth destination to a higher one the network suffers from a higher connection blocking. The crossover availability declines as more calls cross-over and increase the bandwidth used. Slight increase of overall throughput is observed, owing to the fact that the network carries more crossover calls with the increased bandwidth. It is also shown that limitation can be placed on the amount of bandwidth that the crossover calls can occupy to protect the constant-bandwidth calls and to keep the network connection blockings below certain level. In addition, we studied the admission control and bandwidth management. In Chapter 4, a dynamic bandwidth allocation algorithm was presented, and evaluated to be advantageous in terms of equitable bandwidth accessability and overall cell throughput improvement.

The Ethernet is an examples of LANs designed for local communications of narrow band traffic, typically data. Little is known about the performance of these types of LANs when real-time traffic, such as slow motion video (e.g., compressed and properly coded, with 56 Kbps nominal rate) and voice, mixed with conventional data traffic, is carried in the network. In Chapter 5, we presented the analysis and design of a software system for measuring the performance of the Ethernet LAN in such a multimedia environment. The system has been implemented as a skeleton system.

LIST OF FIGURES

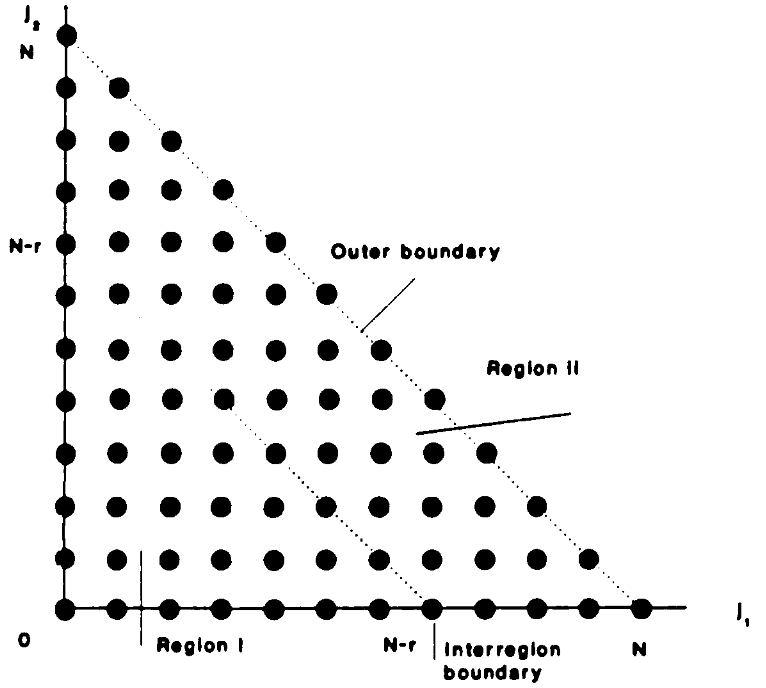


Fig. 2.3 (a) State Space - Trunk Reservaion Scheme

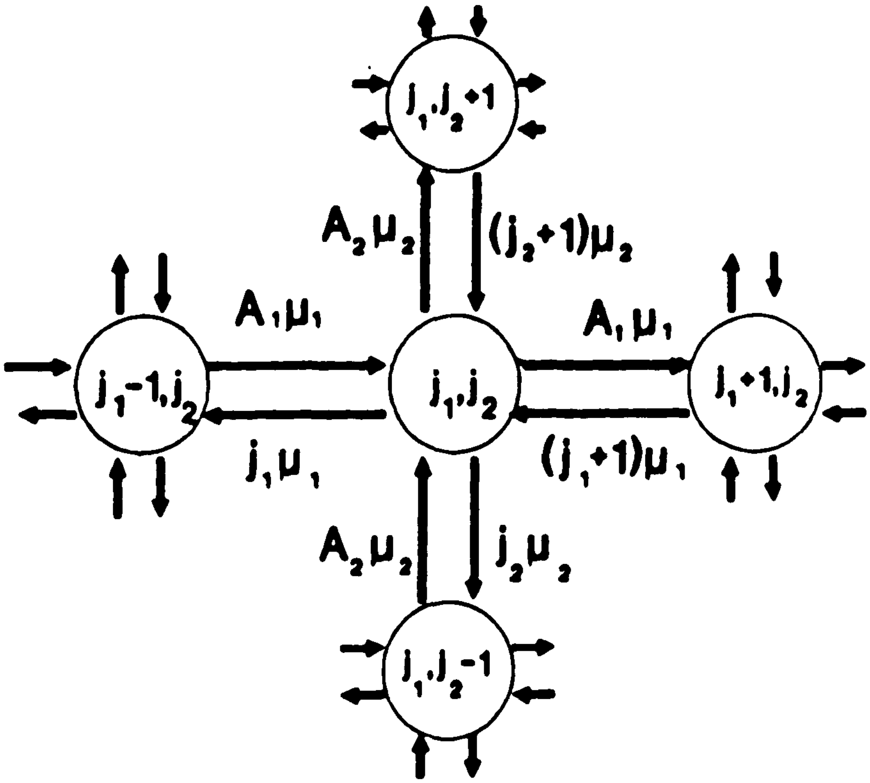


Fig. 2.3 (b) State Transition - Region I

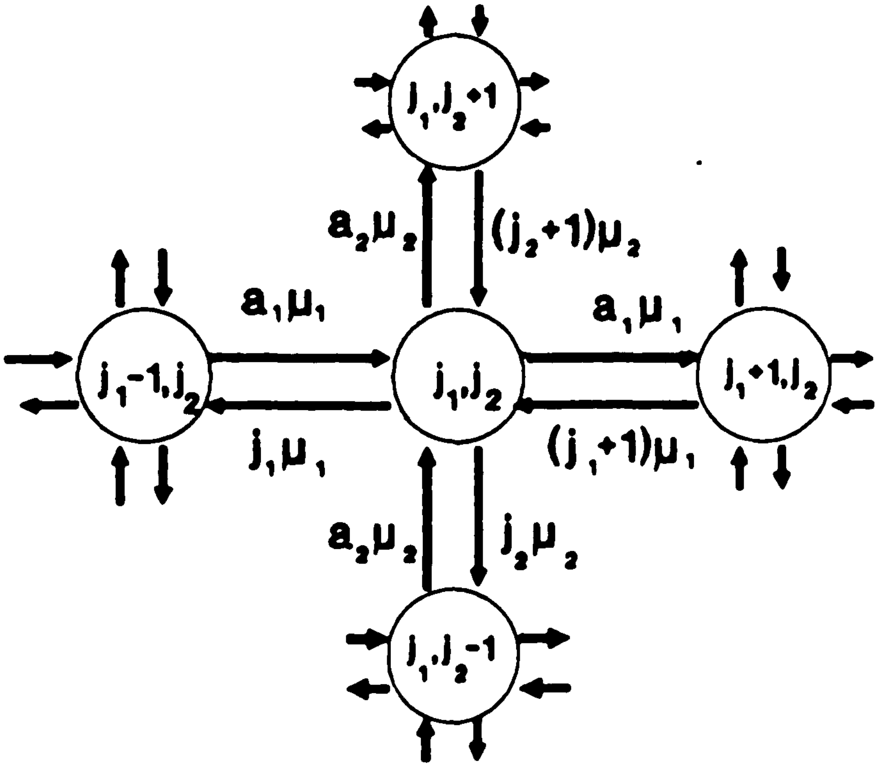


Fig. 2.3 (c) State Transition - Region II

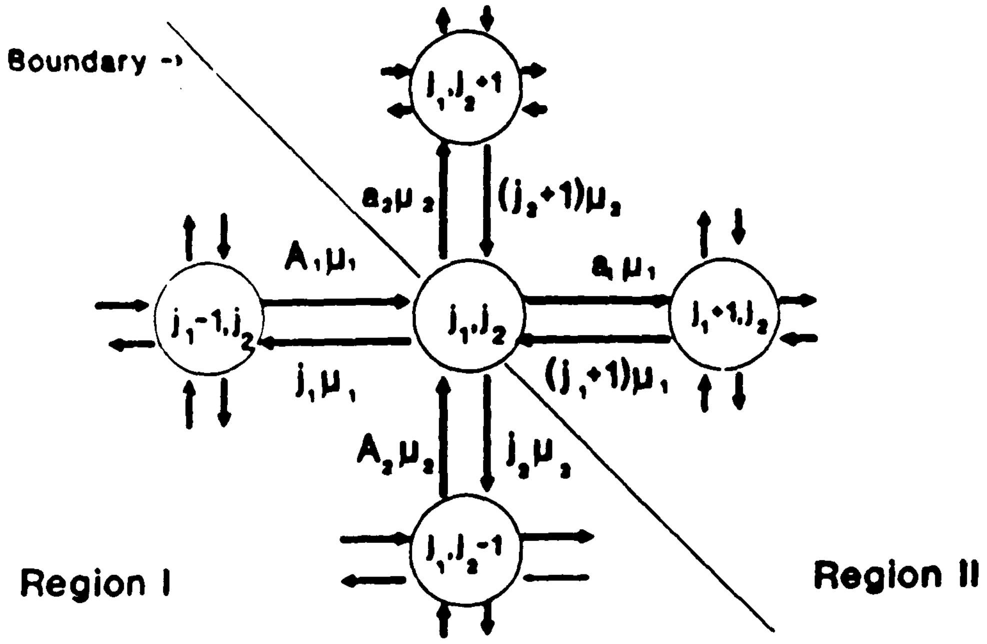


Fig. 2.3 (d) State Transition - Interregion Boundary

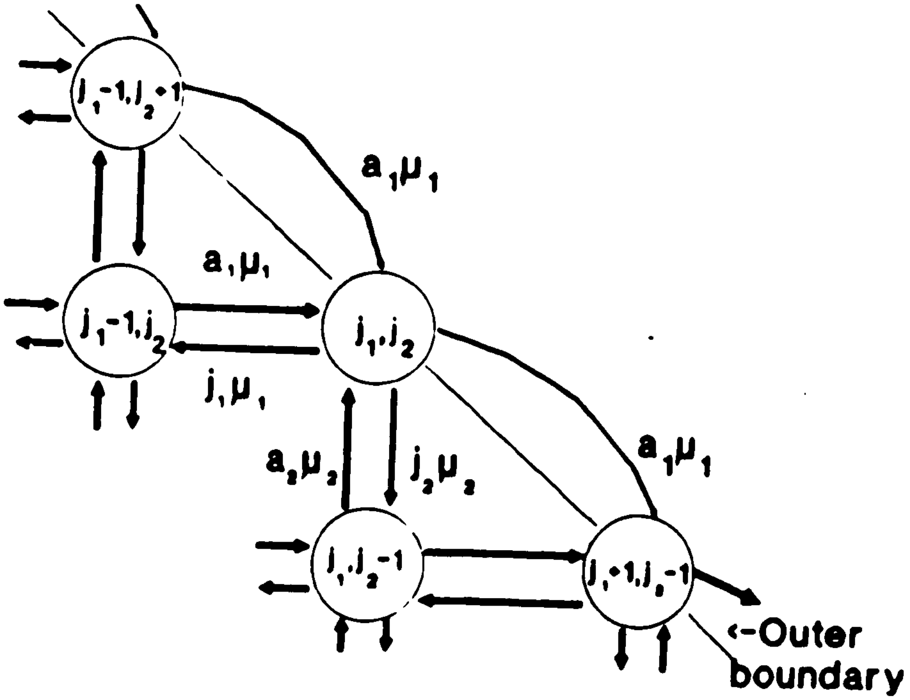


Fig. 2.4 State Transition - Outer Boundary

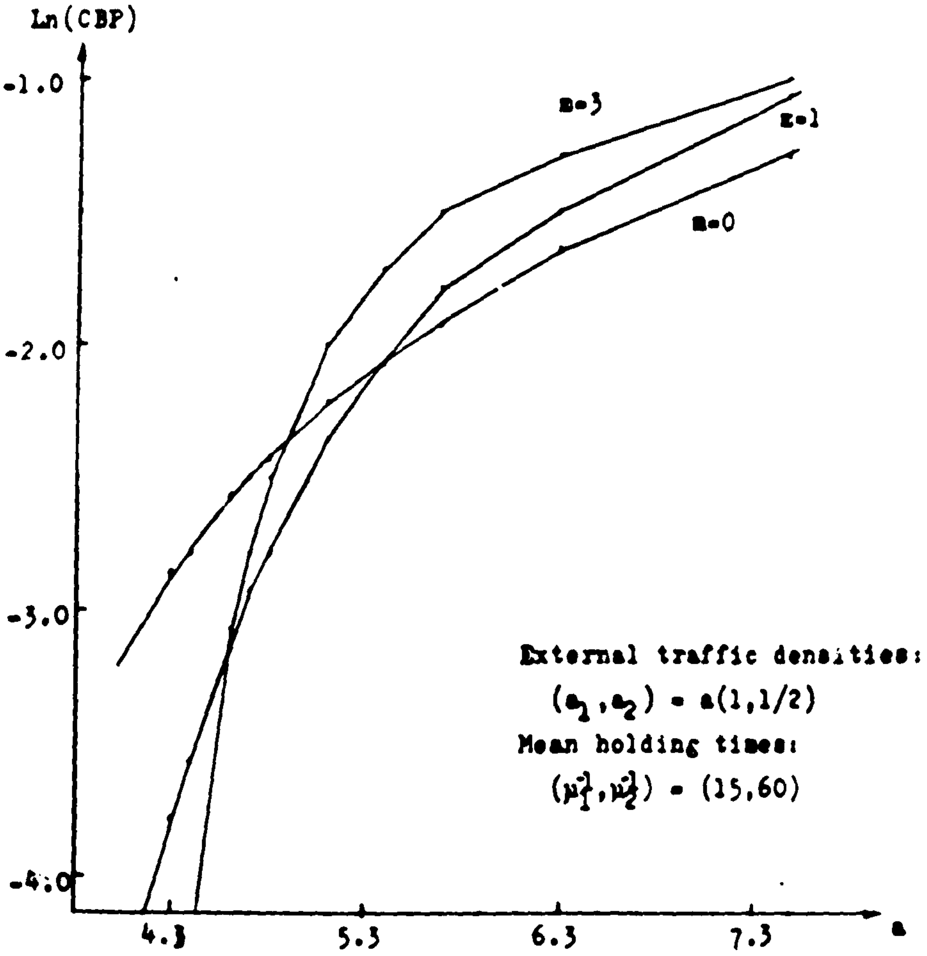


Fig. 2.5 (a) Ln(CBP) vs. external traffic parameter a.
Example for direct routing and alternate routing.

