E-3

- A hop-by-hop passive echo acknowledgement (HBH Echo Ack) along the path. When device i transmits a packet, it waits a sufficient time to allow devices that receive the packet to repeat it. When any of these repeats the packet and the packet is received by device i, it considers it as an Ack.

In a point-to-point network, such as the ARPANET, the channels are fixed so that when node j receives a packet on channel k it knows the device, say i, which has transmitted the packet to it, and thus can transmit a specific HBH Ack to device i. In the Packet Radio System, however, this information is not available. Therefore, the HBH Ack must be <u>independent of the path that the packet travels on</u>. The HBH Echo Ack test used included the following:

1) identification of the packet

2) tests that the MHN of the Echo received is <u>smaller</u> than the MHN of the packet stored. This insures that the packet has advanced along the path, rather than being a retransmission from devices that had the packet previously.

The HBH Echo Ack has several advantages over a specific HBH Ack:

1) it simplifies the repeater (hardware and software) so that it need not constract and manage acknowledgement packets.

2) it reduces the traffic overhead of transmitting specific acknowledgements. This is most significant in a broadcast network.

3) it enables acknowledgement of several devices at a time; in particular, all devices which store the packet with a MHN larger than that received are acknowledged.

4) It enables shortening the transmission path, as described below.

Since the RSP's by repeaters are randomized in time, a terminal frequently does not identify the repeater nearest to the station within range of the terminal. In fact, if two repeaters are labelled on the same path to the station and both are within range of the terminal, there is a higher probability that the terminal will identify the repeater farther away from the station since, on the average, the latter handles less traffic. Suppose a single data rate channel is used throughout the transmission path between terminal and station, the terminal identifies a packet transmitted to it by its specific terminal ID, and the station can recognize any packet destined for it. Then a communication path as shown in Figure 2 may be established. In Figure 2, the terminal is within an effective range to R4, R5, and R6; and the station within an effective range to R1, and R2. The terminal shown identified R6 as the repeater to which it transmits. The path from termianl to station will usually be : T→R6→R5→R4→R3→R2→S; and from the station to the terminal S→R2→R3→R4→T. The end devices, terminal and station, transmit the Echo Ack immediately after receiving the packet, and <u>transmit it with MHN=0</u>; thus they acknowledge all devices which still store the packet. In particular, when R4 transmits a packet towards the terminal it is addressed to R5, however, the terminal may receive this packet and acknowledge both R4 and R5 simultaneously. Similarly, when R2 transmits towards the station it addresses the packets to R1, when the packet is received by the station it acknowledges both R1 and R2 simultaneously.

D. Performance Measures

• Throughput

Considering the set of stations and the set of terminals as the end devices, the system throughput (in packets) is defined as the rate of information packets (IP's) that originated at stations and arrived at terminals, plus the rate of IP's that originated at terminals and arrived at stations.

• Delays

The following delays are measured:

1) Terminal delay to identify specific repeater.

2) Terminal delay to establish communication with station and to negotiate protocols.

3) End-to-end delay for an IP.

4) Terminal interaction delay as a function of the number of IP's transmitted and received. The interaction delay is defined as the time elapsed from terminal origination to departure.

• Blocking and Loss

When a terminal does not successfully identify a repeater (or station) after transmitting an SP for the maximum number of times specified, it is considered blocked. In addition, under certain conditions, terminals will depart from the system without completing communication. This will contribute to additional system loss. The blocking and loss are measured separately since the former indicates the difficulty in entering the system, whereas the latter reflects on the inefficiency of the routing.

- Device Performance

  1) Probability that the station is busy. The station is sampled during the simulation and is assumed busy if it is actively receiving or transmitting; otherwise, it is assumed idle. This measure is an indication of the channel traffic at the station.

  2) Successful completions by repeaters. The number of packets that each repeater has successfully switched are counted. This indicates the distribution of load in the network and reflects on the power duty cycle of repeaters.

- Other Measures

  1) The number of terminals in the system, the total number of packets stored in the system, the number of events to be processed; all as a function of time.

  2) The complete output includes a detailed description of the flow of significant communication events.

TYPICAL COMMUNICATIONS PATH

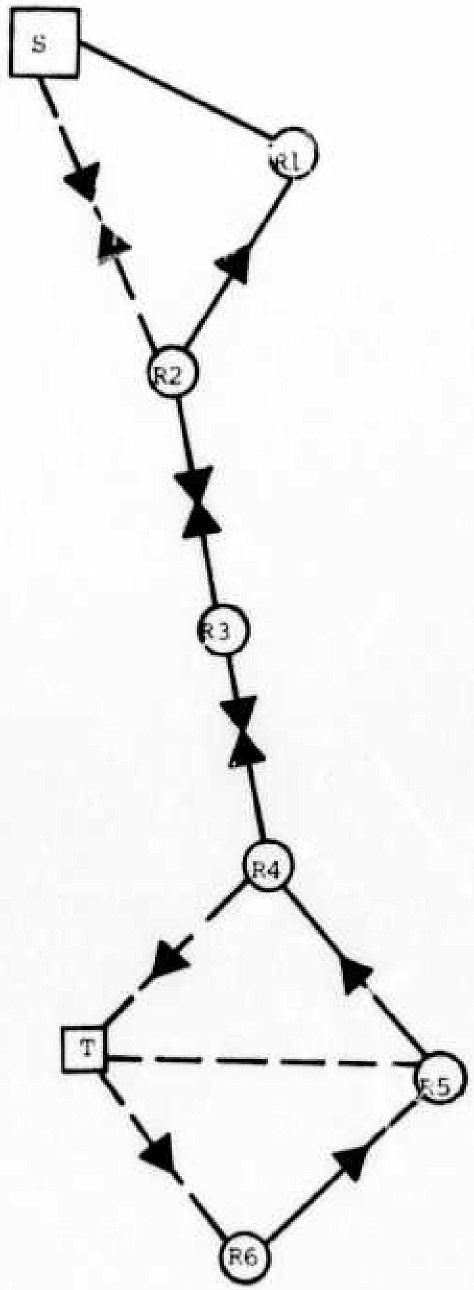

FIGURE 2: The solid lines indicate the labelled path between the repeaters and station. The dashed lines indicate the effective connectivity of terminal and station to repeaters. The arrows indicate the practical transmission path for the particular terminal, to and from the station.

VII. LOGICAL OPERATION OF DEVICES

A. States of Devices

Each device is characterized by a state vector. Some of the state variables will be needed in the physical devices, for example, the label, a parameter indicating the maximum number of transmissions, the maximum handover number to be assigned by repeater and station, the state of occupancy of its storage, etc. Other variables are particular to the simulation. The following are examples:

Operational State of Device

PR - Passive Receive State: The device is in receive state and does not sense any carrier.

AR - Acrive Receive State.

AT - Active Transmit State.

ART- Active Receive and Transmit.

When a device is in state AR or ART, it can be receiving several overlapping packets simultaneously. In the program, we use a common channel configuration (half duplex). Thus, since carrier sense is used for channel access, the device can change to AT only from PR and to ART only from AT.

- Number of Overlapping Packets

This number is incremented by one whenever the device is in state AR of ART and a new packet begins to arrive; and decremented when a packet transmission ends. The number of overlapping packets indicates the number of times an end of packet transmission has to occur before the device changes its state to PR.

- End of Busy Period

This time is recorded for the purpose of saving CPU time. The transmission time of a packet is set to the End of Busy Period plus a random time; otherwise the devide may be called to transmit a packet several times during its busy period.

B. Terminal

When a terminal originates a message, it begins to transmit and retransmit a SP to identify a specific receiver. If it does not identify a specific receiver after a specified number of transmissions, it departs from the system. We say that such a terminal is blocked. When a terminal identifies a specific receiver, it substitutes the label and MHN sent by the receiver itno its IP and begins to transmit its IP. The IP is retransmitted after short waiting periods of time until a HBH Echo Ack is received. At that time, the terminal stores the IP for a longer period of waiting after which the IP is reactivated if an ETE Ack is not received.* The terminal is expecting several IP's

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* We use the term retransmission when a device waits a relatively short period of time (less then 2 IP slots) and is awaiting a HBH acknowledgement. We say that a packet is reactivated when an end device stores the packet, awaiting an ETE Ack. When a packet is reactivated, it goes through the whole process of retransmissions.

from the station, which are ETE acknowledged by the terminal. When all IP's from the station are received and ETE acknowledged, the terminal departs from the system.

C. Repeater

A repeater does not distinguish between IP's or ETE Acks, except for their transmisison time. When an IP (or ETE Ack) is recrived by a repeater (addressed to it) and the repeater has available storage, it stores the packet, decrements the MHN, modifies the packet label according to the routing, and begins to transmit and retransmit the packet, awaiting the Echo Ack. When an Echo Ack is received, the repeater discards the packet. When a repeater is not successful in transmitting along the "shortest" path, it begins to search for an alternate receiver by transmitting SP's. When one is found, it transmits the entire packet to it; otherwise, it discards the packet. When a repeater receives an SP, it checks whether it has available storage, if it does, it makes one attempt to transmit a RSP and then discards it. When a repeater receives a RSP, it tests whether it needs one, if it does, it used the label, otherwise it discards it. The repeater currently simulated has buffer storage for two packets; one exclusively used for packets directed towards the station, and the second for packets toward the terminal. In addition, the repeater can inspect all packets that is receives, which are stored in common arrays in the simulation program. Thus, from a practical viewpoint, buffer storage for three packets per repeater are provided in the simulation program.

