Proceedings of the 1985 Winter Simulation Conference
D. Gantz, G. Blais, S. Solomon (eds.)

A SIZING AND TIMING ANALYSIS OF AN INTEGRATED COMPUTER NETWORK SIMULATOR

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This research addressed the performance characteristics of a computer-communication network that integrates both voice and data traffic on the backbone of the network. Network performance was analyzed using an event-driven FORTRAN simulation model. The model was based on a circuit-switched communications subnet whose trunk lines carry both voice and data traffic simultaneously. A sizing and timing analysis is accomplished by collecting empirical performance data for a variety of system input parameter combinations. The data was analyzed to determine simulation running time needed to reach steady state and to size the input parameter ranges that correspond to feasible network performance.

INTRODUCTION

The recent telecommunications explosion is evidence that computer-communication networks are now providing economical and reliable computer services to a variety of classes of users. The power of computer-communication networks lies in the fact that they provide users shared access to such specialized computer resources as distributed data bases, application software, and hardware facilities. Such a network consists of a communications subnet (or backbone) together with the facilities needed to tie into the backbone. The subnet is comprised of communication processors (nodes) and trunk lines (links) which interconnect the nodes. Within a computer-communication network are several layers of protocols (rules) which are needed to provide efficient, error-free information transfer from an originating node to a destination node.

Classification of a network generally depends on many factors. The topology or geographic configuration, the message routing strategy, and the major functions it performs are each possible criteria which may be used to classify a network. Additionally, another means of classifying a network is by the technology which is used to move information between the nodes of the subnet, namely, the switching discipline used [12].

Switching Techniques

There are three basic approaches to the interconnecting of computers [12]. In a circuit-switched network, a physical path must be set up between the source node and the destination node prior to the start of information transfer. This path may traverse intermediate switching points (nodes). Once the path is established, the end users may communicate over the path. This path remains dedicated to the users until their communications session is completed, at which time the switching equipment disassembles the path. The network links and switching equipment that were in use are once again made available to other users.

The second approach, message (or store-and-forward) switching, is more data oriented. In this discipline, a total set of information (the message) is sent from one user in the system to another user by establishing a link from the "sender" to some intermediate node. Once this node receives the message, it must store and log it for accounting purposes until it can be forwarded to the next node in the routing scheme. Once the message is stored, the node releases the communications link. This process is repeated from node to node until the message is accepted by the "receiving" node. An acknowledgment is usually sent back through the system.

Each of these approaches has advantages and disadvantages in terms of resource utilization, switching times, storage requirements at nodes, etc. Because of its low line overhead and minimal delay [7], circuit switching is particularly useful in applications characterized by a steady, non-bursty flow of information. The telephone networks of the United States are circuit-switched systems. Users do not have dedicated voice channels but compete for limited resources. Circuit switching has remained the most effective approach for voice users [21]; however, early attempts by data users to use circuit switching resulted in problems with switching delay times and circuit utilization.

Store-and-forward message switching with fixed routing was used in many commercial applications and was the major approach utilized by the Department of Defense (DOD) data networks (e.g., AUTODIN). Limited adaptive routing schemes allowed store-and-forward systems to better utilize their links; however, in many cases, link utilization was less than effective and the nodal storage requirements and message manipulation times (logging, storing, and transmitting) were often excessive. As a result, an improved switching philosophy, packet switching, emerged. Using packet switching, the system breaks a message into fixed-size "packets" (although some variable-length packet schemes exist). Each packet is theoretically transmitted from source to receiver over the first available pathway from each intermediate node. In effect, each node is supposed to choose the shortest independent path it can find for each packet. This approach has been shown to provide good circuit utilization and to improve response times and "network throughput". For a period of time the ARPANET [14,15], which was the primary driver of packet switching technology, became the mecca of networking decisions. However, as applications and research soon proved, the lack of packet flow control caused some disastrous problems, and many constraints were placed on packet-switched networks. For example, network users were required to reserve final destination storage space, and trace capabilities were added to packets. Nevertheless, the packet-switched approach has become a widely used switching discipline in data systems [15].

The Case for An Integrated Circuit/Packet-Switched Network

A newer switching approach involves integrated or hybrid switching. In this approach, voice users are handled with circuit-switched components and data users with packet-switched components. Furthermore, voice can and is being digitized, so voice, too, can be handled under a packet-switched scheme. Finally, switching technology has reached the point where it is no longer clear that switching delay is sufficient reason to disregard the circuit-switched approach [5].