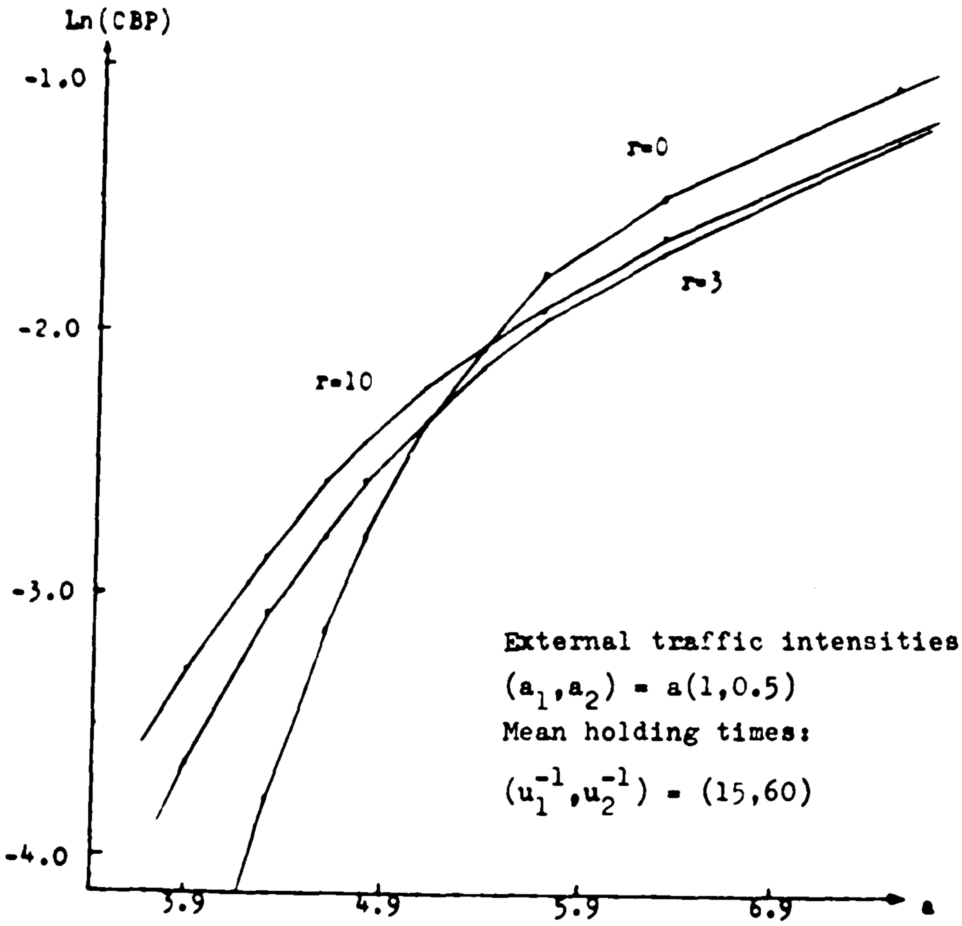


Fig. 2.5 (b) Ln(CBP) vs external traffic intensity a

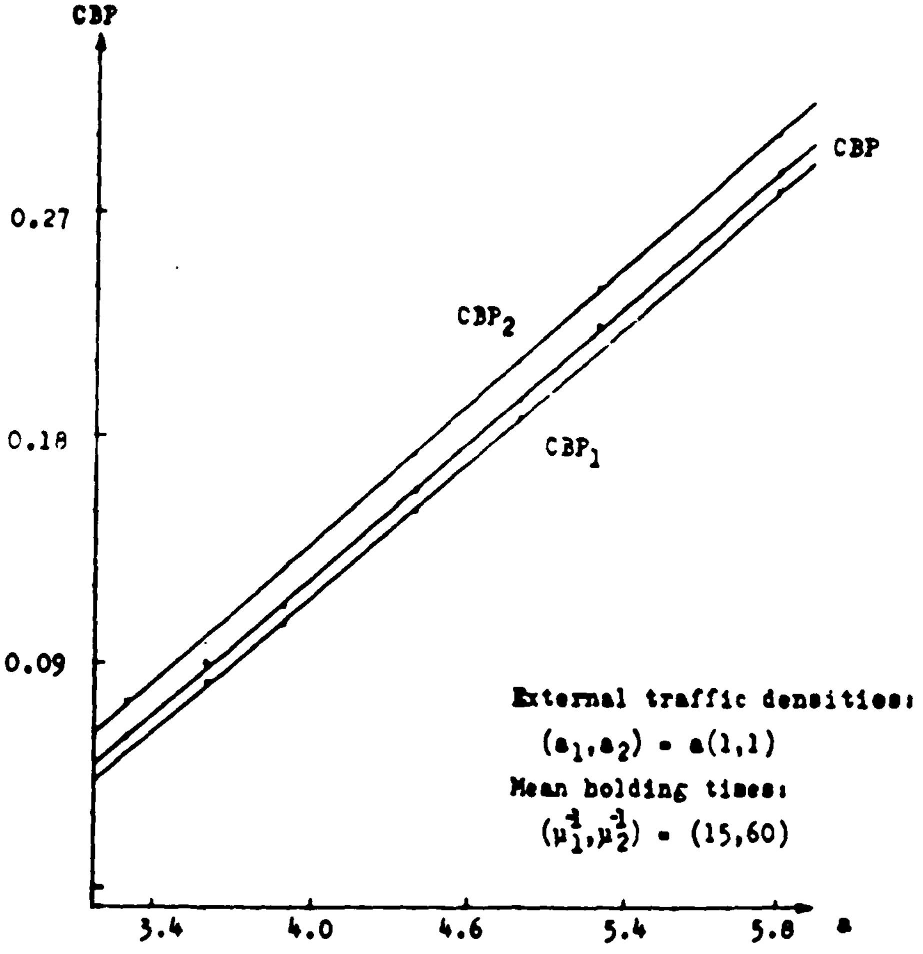


Fig. 2.5 (c) Call blockings under restricted access control