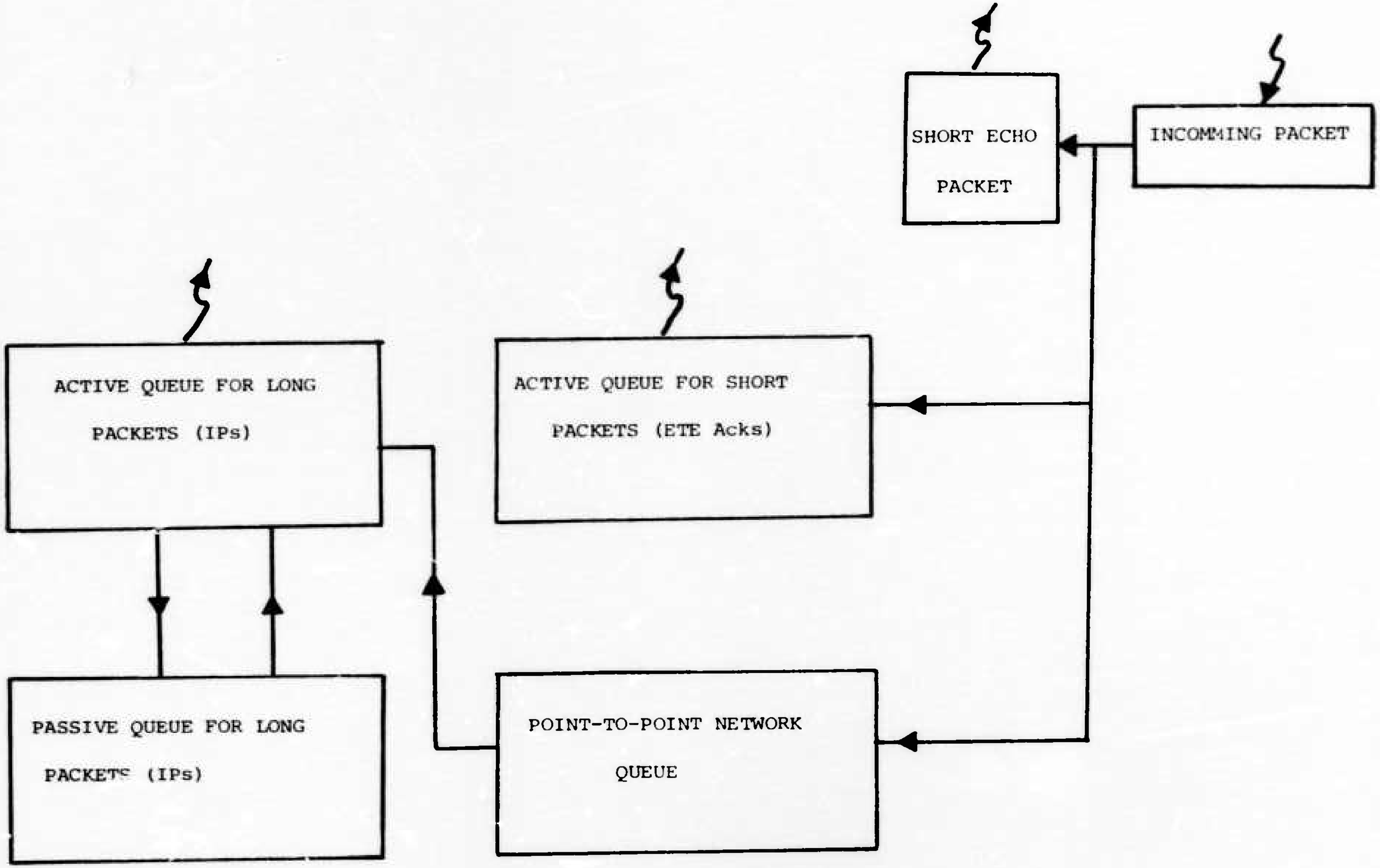

FIGURE 3: STORAGE IN STATION OF IP's AND ETE ACK's.

D. Station

• The storage organization in the simulated station is shown in Figure 3.* There are two queues for active packets. Packets in these queues are active in the sense that they are retransmitted after short random periods of waiting until an Echo Ack is received. The active queue for long packets contains IP's from the station to terminals. Once an IP is Echo acknowledged, it is stored in the passive queue for a longer period, after which it is reactivated if an ETE Ack from the terminal is not received. The active queue for short packets contains ETE Acknowldegement packets to terminals, and these have priority over the long active apckets. The ETE Ack packets are, obviously, discarded once an Echo Ack is received. The point-to-point (PTP) network queue simulates the interaction of the packet radio network with a PTP network. When a new IP is received from a terminal, it is stored in the PTP network queue fir a random time, after which a response message containing several response IP's to that terminal are generated and placed into the active queue for long packets. The station responds immediately to SP's, and ignores RSP's.

E. Flow Diagrams of Devices

Figures 4, 5, and 6 show the flow diagrams of the devices used in the simulator. These diagrams are simplified to the extent that they show "what to do" but not sufficiently detailed to show "How to do it." The latter depends on the particular system simulated, i.e., the routing, the channel configuration, etc.

A device is called from the subroutine EVENT; the calling sequence includes, among others, an interrupt number which indicates to the device, the task that is has to perform. The only event which

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* Figure 3 shows only the storage of IP's and ETE Acks for transmission to terminals in the packet radio network.

is external to a device is that with an interrupt = 1. All other entries to a device are due to events generated by the device itself. Thus, the number of exogeneous events is very small compared to the number of events generated by devices; in particular, when the offered data rate to the system is high.

It is clear that when the offered traffic rate to the system is high, there will be many collisions of packets. Thus, to save computing time, each device was coded so that it does not examine the content of the packet or whether the packet is addressed to it, at the beginning of packet reception (see diagrams). This is done at the end of packet reception, providing there was no interference.

There are parts of the subroutines of Repeater, Station, and Terminal which are identical and thus, coded into subroutines. These relate to the identification of the header content, and the channel access mode. Some of these can be associated with the modem of the physical devices. The parts which differ involve mainly the processing needed once the packet has been identified, e.g., queueing at the station, ETE Ack by end devices, etc.