Recent Defense Communication Agency (DCA) studies have shown the desirability of an all-digital, switched network which integrates voice, interactive and bulk data for the 1980's [17,20]. Several additional studies relating to information processing growth in the next few decades portend new services with substantially increased data flows [1,13,16,18]. As stated by Ross [18], "the spectrum of terminals requiring service is expected to range from 45 bps TTY terminals and 2400 bps vocoders to interactive graphic, digital facsimile, and slow-scan video terminals requiring rates in the tens and hundreds of kilobits per second". To facilitate the implementation of these classes of traffic, both the designers and managers of future computer-communication networks plan to use an integrated technique to move the data through the subnet.

Research Objectives

The overall objective of this research was to obtain and analyze empirical performance data from an integrated circuit/packet-switched computer-communication network. The performance data will be obtained by using a computer network simulation model [2], hereafter referred to as the simulator, which is capable of simulating an integrated computer network. In particular, the specific goals of the analysis were as follows:

(a) Using network performance data as a function of simulated time, determine the time at which steady state occurs for various combinations of network design factors and workload parameters.

(b) Using network performance data obtained from a variety of combinations of selected system input parameters, effectively bound the ranges of these input parameters so that the network performance corresponding to these ranges falls within realizable limits.

THE NETWORK SIMULATION MODEL

Description of the Integration Concept

The model developed in the simulator was based on an integrated circuit/packet-switched network consisting of the following major components (Figure 1):

A. Backbone Circuit Switch (CS) Nodes

B. Peripheral Packet Switches (PS)

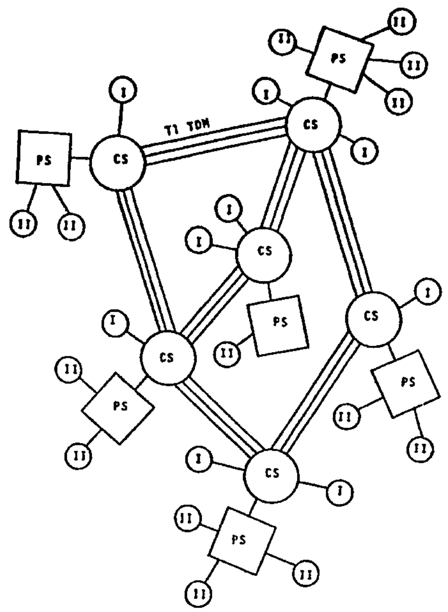

Figure 1. Integrated Network Configuration

C. Invariant Network Synchronous Time-Division-Multiplexed (TDM) Frame Switching Superstructure

D. Digital Network Using T1 Carriers and Digital Switching Nodes

E. Variable Subscriber Data Rates

F. Two Classes of Subscriber Traffic

1. Class I - Real-Time traffic that once started cannot be interrupted (voice, video, facsimile, and sensor)

2. Class II - The general class of store-and-forward (packet based) data, such as interactive, query/response, bulk, and narrative/message

The backbone CS network nodes and peripheral packet switches form the nucleus for a distributed computer-communication network in which the transmission of data/voice between any two subscribers is accomplished via adaptive sharing of the high capacity T1 link using the concept of the SENET (Slotted Envelope Network) [3] self synchronizing switching superstructure. This concept treats the available bandwidth on a digital link as a resource for which all forms of communication must compete. Using SENET the T1 link is synchronously clocked into frames of b time duration which are assumed invariant throughout the network. Using a competitive allocation methodology, each frame is structured to encompass a diversity of trunk communication rates in order to serve a multiplicity of diverse Class I/II subscriber traffic. The allocation implementation partitions the frame into several data slots (channels of variable time duration). The self synchronizing capability within each frame was implemented by assuming a Start-of-Frame (SOF) marker (a few bits at the start of each frame) to indicate the beginning of each a contiguous series of constant period frames.

Class I subscribers are directly terminated to circuit switches to preclude packetizing and any unnecessary routing overhead through packet switches. Each dial-up Class I connection results in a physical subscriber-subscriber connection for the duration of the call, or a system "loss", similar to a telephone dial-up process.