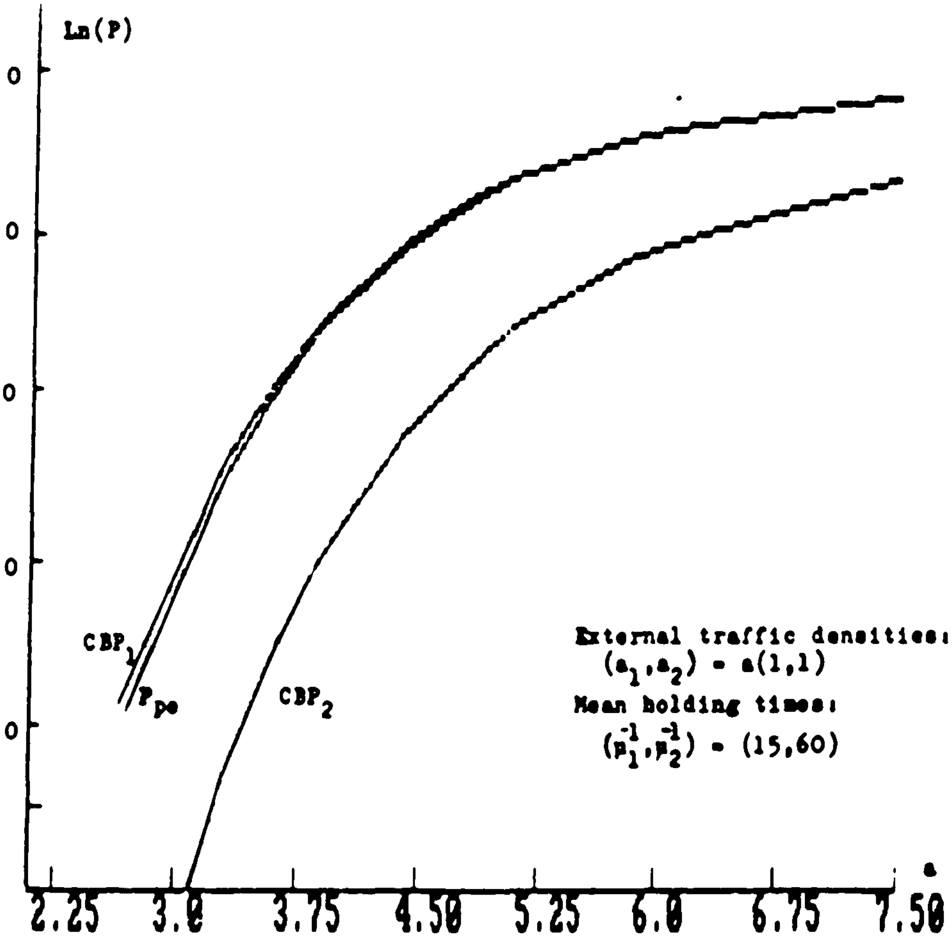
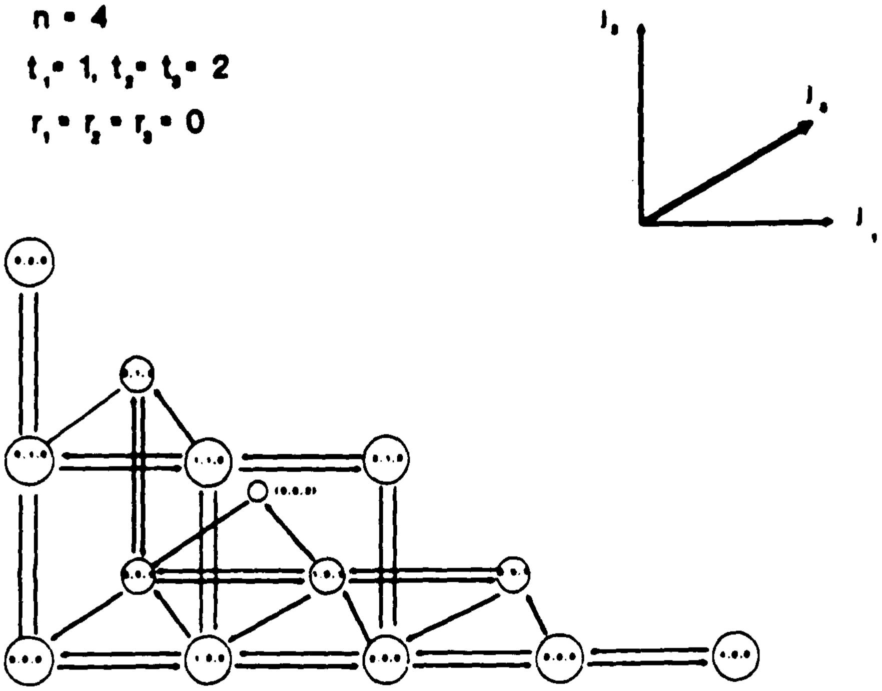


Fig. 2.5 (d) Call Blocking and Preemption Probabilities (logorithmic) vs. External Offered Loads - Restricted Access/Preemption.