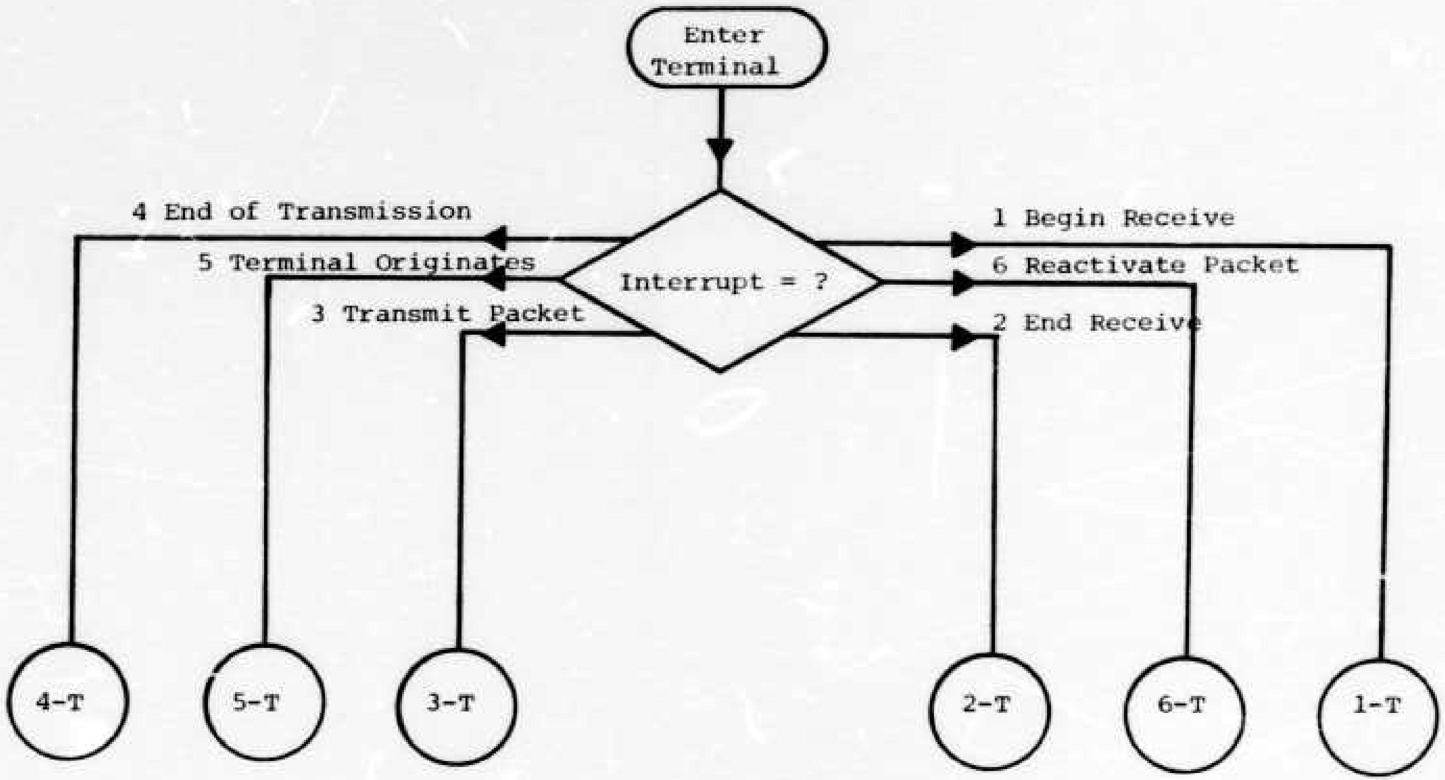

FIGURE 4A

TERMINAL DEVICE FLOW DIAGRAM

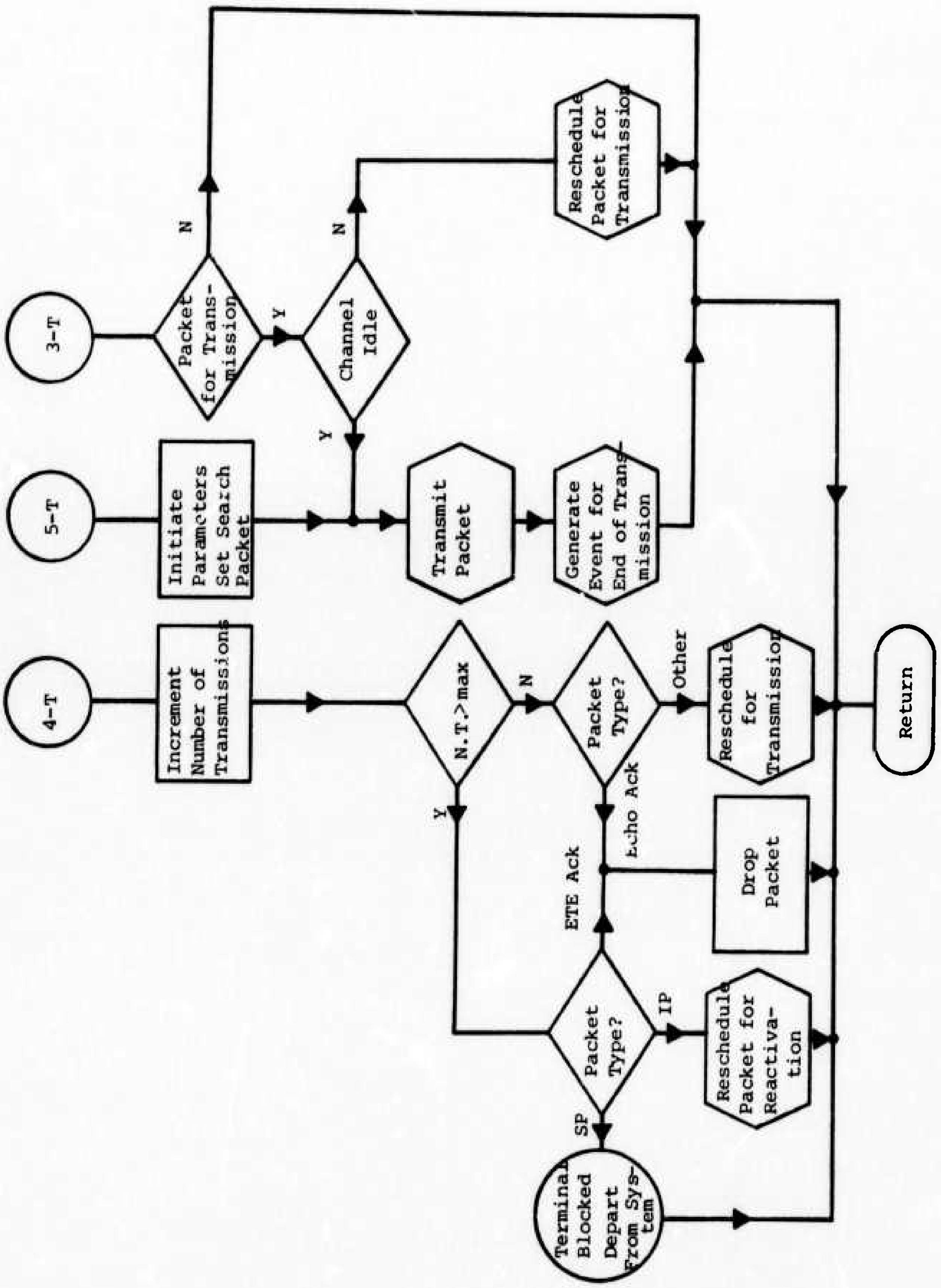

FIGURE 4B

TERMINAL DEVICE FLOW DIAGRAM

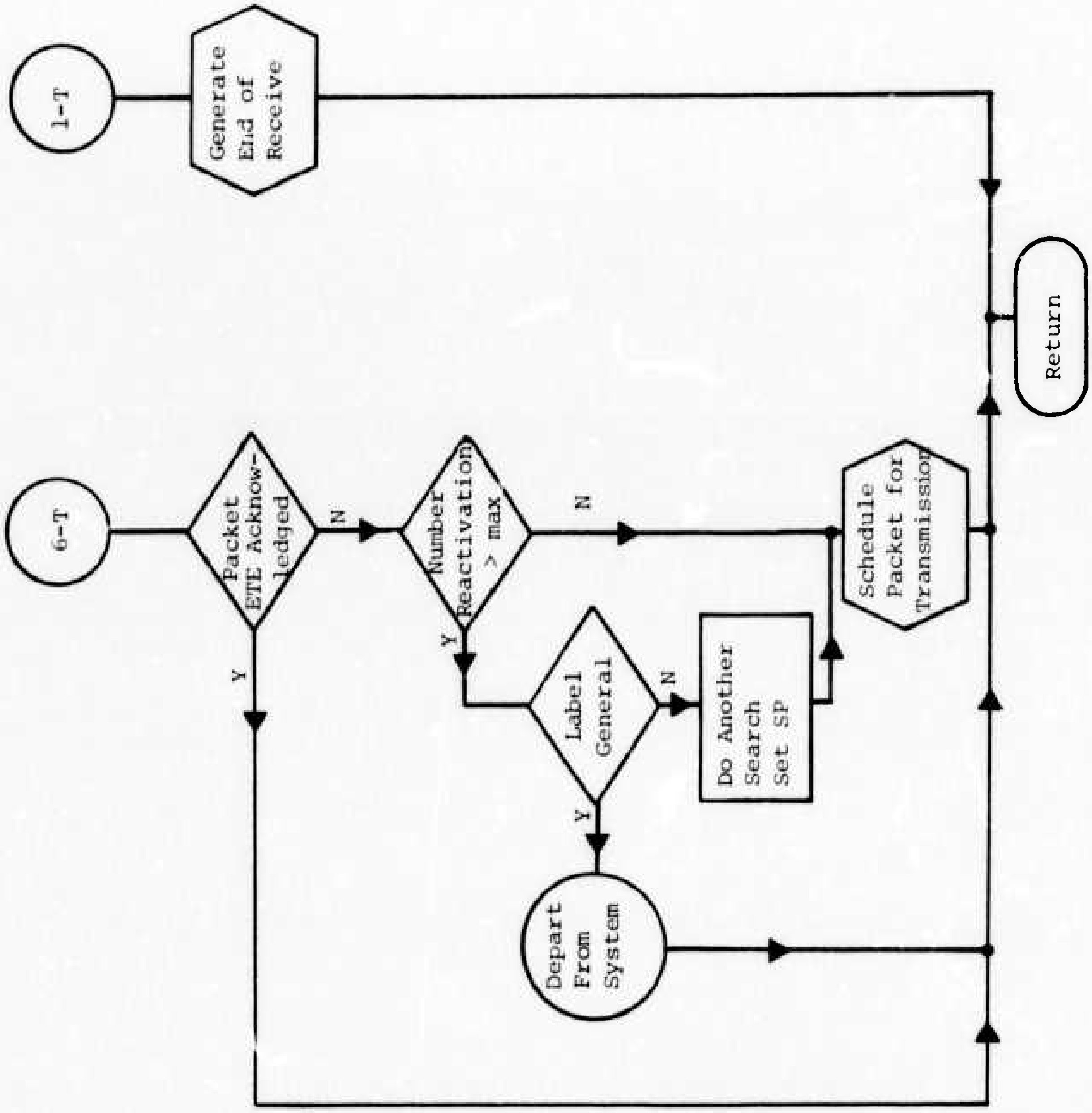

FIGURE 4C

TERMINAL DEVICE FLOW DIAGRAM

TERMINAL DEVICE FLOW DIAGRAM

Network Analysis Corporation

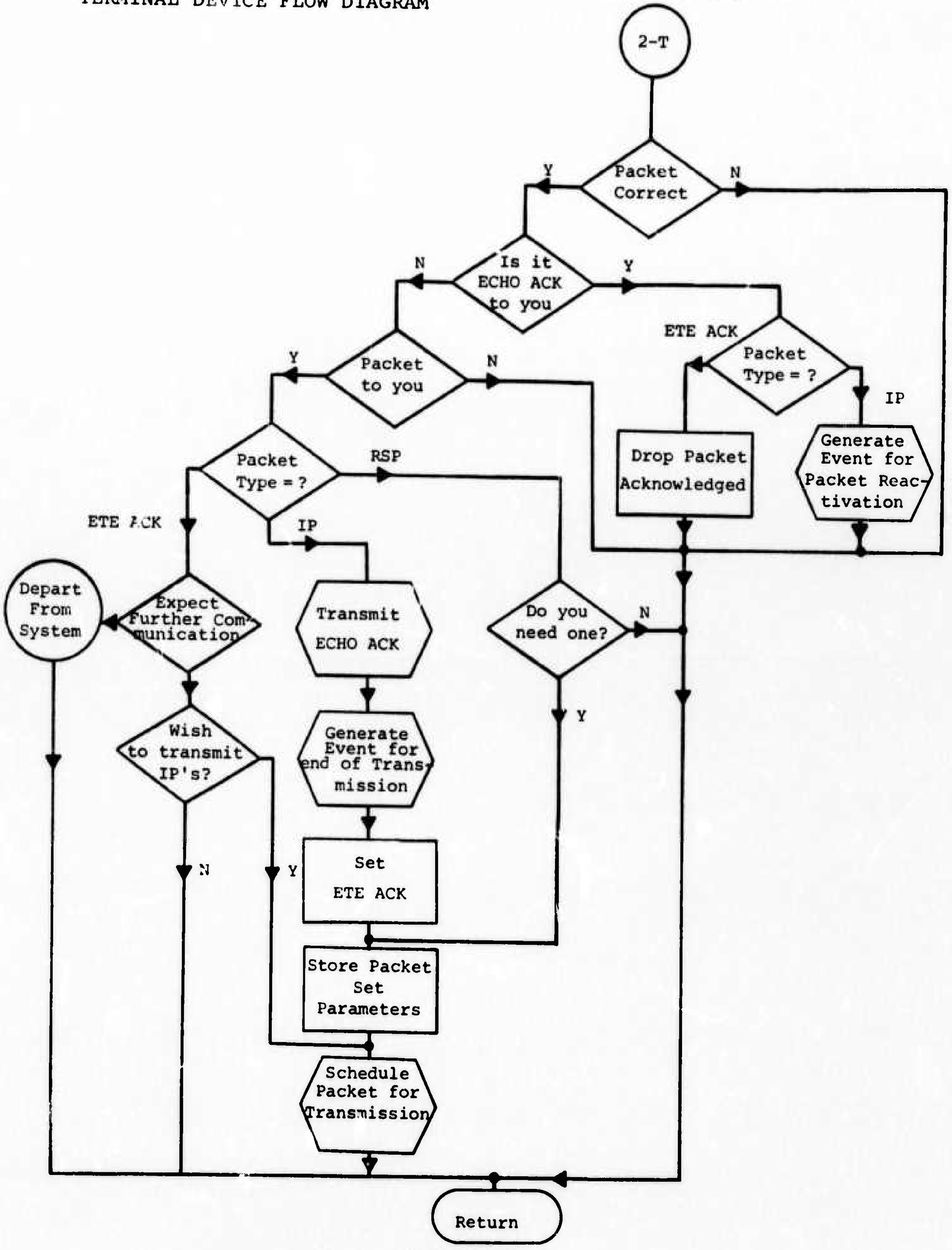

FIGURE 4D

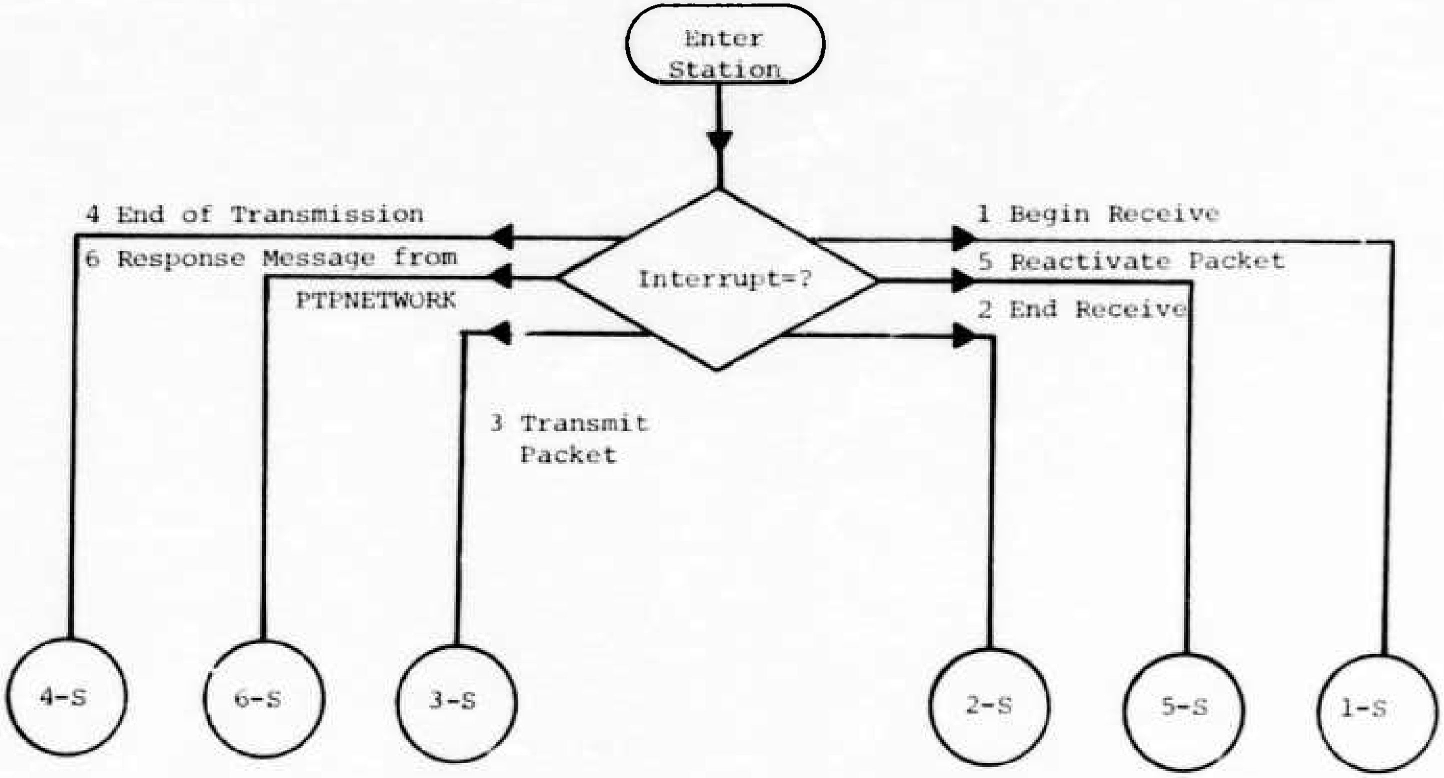

FIGURE 5A

STATION DEVICE FLOW DIAGRAM

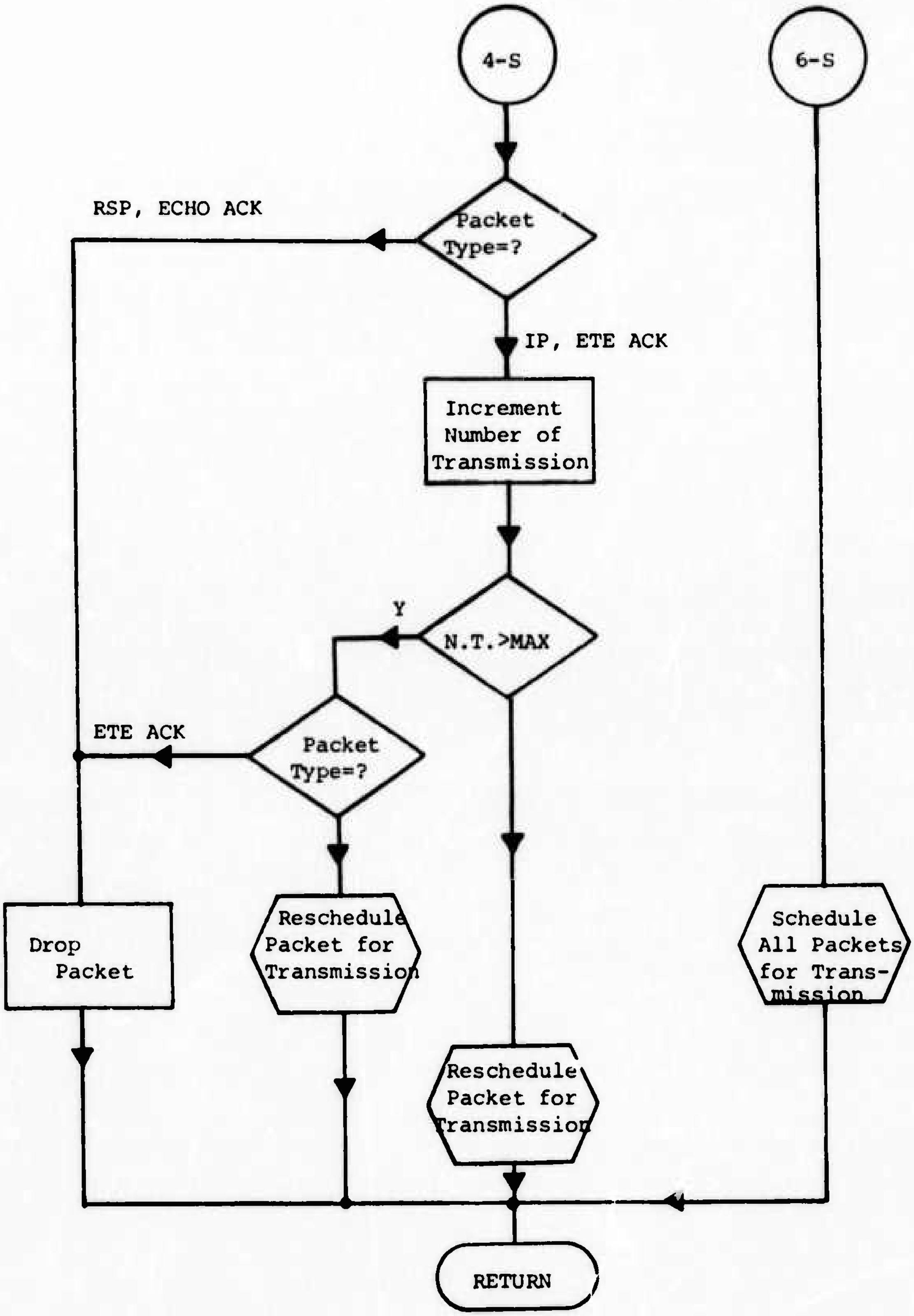

FIGURE 5B

STATION DEVICE FLOW DIAGRAM

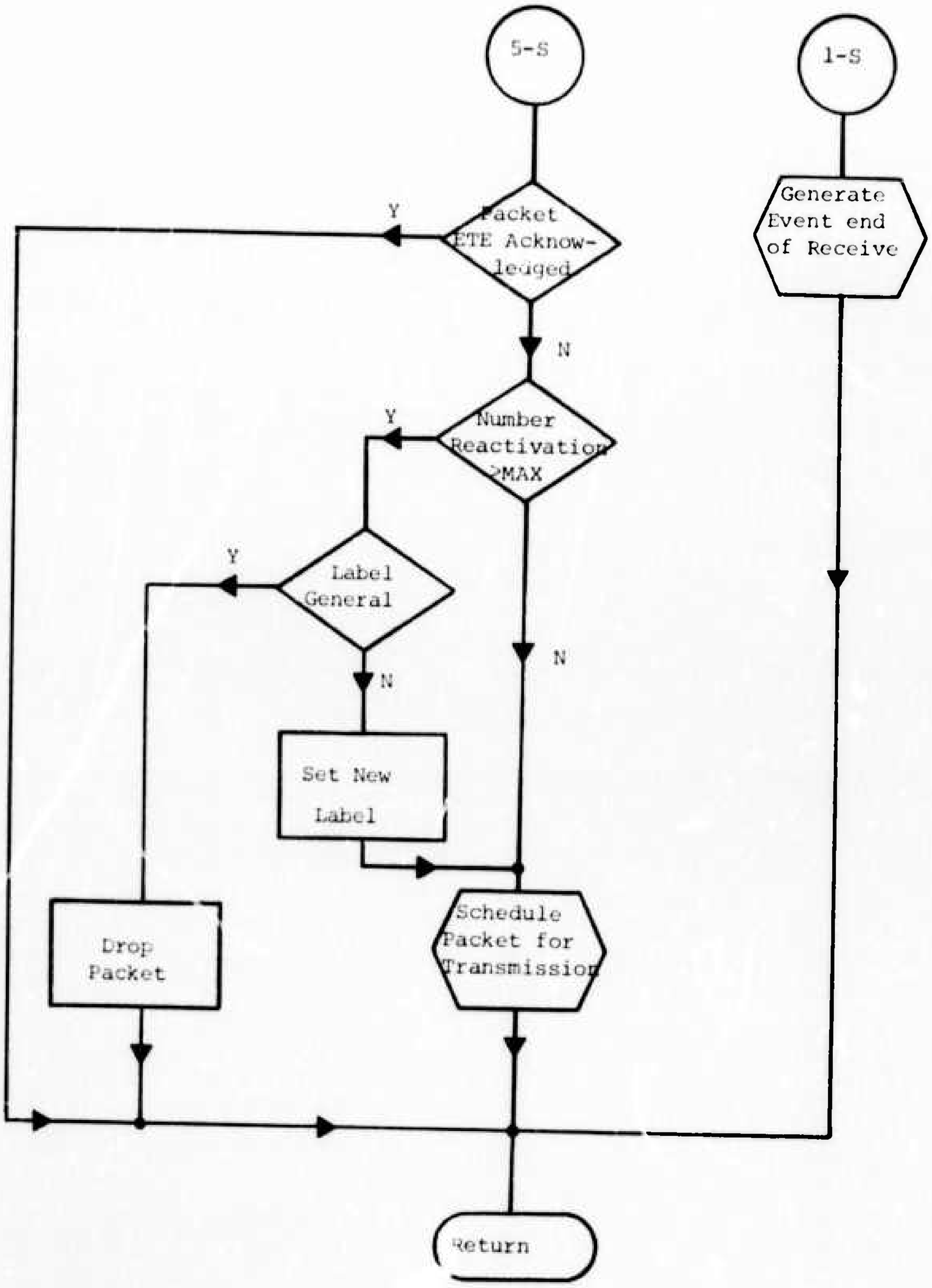

FIGURE 5C

STATION DEVICE FLOW DIAGRAM

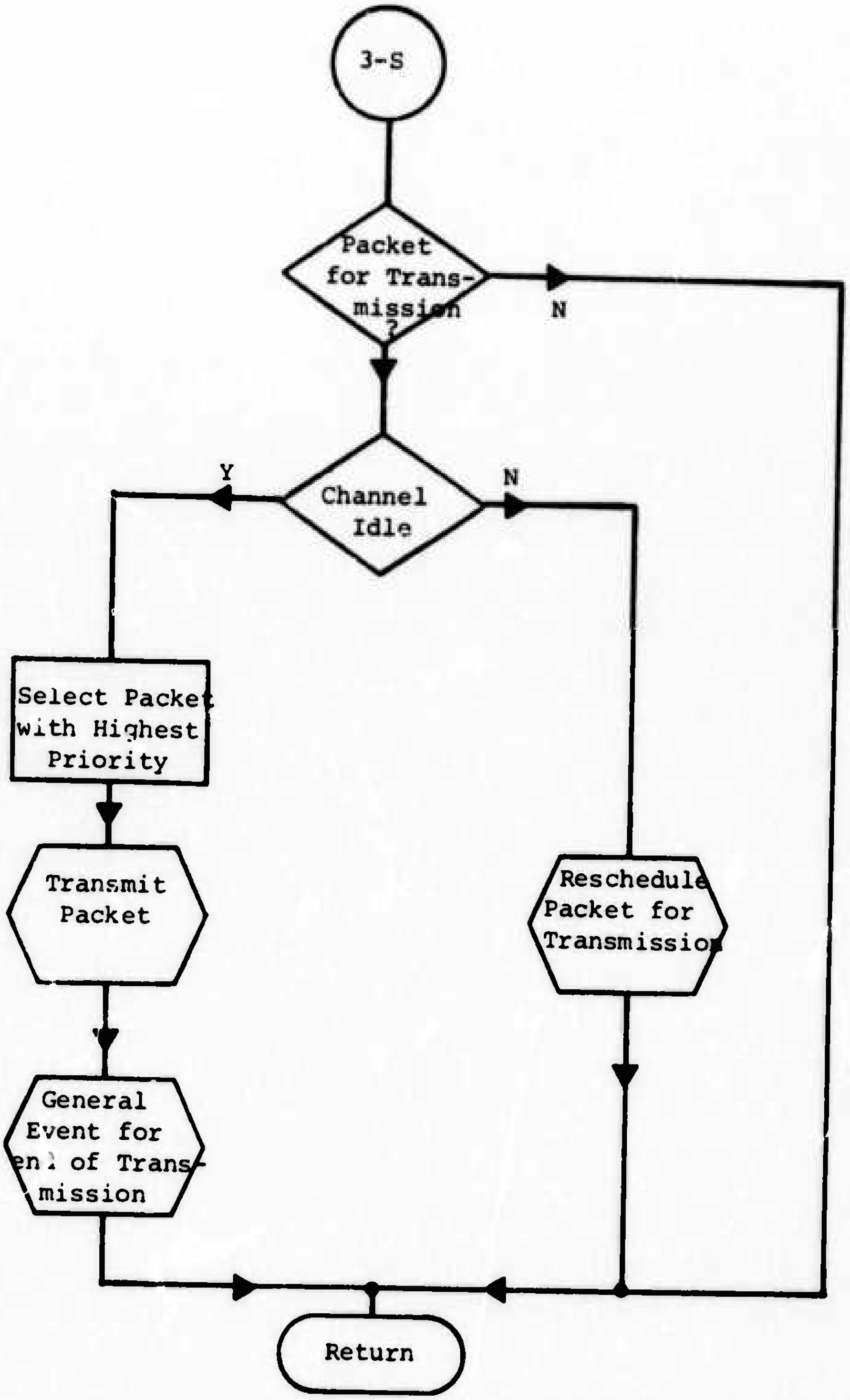

FIGURE 5D

STATION DEVICE FLOW DIAGRAM

STATION DEVICE FLOW DIAGRAM

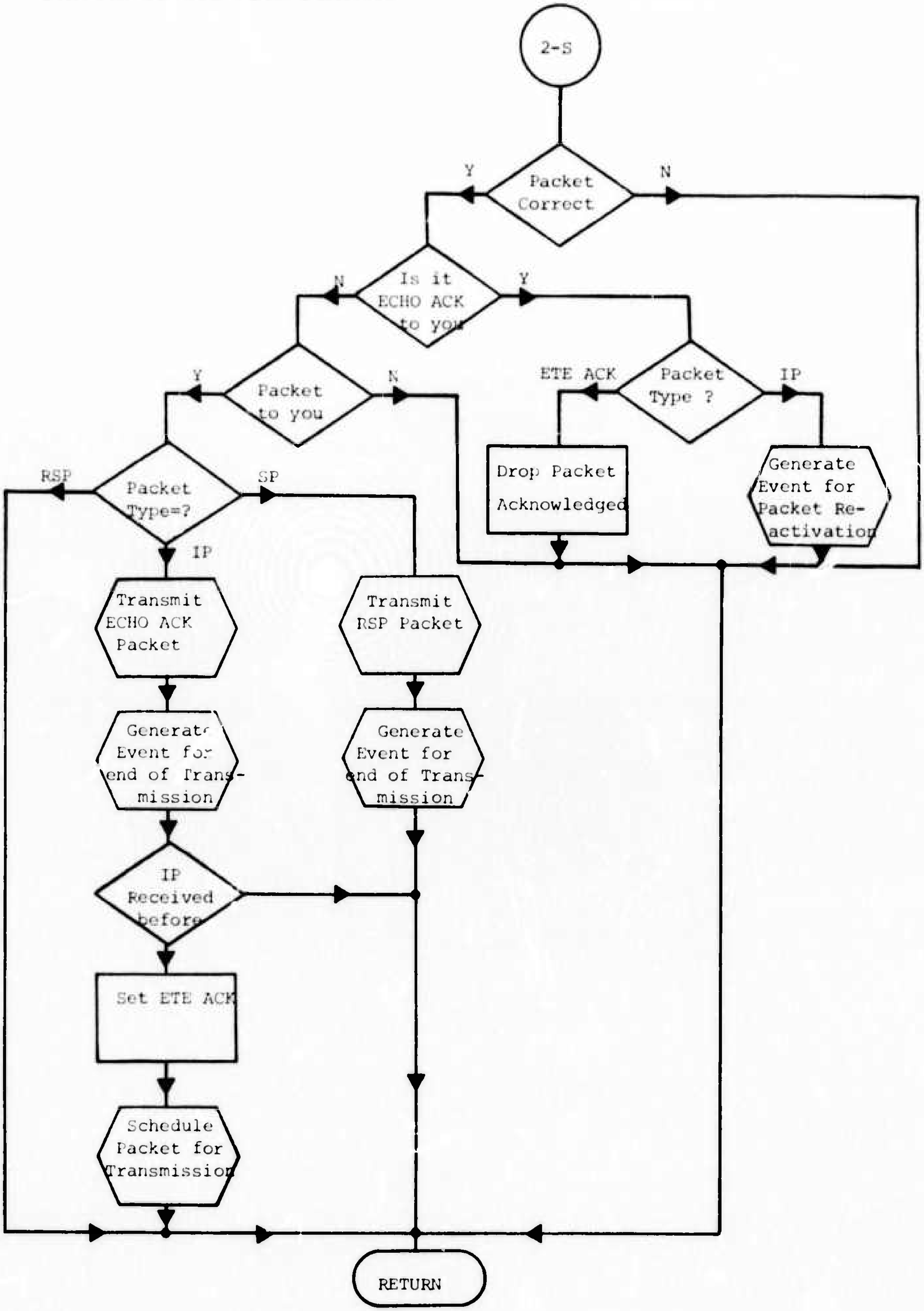

FIGURE 5E

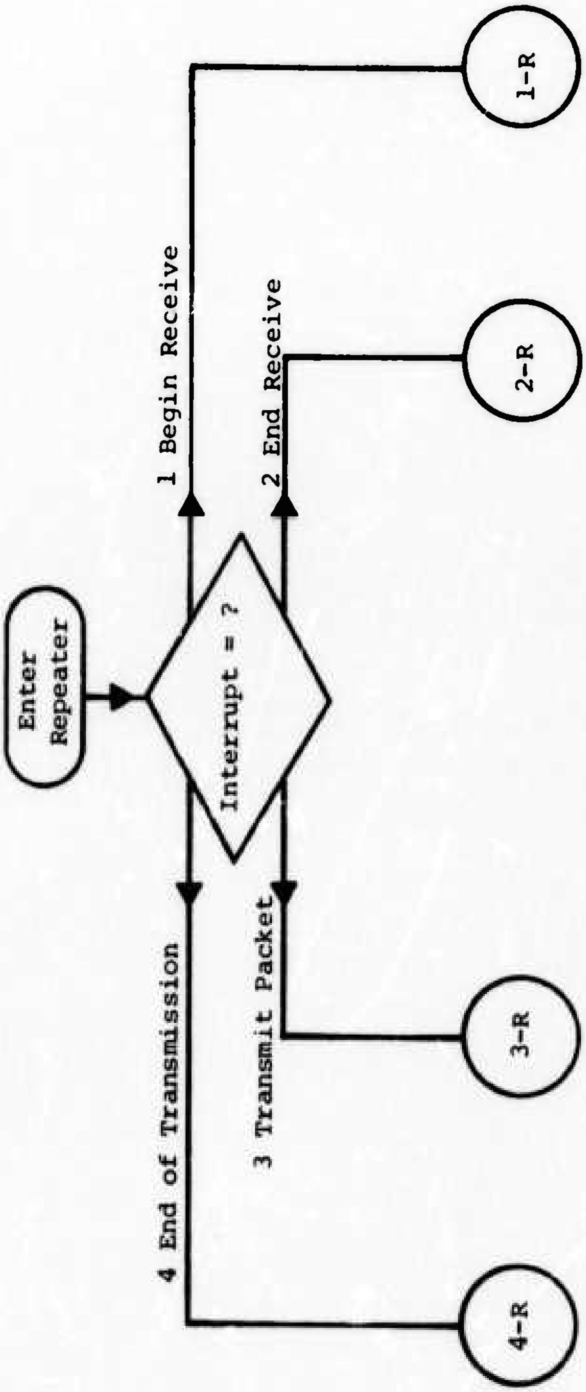

FIGURE 6A

REPEATER DEVICE FLOW DIAGRAM

REPEATER DEVICE FLOW DIAGRAM

Network Analysis Corporation

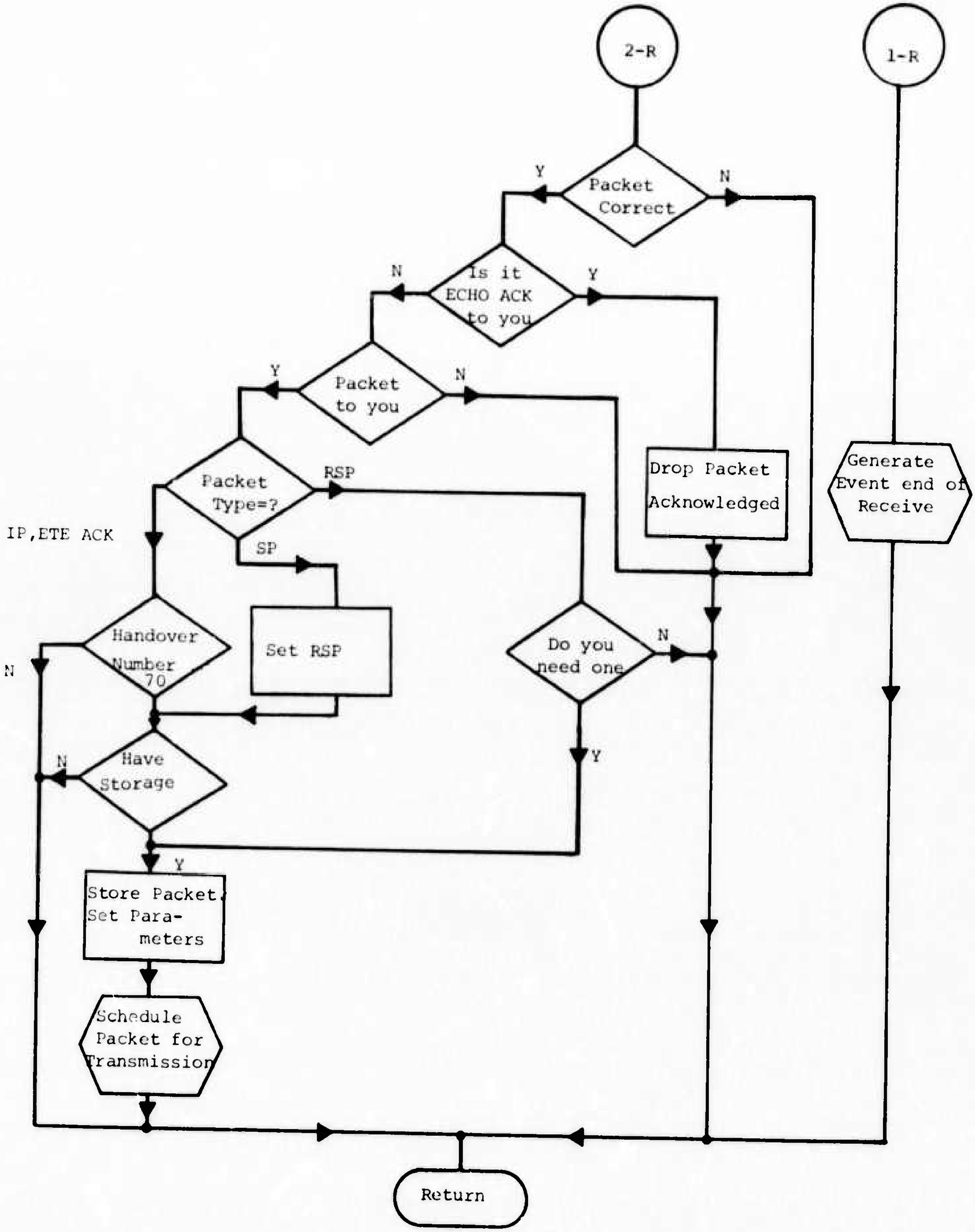

FIGURE 6B

REPEATER DEVICE FLOW DIAGRAM

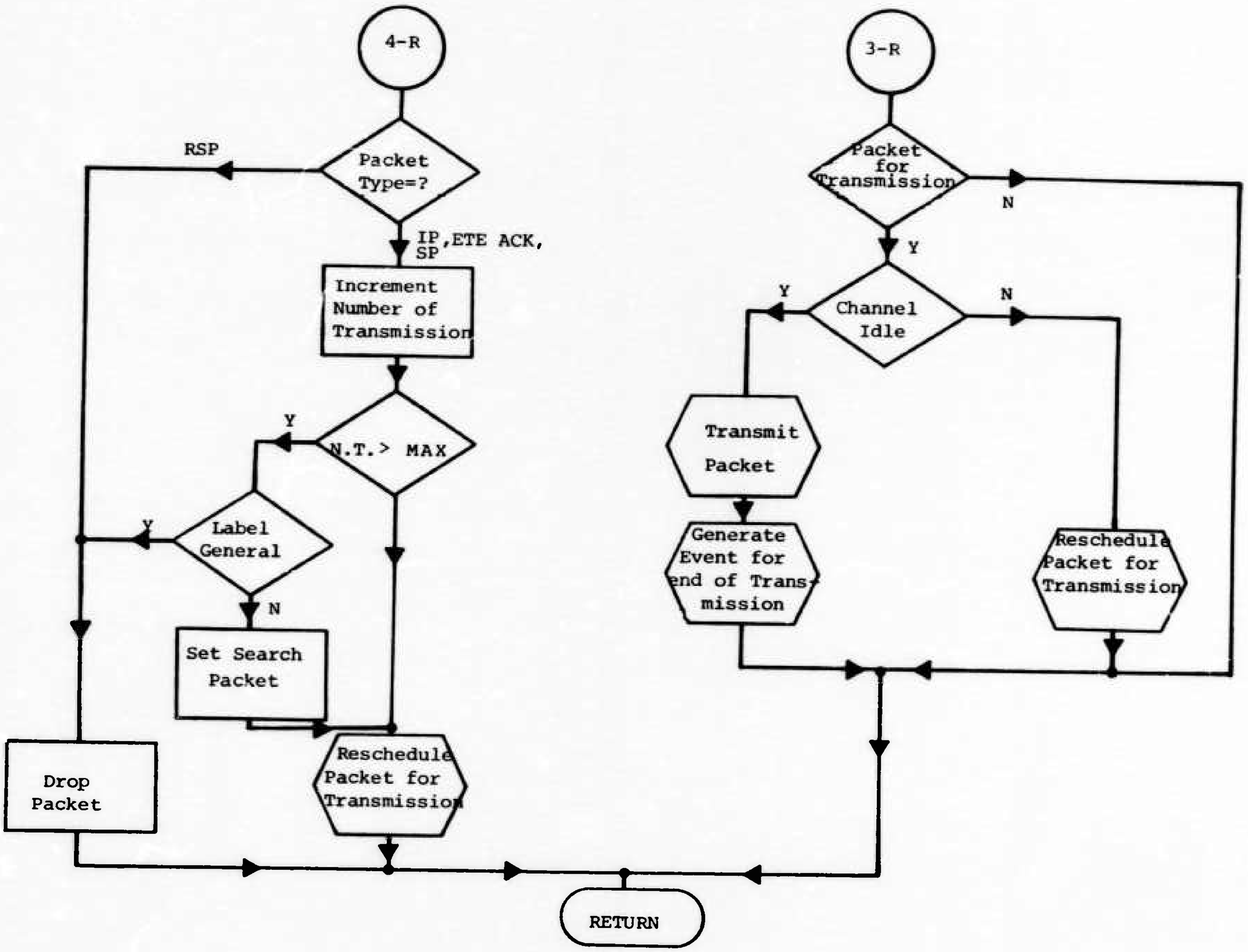

FIGURE 6C

VIII. SUBROUTINES OF THE SIMULATOR

A. Data Structure and Management Subroutines

EVENT	Takes the next event out of the Event Data Structure for execution.
INHEAP	Adds an event to the heap in the Event Data Structure.
INMESS	Allows the introduction of special messages such as acknowledgements, control messages and the like into the Message Data Structure.
INPUT	Reads the input parameters and determines the placing of repeaters and stations.
MESSREL	Is called by device routines to release a message as soon as all packets representing the message are deleted.
NEWMESS	Generates next exogenous message and adds to the Message Data Structure.
NEWPACK	Adds a new packet to the Packet Data Structure.
NXTEVNT	Adds a new event to the Event Data Structure.
OUT	Prints out intermediate data for debugging.
OUTHEAP	Takes the index of the next event time from the heap.
PRSIM	The driver routine. (main program)

MEASURE	Collects data on system performance.