Co-located with the circuit switch nodes (although not a design requirement) are packet switches, terminating all Class II subscribers. The transmission of data between the packet and circuit switches is accomplished using Time-Division-Multiplexing on the network side, while the packet switch/subscriber interface is dependent upon the individual terminal hardware configurations. The packet switches are primarily responsible for management of packets between input terminals and the circuit switches, placing traffic on queues according to a regional routing policy, and performing connection initiation, circuit disconnect, and coordination with other packet switches, depending on the system loading.

The regional routing doctrine for each packet switch, coupled with virtual switch connections, reduces overhead and the traffic congestion problem. As traffic is entered into a packet switch from subscriber terminals, it is queued for the relevant destination packet switch. Unlike the SENET scheme, a circuit switch connection is then initiated/terminated by the packet switch on behalf of this traffic. A circuit switch connection can be established for a single transaction similar to an interactive communication, or on a multiple transaction basis if the traffic is bulk data, message/narrative traffic, or several users queued for the same destination packet switch. This routing scheme (1) insures minimal queue build-up within the backbone, and (2) enforces an end-to-end flow control strategy. Progressive alternate routing is used in this model. With this method each circuit switch node has a primary and an alternate path. If blocking occurs at some node during connection initiation, the alternate route is tried for route completion. If this connection fails, the transaction is either queued at the packet node or considered a system loss at the circuit node, depending on its class.

Description of the Queueing Model

For the integrated computer-communication network described, the numerous inter-nodal conditions and variables preclude any exact analytic solution. However by decomposing the network into nodal queueing models, the simulation model can be represented as a system of simple queueing models.

The traffic flow at each packet switch is described as follows:

1. Each Class II subscriber communicates with the packet switch via independent, Poisson transaction arrivals and exponentially distributed transaction interarrival times.
2. The message lengths (packets per message) are assumed to be geometrically distributed. This conforms to the study of multiaccess computer communications by Fuchs and Jackson [6].
3. Each packet switch can be thought of as a M/M/C system (Kendall notation) [8], with infinite storage.
4. Packets are placed on the packet switch queue and served on a first-come-first-served (FCFS) basis.

The traffic flow entering each circuit switch node originates at neighboring circuit switch nodes, or locally terminated Class I subscribers. Since all traffic entering from other terminated Class I subscribers see a physical connection, only the Class I subscribers enter into a serving mechanism process at the circuit switch node. These subscribers are assumed to possess Poisson arrival and exponential service distributions. Thus, the M/M/C/C queueing model suffices to represent this network model. Both the PS and CS queueing systems are impacted by channel availability, since the model policy forces data and voice subscribers to compete for the available slots. In summary, delays and queues at each node are approximated closely by M/M/C/ and M/M/C/C queueing models (Figure 2). Since each of these models is globally impacted by channel availability, the network simulation provides performance measures for end-to-end packet delay and voice call blocking.

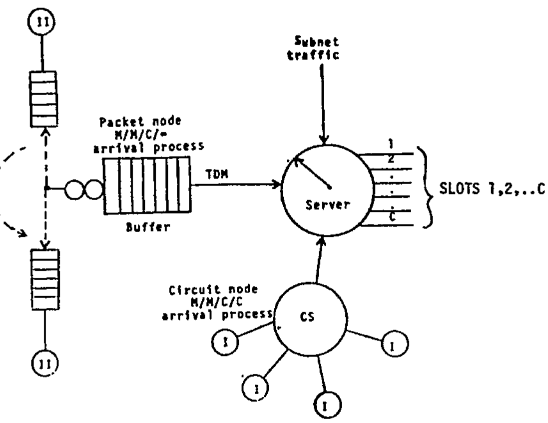

Figure 2. Network Nodal Queueing Model

Properties of the Model

Since the communications activity is centered around nodal activity, the model is node-based. There are three principle nodal tables - routing, channel, and queue tables. The routing tables are used to determine the output channel (link) a transaction will take through a given node to reach its destination. This table is created in the initialization phase of the program. Since each link is full-duplex (FDX), two indices, representing sender and receiver, are required; and each FDX link is modeled by two independent channels. Information stored and updated within various channel table entries at each node is used in the gathering channels, and various transaction oriented delay statistics.

An event table reflects network status disturbing events. For example, an event table may contain the time of the next arrival or departure at a node. The simulation is driven by event changes that occur at each node.