$n = 4$

$t_1 = 1,\ t_2 = t_3 = 2$

$r_1 = r_2 = r_3 = 0$

Fig. 3.1 State Space - Example

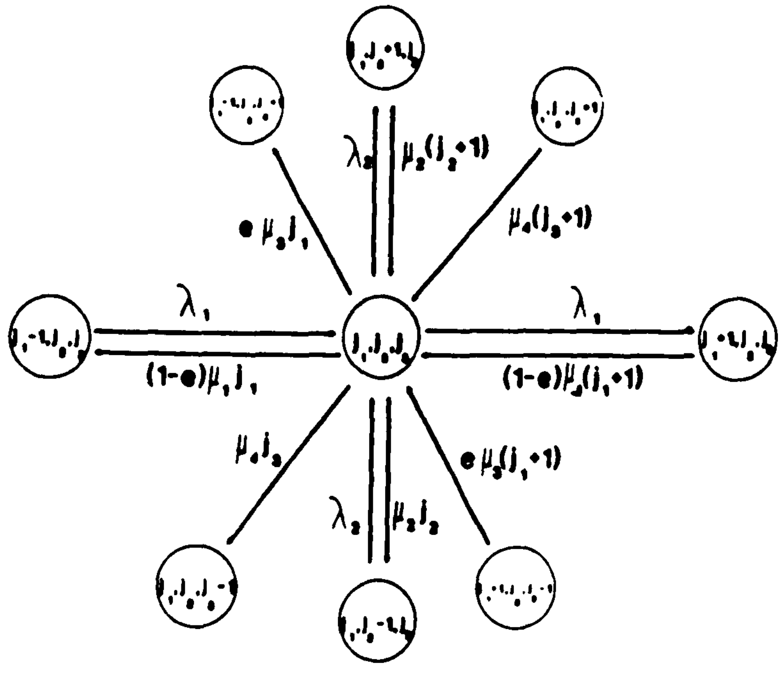


Fig. 3.2 Typical State Transitions

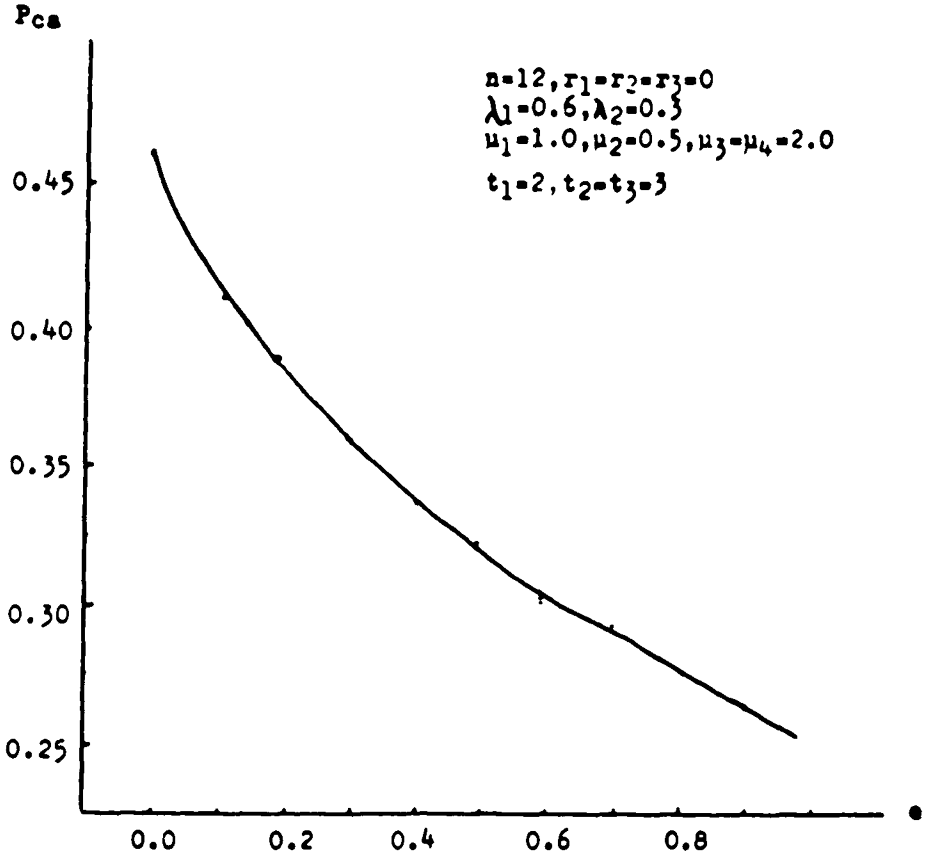


Fig. 3.4 (a) Crossover Availability vs Crossover Rate

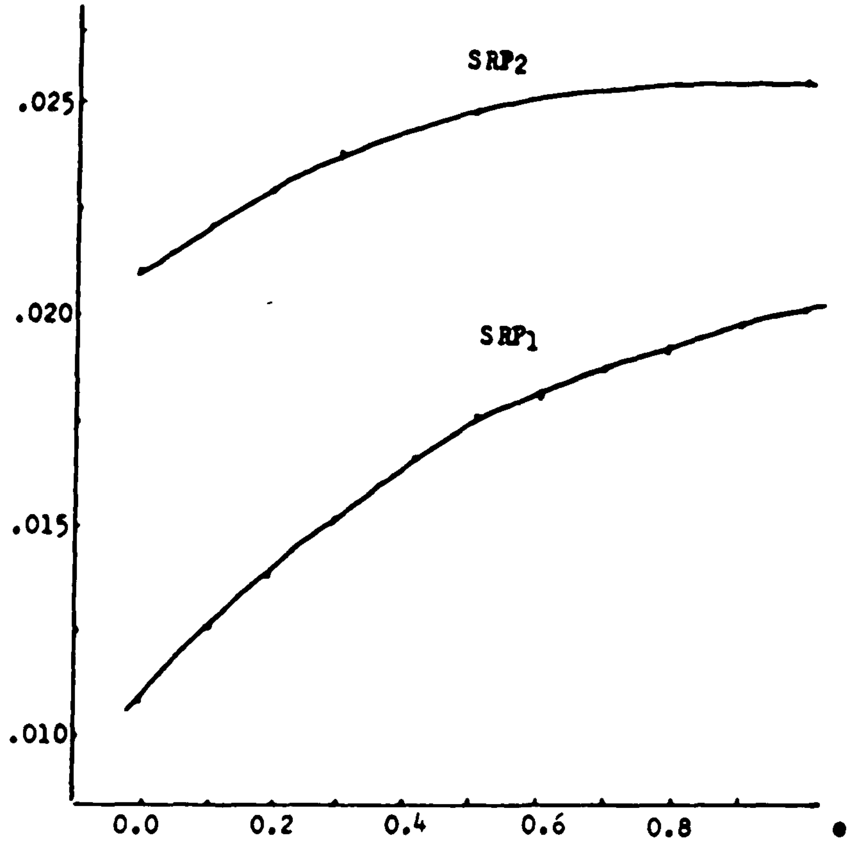