MCOUNT	Counts the number of packets associated with each terminal which are stored in the system.

B. Communication and Device Subroutines

REPEAT	Main subroutine of repeater.
STATION	Main subroutine of station.
TERMINAL	Main subroutine of terminal.
DEVINIT	Reads parameters which devine the particular communication system, labels, and flow control parameters. Initializes states of devices.
BGNRCV	Maintains states of devices related to the RF channel (e.g., number of overlapping packets).
ENDRCV	Same as above at the end of packet reception.
ECHO	Records that a device is receiving an echo acknowledgement.
SRER	Called when a non-overlapping packet is received for testing packet type and label.
ALTROUT	Called after repeater receives an RTS, checks whether repeater needs one, and has not used one before.
SRTTRT	Transmits packet and generates an event for the end of packet transmission.

REPNEXT	Determines which packet of a repeater is to be transmitted next.
TERSTOR	Stores correct IP's received by terminal and generates event for transmission of an ETE Ack.
SNREEKO	Called by station after receiving an Echo Ack. Identifies and maintains the queue in which the acknowledged packet is stored. If it is in IP, then it transfers the packet to another queue where it waits for an ETE Ack or for reactivation.
SHIFTQ	Shifts packets in the various queues of the station.
SNREPAK	Called by station after correctly receiving a packet. If the packet is an ETE Ack, then subroutine drops the packets acknowledged, maintains proper queues and the message counts. If it is an IP, subroutine verifies that same packet has not been received before, and if so, it generates packet and an event for transmission of an ETE Ack, and also generates a random time and an event for the arrival of the response message from the PTP network.
RESPONS	Called by station, sets all response packets to a terminal into the active queue and generates events for transmitting them.
SEKOTRT	Used by station to transmit an Echo Ack for the last hop.