NETWORK PERFORMANCE ANALYSIS

Network Input Parameters

The simulator input consists of the following 16 parameters:

(1) Number of Nodes

(2) Number of Links

(3) Number of Time Slots

(4) Number of Slots Required by a Data Packet

(5) Frame Time

(6) Nodal Switching Delay

(7) Circuit Switch Arrival Rate (voice calls/min)

(8) Packet Switch Message Arrival Rate (messages/sec)

(9) Simulation Start Time

(10) Simulation End Time

(11) Packet Switch Saturation Level Indicator

(12) Voice Digitization Rate

(13) Packet Switch Buffer Size

(14) Voice Call Service Rate

(15) Number of Bits per Packet

(16) Average Number of Packets per Message

Due to the numerous combinations of system input parameters, the simulator can generate empirical data for multiple relationships between network topologies and workloads. "Topology" here refers to the manner in which the nodes are distributed and interconnected, as well as to the capacities of the interconnecting links. The 10-node network configuration in Figure 3 was deemed sufficiently complex to demonstrate the practicality of an integrated network based on an underlying circuit-switched subnet.

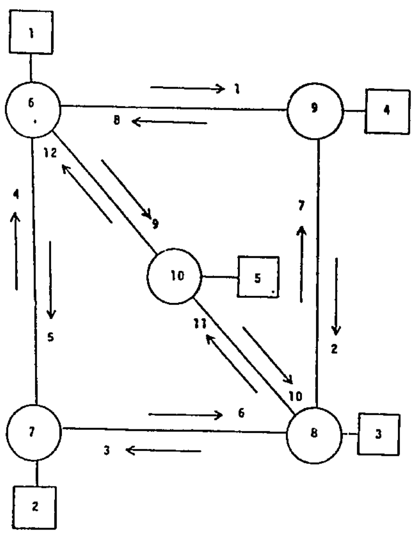

Figure 3. 10-Node Network Configuration

Link capacity, defined in terms of bits per second (BPS), can be seen to be a function of the input parameters. Specifically,

$$\begin{aligned}\text{Link Capacity} &= w(\text{bits/packet}) \cdot x(\text{packets/slot}) \cdot \\ &\quad \cdot y(\text{slots/frame}) \cdot z(\text{frames/ms}) \cdot \\ &\quad \cdot 1000(\text{ms/sec}) \\ &= 1000\, w\, x\, y\, z \text{ (BPS)},\end{aligned}$$

where w = Parameter 15
x = 1/Parameter 4
y = Parameter 3
z = 1/Parameter 5.

In our analysis, we have held parameters 4, 5, and 15 constant, while variations in parameter 3 (SLOTS) have been used to change link capacities.

The traffic load imposed upon the network is given by parameters 7, 8, 14, and 16. Parameters 7 and 14 are the voice arrival and service rates, where CS and SERV will be used to denote these two input parameters, respectively. If parameters 8 and 16 are multiplied, the packet switch arrival rate is obtained in the desired unit, "packets per second". We will use PS as the packet switch arrival rate in packets per second. It is assumed that the workload is uniformly distributed throughout the topology. That is, the arrival and service rates will be the same for each node of a given type. Thus, the input parameters of concern in this study (and consequently the ones that will be varied) are parameters 7, 8, 14, and 3. Hereafter, these four parameters will be denoted as CS/PS/SERV/SLOTS, noting that the value for PS is actually the product of parameters 8 and 16, where the value of parameter 16 remains fixed at 10. All other parameters have been held fixed at levels consistent with currently existing networks [3,4,9,11,16,18,19].

Network Performance Data

For each possible combination of input parameters, the simulator generates the following network performance parameters:

(1) Mean Packet Delay (MPD)

(2) Average Link Utilization (ALU)

(3) Packet Throughput Per Link (THR)

(4) Data Transactions in the System (i.e., average queue length (AQL))

(5) Fraction of Calls Blocked (BLK)

Specification of bounds on these performances parameters generally establishes a grade-of-service for the network. The simulator generates other statistics as well (e.g., packet switch loading and a frequency distribution of packet delay times), but this study concerned itself primarily with the five listed above.

Preliminary Analysis

Original estimates for a range of CS arrivals which results in feasible network performance for this particular topology were given as 0-20 calls per minute [2]. What constitutes "feasible" network performance is of course somewhat arbitrary, but for our purposes we have elected to use a mean packet delay of no more than about 10 seconds. Although arbitrary, this figure is realistic in the sense that any packet delayed over 10 seconds would in almost any packet-switched network certainly be "timed-out" of the network. Yet, on the other hand, 10 seconds is not so small as to stifle the sizing of the input parameter ranges.