Fig. 3.4 (b) Connection Request Blockings vs e

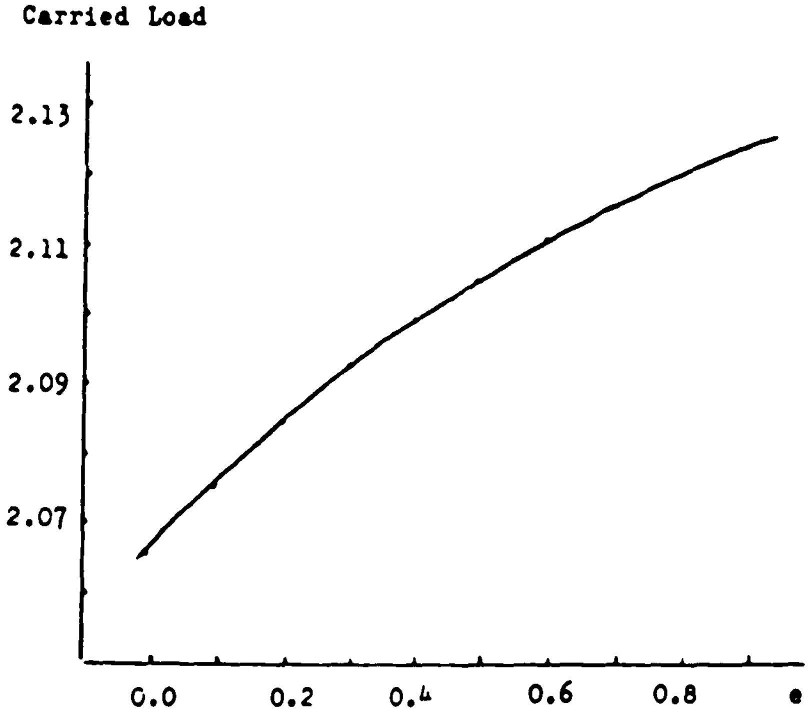


Fig. 3.4 (c) Throughput (weighted) vs Crossover Rate e

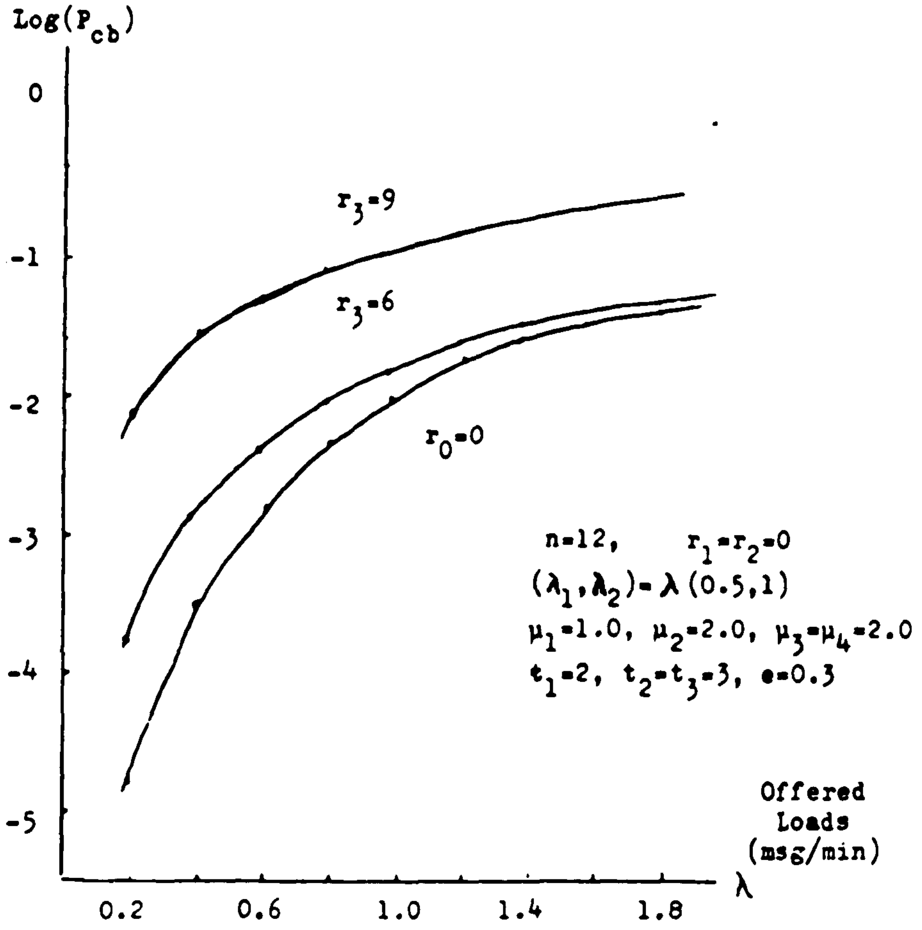


Fig. 3.4 (d) Effect of Crossover Limiting Parameter r_3

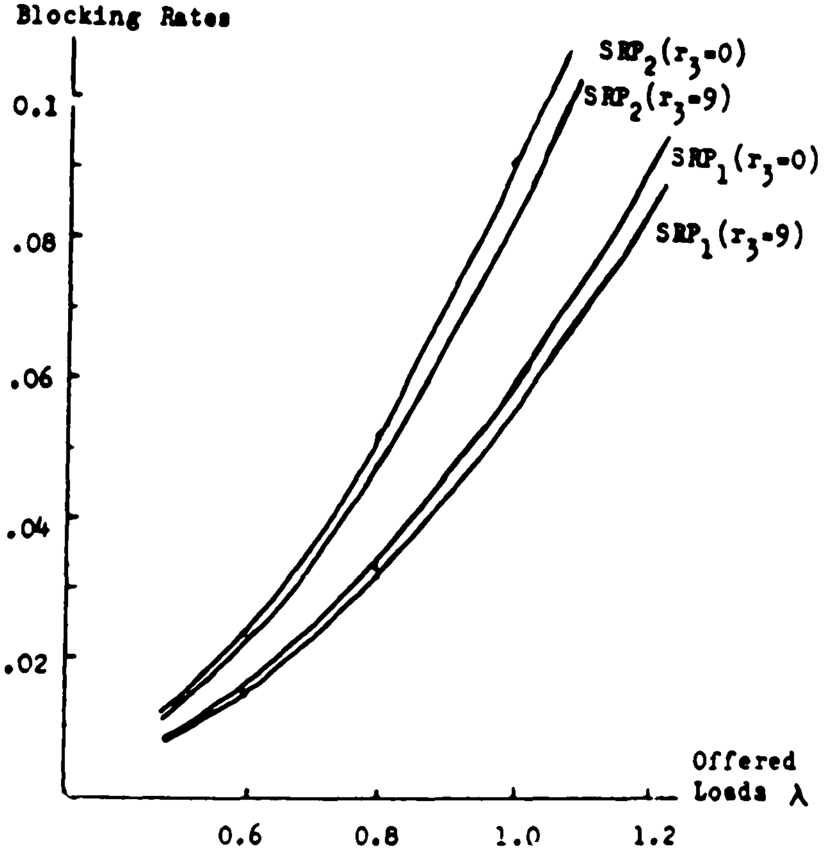