CONNECT	Determines the most efficient repeater to which station should address packet when transmitting to a terminal.

TRANSMT Called by a device which transmits a packet. Puts packet in list structure; determines all the devices that should receive the packet, the exact time for beginning to receive it; and generates the events to devices.

C. Summary of Acronyms

Ack - Acknowledgement

AR - Active Receive

ART - Active Receive and Transmit

AT - Active Transmit

ETE - End-to-end

HBH - Hop-by-hop

IP - Information packet

Label- An address assigned to a device for routine purposes

MHN - Maximum handover number

MNT - Maximum number of transmissions

RP - Passive Receive

PTP - Point-to-point

RSP - Response to search packet

SP - Search packet

IX. OBSERVATION OF TRAFFIC FLOW IN THE PACKET RADIO NETWORK

The first system simulated was a Common Channel Single Data Rate system, in which the station is routing traffic as a repeater (Waive Station). We denote the system as CCSDR (NS). The system defined has a single data signalling rate for communication between terminal and repeater (or station) and in the repeater-station network; the channel is used in a half duplex mode. When the station is routing traffic as a repeater, it cannot receive packets not specifically addressed to it.

In all experiments reported here, the labels of repeaters and staiton were preassigned. The hierarchical (directed) labelling scheme of the system in this experiment are shown in Figure 7. Figure 8 shows the connectivity of the repeaters and station. That is, when a device transmits, all the devices connected to it by line are within an effective range and "hear" the transmission.

The objective of the first series of experiments was to observe the detailed operation of devices and the efficiency of the system. The following observations were made:

1. The "critical hop" in the system is that between the first level repeaters and the station. This was concluded by observing the frequency at which repeaters begin to search and at which they discarded packets, and from the observation that there is no significant difference in the delay when the number of hops from the station that a packet travels is increased.

2. There is a higher probability of end-to-end successful completion when routing from the station to a terminal than when routing from a terminal to the station. Practically, there is almost "no" difference in time delay between the delay of an information packet from the terminal arriving at the station and the time that the terminal receives an ETE Ack from the station.

3. Many packets associated with terminals that have departed from the system are routed in the network.

The effect of improving the routing capaciblities of the station can be readily observed. In particular, one can see in Figures 7 and 8 that while the connectivity station is 7, there are only 4 repeaters labelled from the station. Consequently, the station is busy of the time with non-useful traffic. This situation can be improved by changing the routing of the station so that: (1) it receives any packet that it can hear and which is (eventually) addressed to it; and (2) it transmits response packets to the repeater nearest to the terminal along the routing path that it can reach. This change was implemented for all system studies subsequent to the initial experiments. Apart from the change implemented, the observation suggests that particular attention should be given to the design of the repeater network in the neighborhood of the stations. It is also noted that these repeaters have a higher power duty cycle since they handle all packets collected from other parts [illegible] the network. The routing change made at the station enables the allocation of many more repeaters in the neighborhoos of the station, than are functionally needed, without resulting an increase in the artivicial traffic generated. The exact labelling of these repeaters is also not critical.