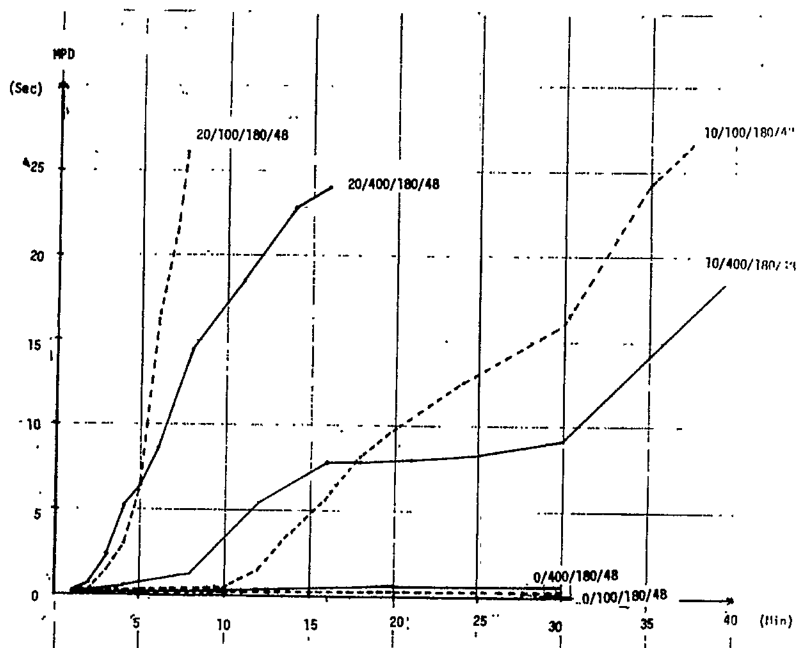

Figure 4. Mean Packet Delay (MDP) For 3X2 Experiment

A 3x2 mini-experiment was conducted to test the 0-20 CS range and to assess the relative performance sensitivities to the parameters CS and PS. The six runs corresponded to three levels of CS (0,10,20) and two levels of PS (100,400), where SERV=180 and SLOTS=48 were constant in all runs. The performance data from these runs clearly indicate that the CS rates of 20 and 10 both produce unacceptable performance. For example, Figure 4 shows the mean packet delay (MPD) as a function of time for all six runs. For CS=10 and CS=20, the MPD far exceeded 10 seconds, even prior to the system reaching steady state. Similarly, in both cases, the fraction of calls blocked was unacceptable in that more than 25% of all calls were blocked.

The determination of when the system reached steady state was made by observing how the performance parameters behaved as a function of time. Table 1, which is the performance data for the case 0/100/180/48, reveals how the performance data can indicate steady state. This data suggests that steady state has been reached as early as 15 minutes after the start of simulation.

Experimental Analysis

Now that the CS range has been pared to CS < 10, we proposed rotatable second order experimental design [10] in which only the center point and the axial points on the network loading side of the design center line are evaluated. The five levels for each of the four parameters were chosen as follows:

SYSTEM PERFORMANCE MEASURES FOR THE GIVEN INPUT PARAMETERS

NODES	LINKS	SLOTS	RATIO	SLOT TIME	NODE DELAY	CS ARRIVAL	PS ARRIVAL	MSG START TIME	END TIME	PACKET LOADING	VDR RATES	@ SIZE	CS SERVICE	PACKET SIZE	PACKETS PER MSG
10	12	48	3	10MS	50 MS	0MIN	10SEC	0 MS	1800301MS	99000	32000KBS	1800	180SEC	1000B	10