Fig. 3.4 (e) Effect of r_3 on Virtual Connection Blockings

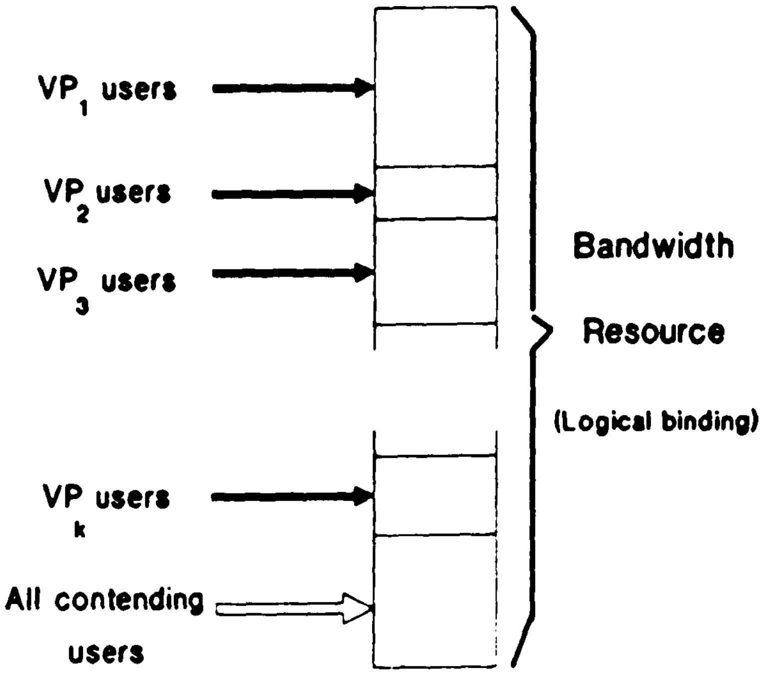


Fig. 4 Bandwidth Allocation Scheme

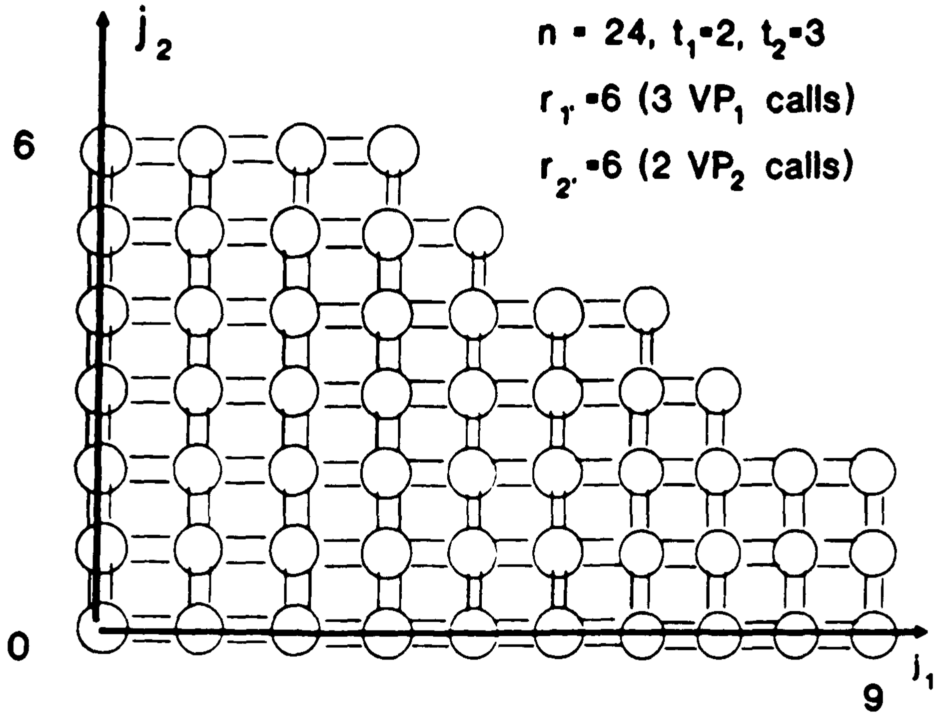


Fig. 4 (a) State Space - Example

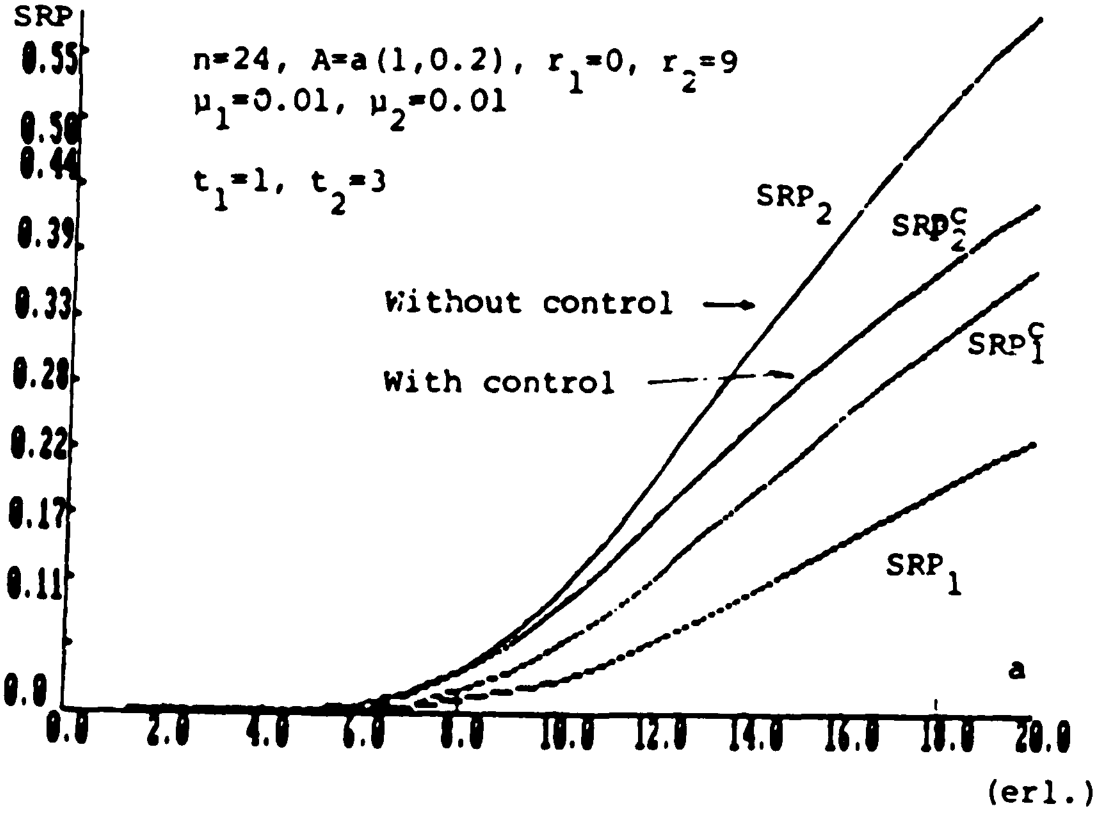
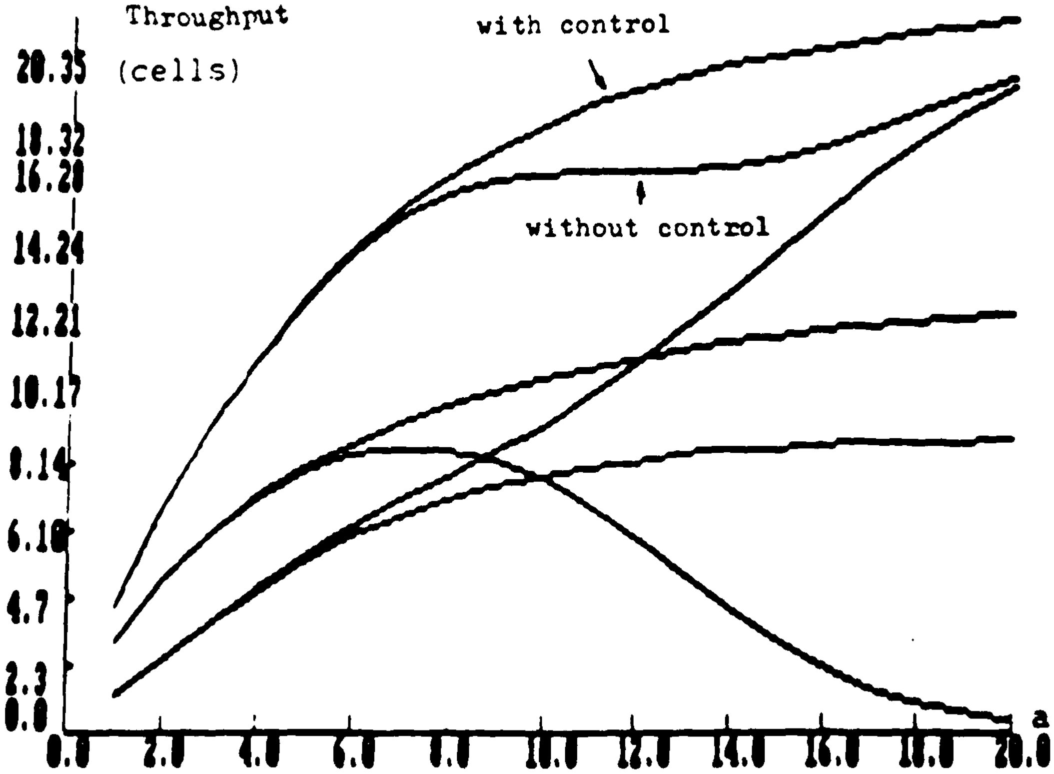