One of the reasons leading to observation 2 is that the station has a higher probability then the first level repeaters of successful transmission over the critical hop, because it is the largest user and does not interfere with its own transmissions. Theoretically, one may expect a similar conclusion when considering transmission in a section of the network in which two repeaters, one of which "homes" on the other, compete. This, however may not be realized in the system simulated because of the limited storage available in repeaters.

Observations 2 and 3 suggest a change in the Terminal-Station protocol. The basic question is whether a terminal should release itself from the system or whether it should be released by the station. The former was initially simulated. It was observed that in many cases, a terminal departed from the system after receiving an Echo to the ETE Ack for the last IP without this ETE Ack arriving at the station. This resulted in the reactivation if IP's by the station for this terminal, the routing of these packets in the net, and then the maximum number of transmissions and search by the repeater nearest to the terminal. The protocol simulated in the systems discussed later is such that the last packet must always be from the station to the terminal. This transmission may be considered as a terminal release packet. Another change in protocol implemented is that whenever possible, the terminal acknowledges a sequence of packets rather than individual ones, to reduce the overhead in the direction towards the station.

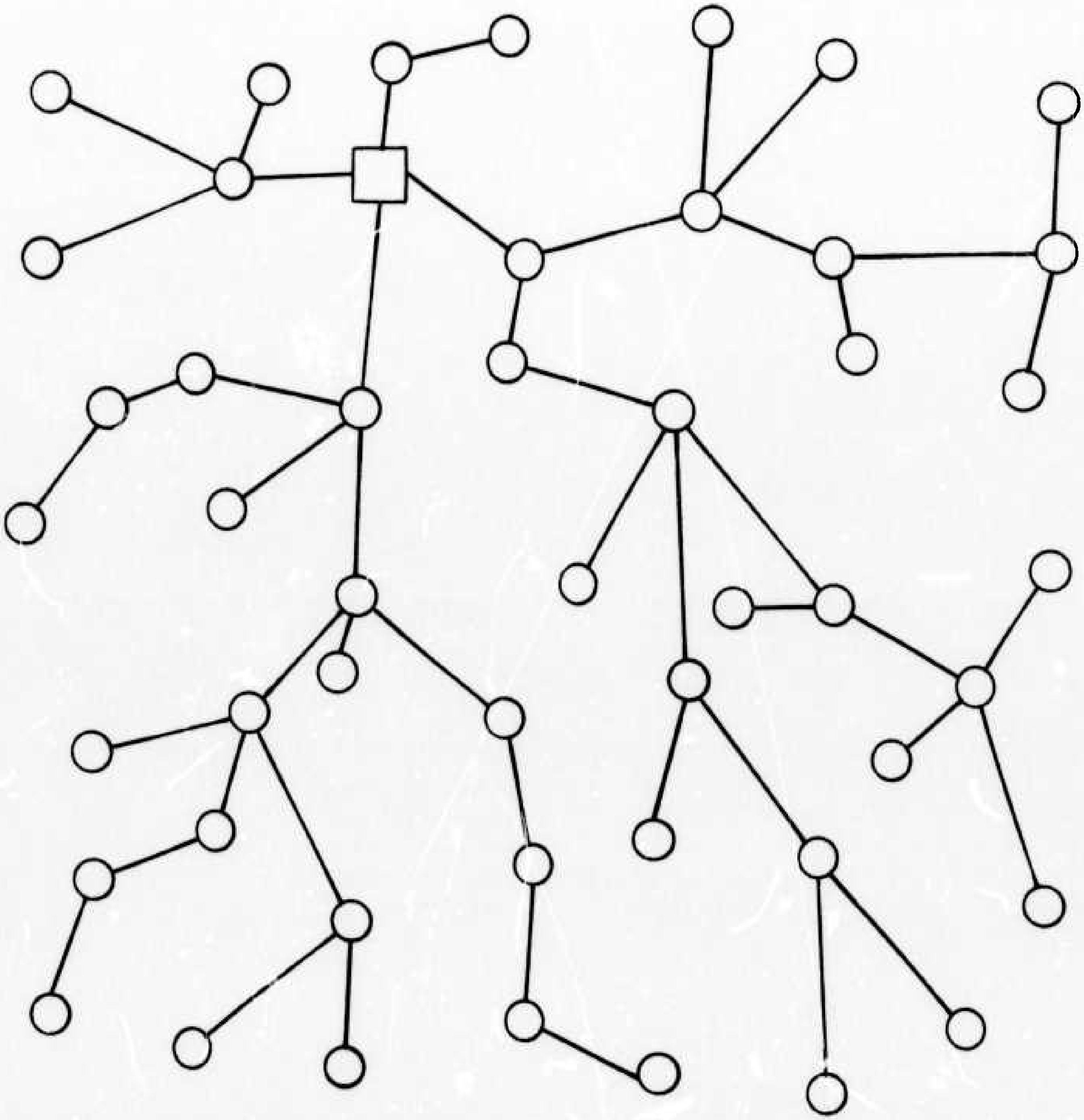

FIGURE 7

HIERARCHIAL LABELLING SCHEME

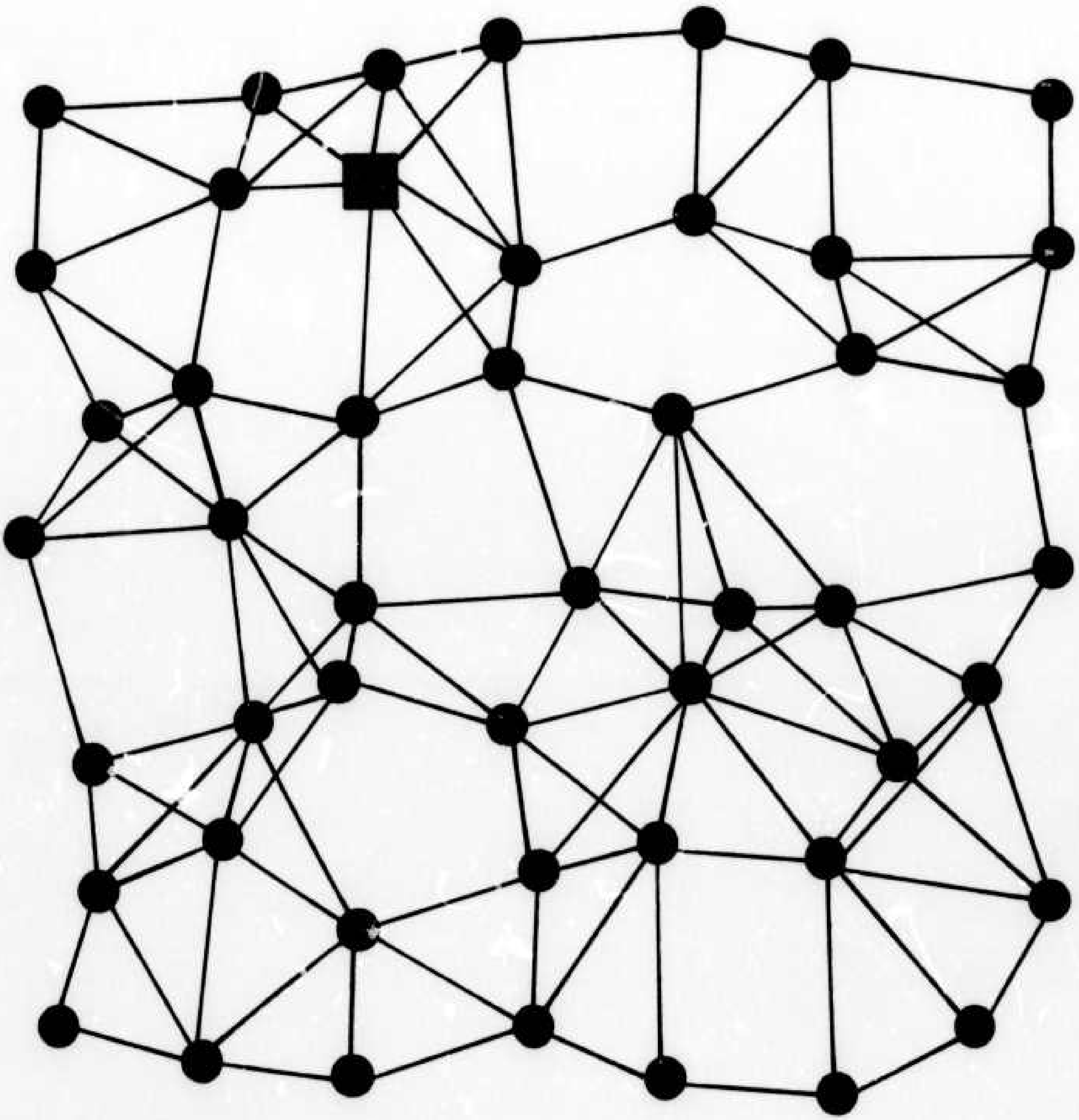

FIGURE 8

CONNECTIVITY OF REPEATERS & STATIONS

X. THE TRADEOFF BETWEEN TRANSMISSION RANGE OF DEVICES AND NETWORK INTERFERENCE

For the experiments discussed in the previous section, it was assumed that Repeater-Repeater range is the same as Terminal-Repeater range. This, however, is not always a realistic assumption since repeaters can be placed on elevated areas and can have more power then terminals, (especially hand held terminals). Thus, if repeaters are allocated for area coverage of terminals, the repeater range will be higher than terminal range and higher network connectivity or device interference will result.

The problem which then arrises is to determine the impact of this interference on system performance. Alternatively, one may seek to reduce repeater transmission power when transmitting in the repeater-station network. To study this issue, two CCSDR systems were simulated, one with high interference CCSDR (HI), and the other with Low Interference CCSDR (LI). The routing labels of the two systmes were the same and are shown in Figure 7. The interference of the CCSDR (LI) system is shown in Figure 8 and the interference of the CCSDR (HI) system in Figure 9. Figure 9 shows only the connectivity of two devices in the network.

The results are shown in Figure 10 and Table 1. Figure 10 shows the throughput of the two systems as a function of time while Table 1 summarizes all other measures of performance. The third row of Table 1 summarizes performance of the high interference system under an improved set of repeater labels. This experiment is discussed in detail in the next section. It is clear that the high interference system is much better than the low interference system. The only measure of the low interference system which is better is terminal blocking which is a direct result of the low interference feature. In fact, CCSDR (LI) is saturated at the offered traffic rate. This can be seen from the fact that the throughput is decreasing as a function of time; the relatively high total loss; and the low station response*. The CCSDR (HI) with improved labels compared in Table 1, is better than the other two systems. This demonstrates the importance of proper labelling. The experiments of this section demonstrate that it is preferable to use high transmitter power to obtain long repeater range, despite the network interference that it results.