TIME INT (MIN)	MEAN PAC DELAY (SEC)	AVG LINK UTILIZ. (%)	AVG NO PAC PER LINK (DELTA)	AVG LINK THRUPUT (PAC/SEC)	DATA ARRIVALS PER INT	DATA DEPARTS PER INT	VOICE ARRIVALS PER INT	VOICE DEPARTS PER INT	MEAN Q LENGTH (TRAN)	FRAC OF CALLS BLOCKED
1	0.107	0.061	10567	176	301	291	0	0	5.0	0.0
2	0.107	0.064	11805	186	299	306	0	0	2.2	0.0
3	0.108	0.066	12085	191	305	306	0	0	1.8	0.0
4	0.108	0.066	11536	192	296	291	0	0	3.8	0.0
5	0.112	0.065	10267	187	264	267	0	0	2.6	0.0
6	0.113	0.064	10167	184	275	275	0	0	2.6	0.0
7	0.114	0.063	9875	182	259	261	0	0	1.8	0.0
8	0.115	0.062	10445	181	300	298	0	0	2.2	0.0
9	0.117	0.063	11510	182	305	305	0	0	2.2	0.0
10	0.117	0.063	11590	183	306	305	0	0	2.2	0.0
11	0.118	0.063	10758	183	287	287	0	0	2.2	0.0
12	0.119	0.063	11370	183	298	300	0	0	1.8	0.0
13	0.119	0.063	10567	183	285	291	0	0	3.4	0.0
14	0.120	0.064	12683	185	314	315	0	0	3.0	0.0
15	0.119	0.064	11674	185	311	313	0	0	2.2	0.0
16	0.119	0.064	10379	185	262	259	0	0	3.4	0.0
17	0.119	0.064	10261	184	271	268	0	0	4.6	0.0
18	0.120	0.063	10987	184	267	275	0	0	1.4	0.0
19	0.120	0.063	10340	183	285	280	0	0	3.4	0.0
20	0.121	0.064	11797	184	316	316	0	0	3.4	0.0
21	0.121	0.064	11772	184	304	304	0	0	3.4	0.0
22	0.121	0.064	11222	185	280	285	0	0	1.4	0.0
23	0.121	0.064	11068	185	289	284	0	0	3.4	0.0
24	0.121	0.064	10733	184	284	288	0	0	1.8	0.0
25	0.121	0.064	11006	184	290	286	0	0	3.4	0.0
26	0.121	0.064	11760	185	304	306	0	0	2.6	0.0
27	0.121	0.064	11285	185	298	299	0	0	2.2	0.0
28	0.120	0.064	11390	185	293	289	0	0	3.8	0.0
29	0.120	0.064	11415	189	298	300	0	0	3.0	0.0
30	0.120	0.064	11303	185	295	295	0	0	3.0	0.0

Table 1. Performance Data For Case 0/100/180/48.

CS:	1	2	3	4	5
PS:	100	200	300	400	500
SERVE:	120	150	180	210	240
SLOTS:	24	32	40	49	56
			design center line		

The purpose of this design was to hypothesize a set of variable ranges which will allow us to systematically estimate performance sensitivity within these ranges. To obtain such estimates for each of the four parameters, we performed a series of 5 runs. The first of these cases is the design center, namely 3/300/180/40. The other four cases correspond to the axial points on the heavy loading side of the network, namely 5/300/180/40,

3/500/180/40,

3/300/240/40, and

3/300/180/24.

Figures 5 and 6 depict the mean packet delay (MPD) and average link utilization (ALU) sensitivities, respectively, to each of the four input parameters. The horizontal axis represents percent change from the design center. The results indicate that both MPD and ALU are very sensitive to the number of slots, thereby suggesting that the SLOTS range be decreased. Using a MPD value of 10 seconds as a criterion, we see from Figure 5 that an approximately 30% change in the number of slots will cause a 10 second delay. Hence, we restrict our SLOTS range to (28,52) instead of (24,56).