Fig. 4.1 Service Request Blocking Rate



$n=24, A=a(1,0.2), r_1=1, r_2=15, \mu_1=\mu_2=0.01, t_1=1, t_2=15$

Fig. 4.2 Overall Throughput

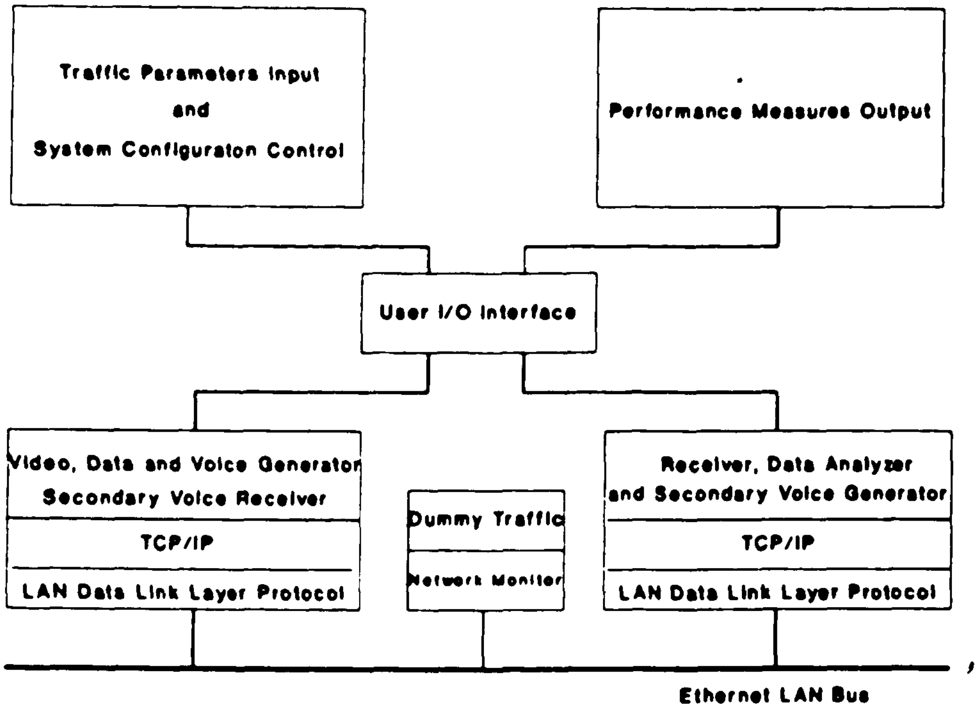


Fig. 5.1 Performance Measurement System Configuration

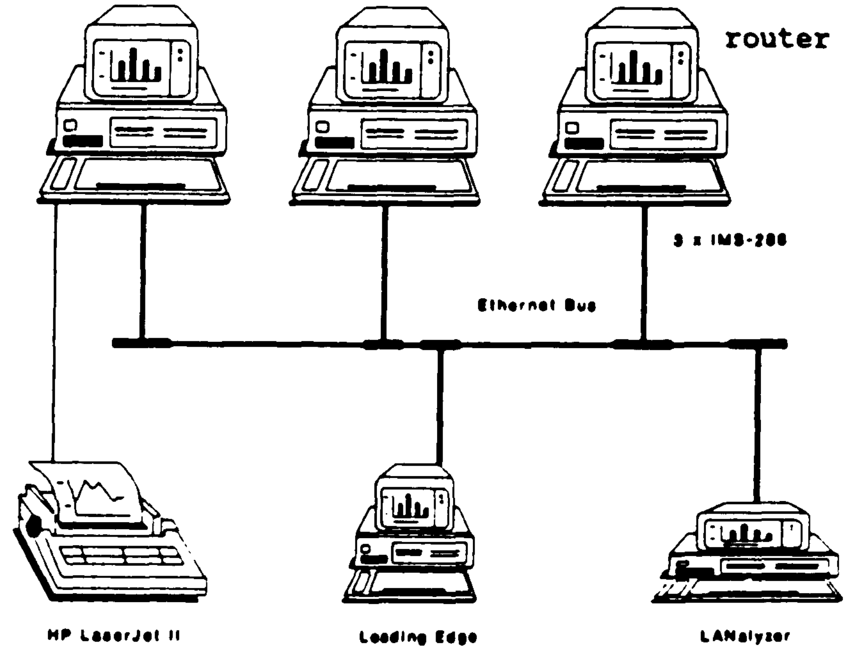


Fig. 5.2 Ethernet LAN System Configuration

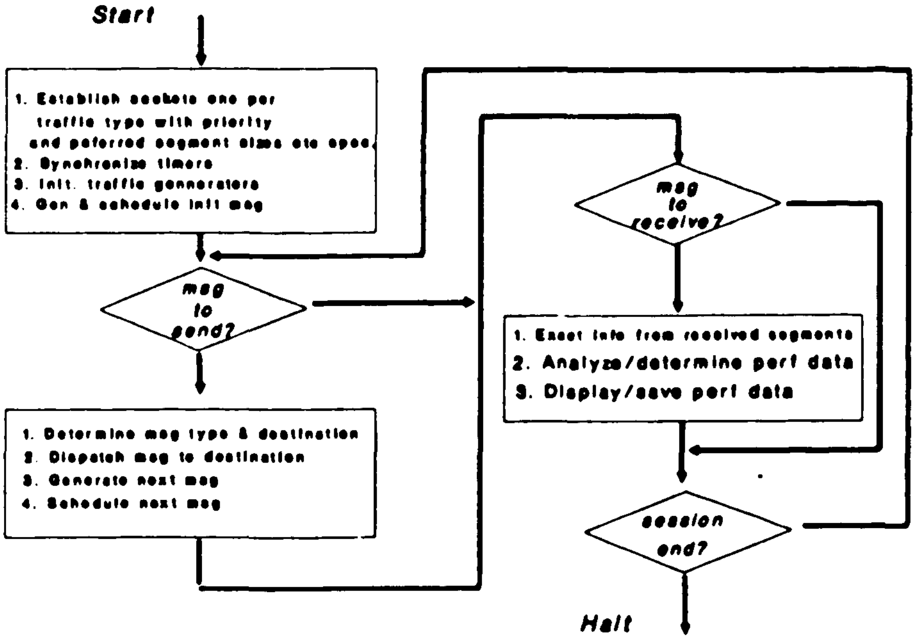


Fig. 5.3 Measurement System Flowchart

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