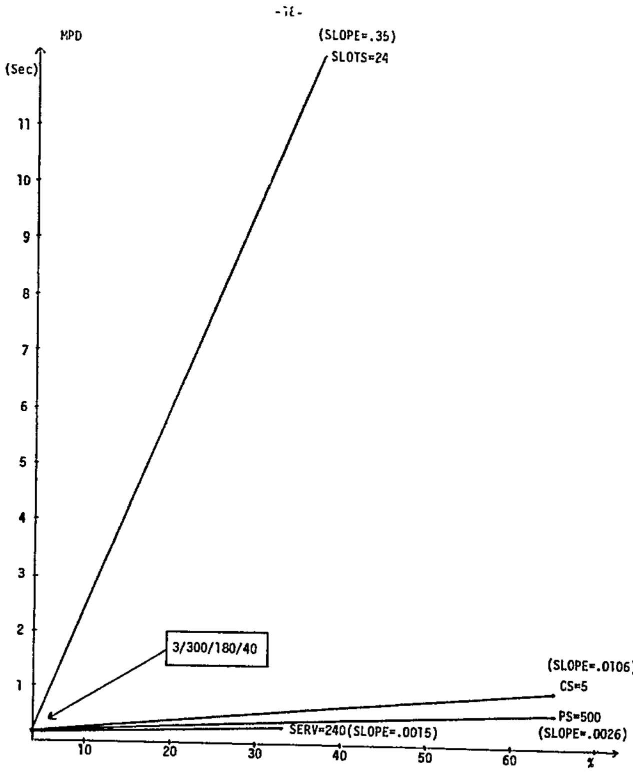

Figure 5. Mean Packet Delay (MPD) Sensitivity to 4 Input Parameters

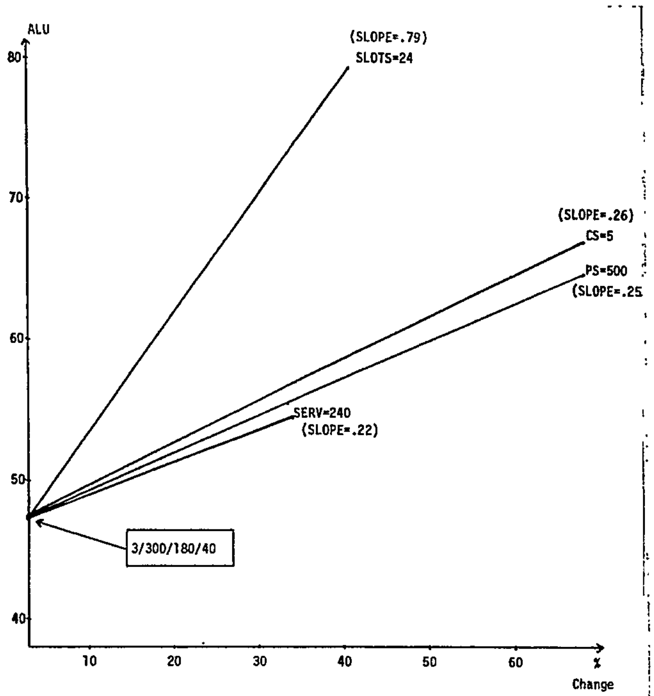

Figure 6. Average Link Utilization (ALU) Sensitivity to 4 Input Parameters

Alternately, the ranges on the other three variables could be increased. The sensitivity to SERV is quite limited, so we arbitrarily increased the voice call service time range to (60,300) seconds instead of (120,240) seconds. Realistically, a range of 1-5 minutes for average call duration seems adequate. The ranges of CS and PS are not as easily dispensed with. In the CS case, we have already observed that MPD was out of bounds for CS=10, but that for CS=5(Table 2, Run #10), the MPD was quite reasonable. This suggests checking some point between 5 and 10. We chose 7. In fact, since we already have data for 0/400/180/48 and 10/400/180/48, we elected to run two more cases, 5/400/180/48 and 7/400/180/48, to check the CS spectrum form 0 to 10. The results are shown in Figure 7, which depicts a sharply increasing MPD sensitivity as the CS value increases beyond 5. Being consistent with our previous MPD-10 criterion, Figure 7 suggests a maximum CS level of about 8.

To adjust the PS range, we elected to run two more cases, 5/500/180/40 and 5/700/180/40, to complement the existing case 5/300/180/40 (Run #10). Figure 8 shows the MPD for each of these cases and suggests that a maximum PS level of 600 should be adequate.

Summary

Table 2 summarizes the input parameters used and the performance parameters for each of the 15 simulation runs performed in this study. Additionally, the time at which the performance data indicated steady state is also given for each case. We conclude that the following ranges for the input parameters reflect an input domain for which network performance is either at or fairly near a feasible grade of service.

SUMMARY OF EXPERIMENTS

	INPUT				OUTPUT					
RUN #	CS	PS	SERV	SLOTS	MPD	ALU	THR	AQL	BLK	STEADY STATE
1	0	100	180	48	.119	.064	185	2	0.0	15
2	0	400	180	48	.127	.252	724	10	0.0	20
3	10	100	180	48	28.9	.844	206	112	.256	40
4	10	400	180	48	19.1	.891	816	117	.299	40
5	20	100	180	48	75.7	.900	171	168	.500	*
6	20	400	180	48	23.1	.920	425	77	.510	*
7	3	300	180	40	.233	.474	566	16	.002	35
8	3	300	180	24	14.1	.788	624	134	.157	35
9	3	300	240	40	.284	.548	576	39	.008	35
10	5	300	180	40	.937	.648	594	73	.028	36
11	3	500	180	40	.406	.641	981	77	.032	36
12	5	400	180	48	.315	.593	764	35	.012	30
13	7	400	180	48	2.63	.749	796	105	.081	31
14	5	500	180	40	4.36	.796	1018	134	.135	31
15	5	700	180	40	15.1	.895	1384	102	.289	35

* Data at 15 minutes of simulation - steady state not yet attained.

Table 2. Summary of Experiments

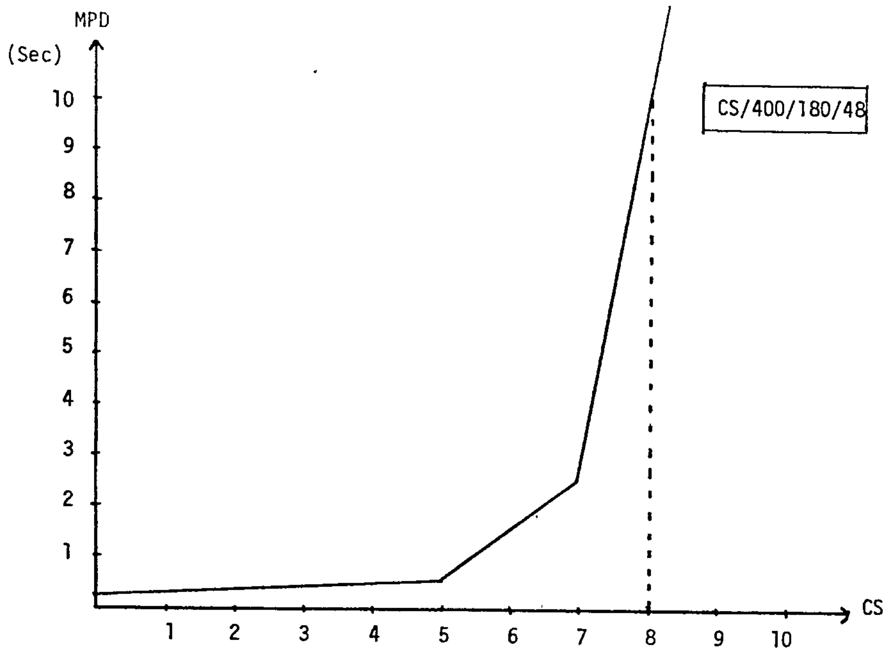

Figure 7. MPD Sensitivity as a Function of CS

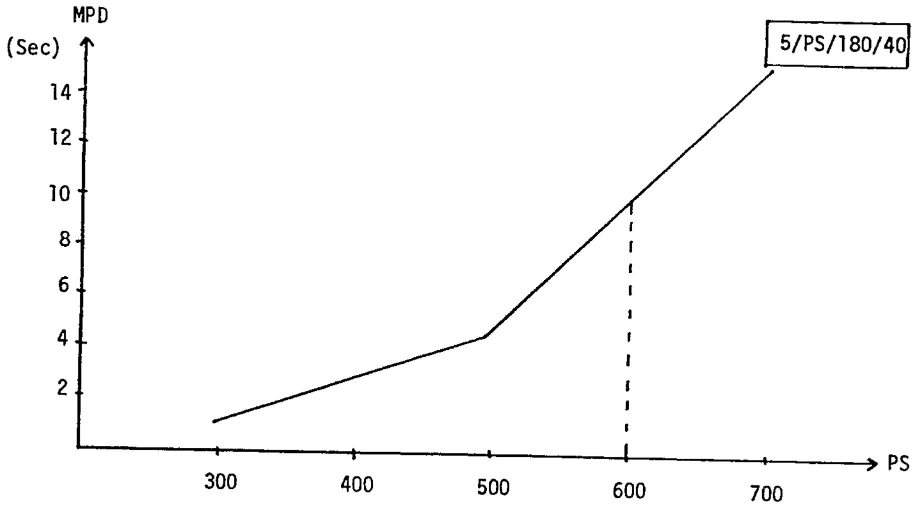

Figure 8. MPD Sensitivity as a Function of PS

CS: 0-8 (calls/minute)

PS: 0-600 (packets/sec)

SERV: 60-300 (sec)

SLOTS: 28-52 (capacity indicator)

Furthermore, 40 minutes of simulation time appears to be sufficient time to reach steady state for this domain of input parameters. The parameter ranges resulting from this study represent a necessary first cut in the sizing of the network simulator and constitute an excellent starting point for a detailed sensitivity analysis of the model.

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