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* The average number station response packets assumed for these studies is 2.0.

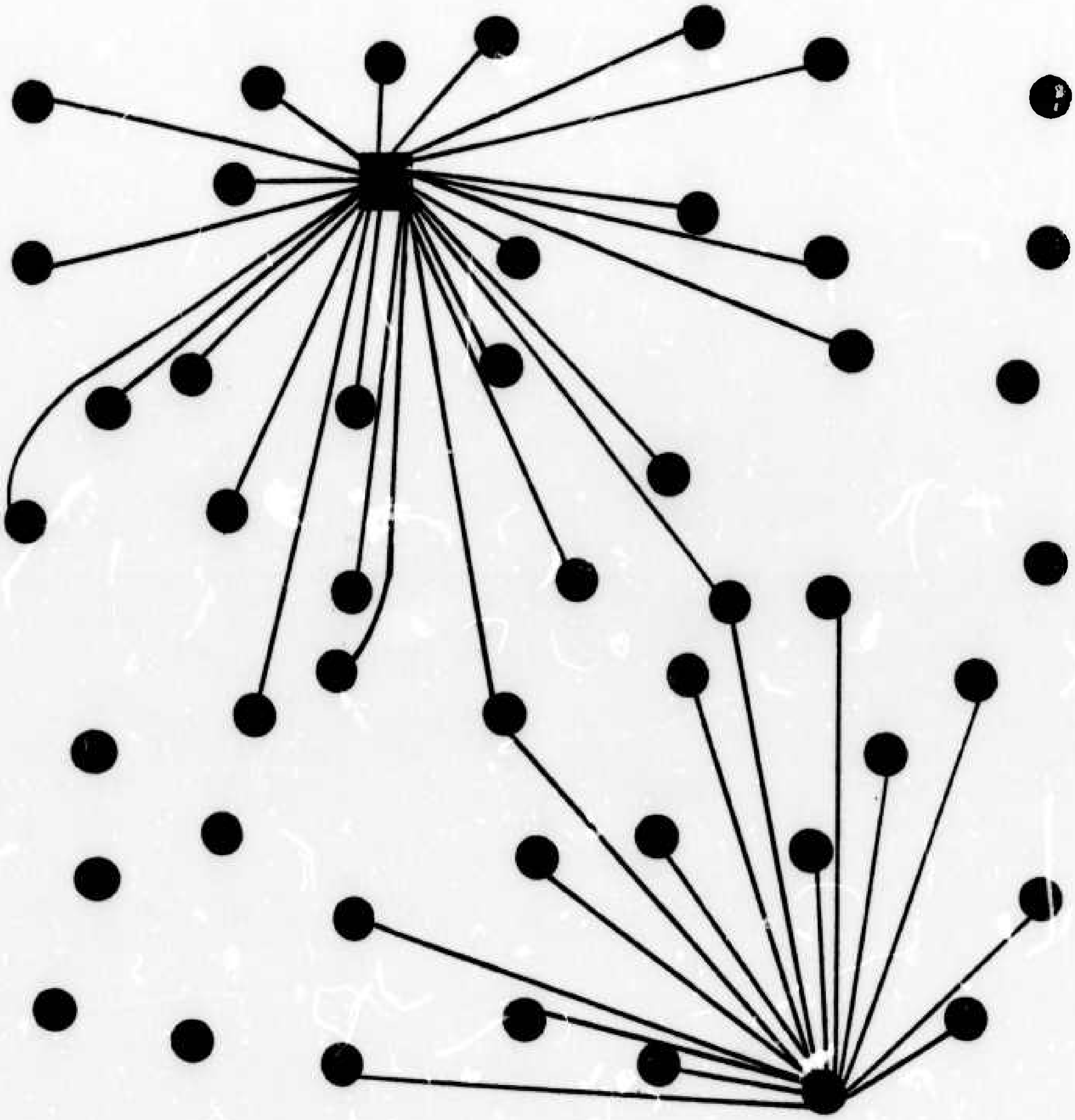

FIGURE 9

INTERFERENCE OF CCSDR (HI) SYSTEM

THROUGHPUT VS. TERMINAL SLOTS: CCDR (HI) & CCSDR

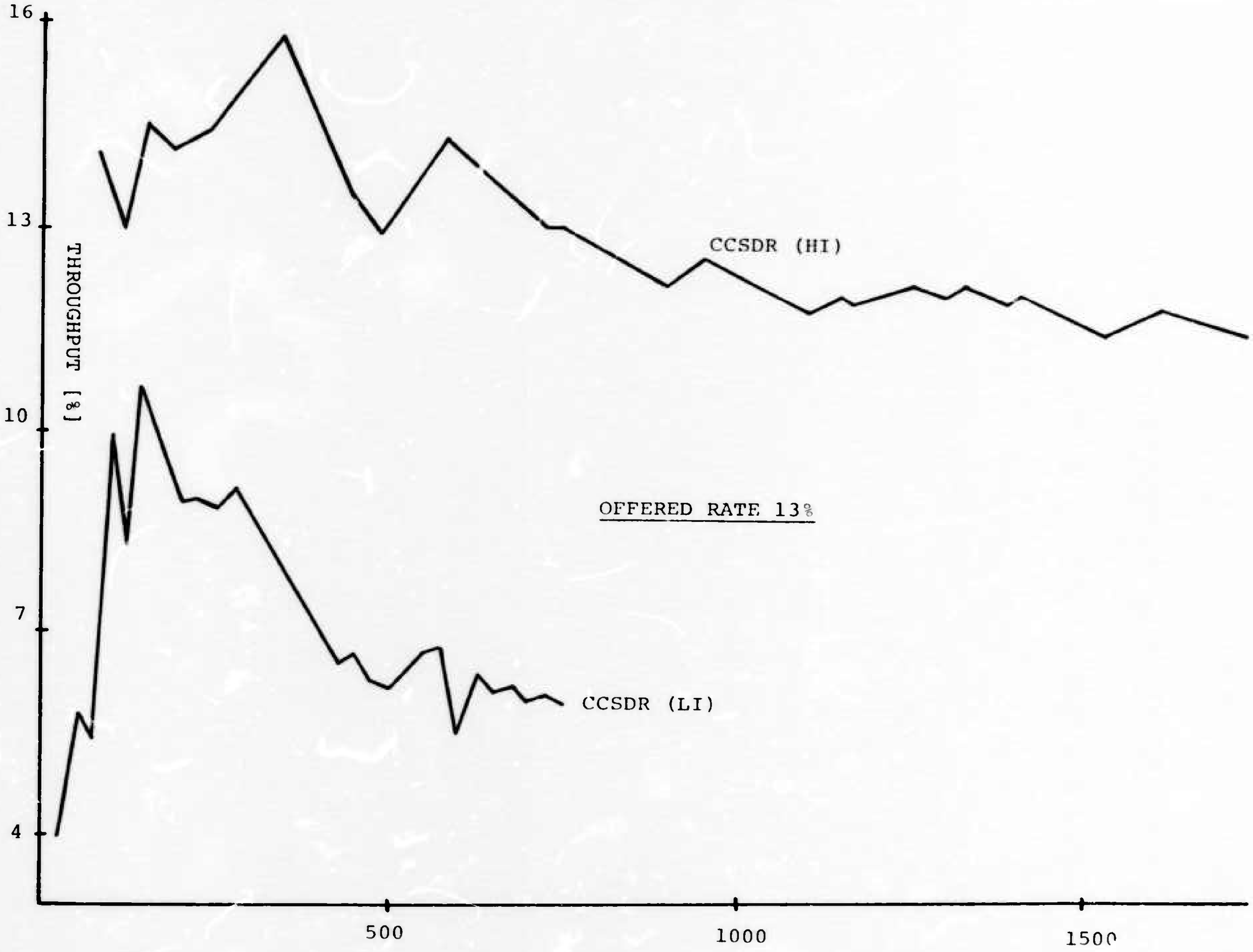

FIGURE 10

	OFFERED RATE [%]	THROUGHPUT [%]	DELAY OF IP TO STATION [Terminal Slots]	RATE OF STATION RESPONSE	PROB. STATION BUSY	% OF IP BLOCKED	TOTAL % OF IP LOSS	TERMINALS REMAINING
CCSDR (LI)	13	5.95	40.11	1.14	.53	2.98	32.53	13
CCSDR (HI)	13	10.55	23.93	1.81	.43	9.83	9.83	13
CCSDR (HI) (Improved Labels)	13	12.14	16.61	2.06	.50	10.63	11.41	10

TABLE 1

XI. SINGLE VERSUS DUAL DATA SIGNALLING RATES NETWORKS

The results of the previous section demonstrate that a better performance system is obtained when repeaters and station use high power to obtain long range despite the interference that results. We now examine the problem of whether repeaters and station should use their fixed power budgets to obtain a long range with a low data rate channel or have a short range with a high data rate channel. The following systems were studies.

- A CCSDR (HI) of the previous section with improved labels, which we denote by CCSDR. That is, we take advantage of the high range to improve the routing labels of repeaters and obtain fewer hierarchy levels. The routing labels used are shown in Figure 11, and the connectivity is shown in Figure 9.

- A Common Channel Two Data Rate (CCTDR) system with the routing labels as in Figure 7 and connectivity as in Figure 8.

In the CCTDR system, the terminal has a low data rate channel, the same rate as in the single data rate system, for communication with a repeater or station. Repeaters and station have two data rates. The high data rate is used for communication in the repeater-station network. The two data rates use the same carrier frequency so that only one can be used at a time.

The two systems are tested with offered rates of 13% and 25%.* The throughput as a function of time for the two runs are shown in Figures 12 and 13, respectively; and the summary of other measures is given in Table 2. The comparison demonstrates that the CCTRD

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* In the simulation runs we used the inverse square low for the relation between data rate and distance, rather than the result in [9];this however, favors CCSDR.

system is superior to the CCSDR system, in terms of throughput, delay, and other measures. One can see that the CCSDR system is saturated at an offered rate of about 13%.

- Effect on Blocking Level

In Table 2, one can see that one reason for the relatively low throughput of the CCSDR system at an offered rate of 25% is due to blocking. Furthermore, the fraction of time that the station is busy has decreased. This may suggest that the station may be able to handle more terminals providing they are able to enter the system. To examine this point, we ran the CCSDR system with offered rate of 25%, and relaxed the constraint for entering the system. Rather than resulting in better performance, this step resulted in reduction in blocking and increase in delay. The throughput increased to 12.63%, the blocking decreased to 18.35% and the total loss decreased to 30.73%. On the other hand, the delay increases to 57.82, the fraction of time the station is busy increased to .57, and the rate of station response decreased to 1.32.

To conclude, when we enabled more terminals to enter the system, the throughput increased insignificantly, from 12.20% to 12.63%; on the other hand, the average packet delay increased significantly, from 34.97 to 57.82 terminal slots. This suggests that one of the important design problems in the packet radio system is the blocking level of terminals.

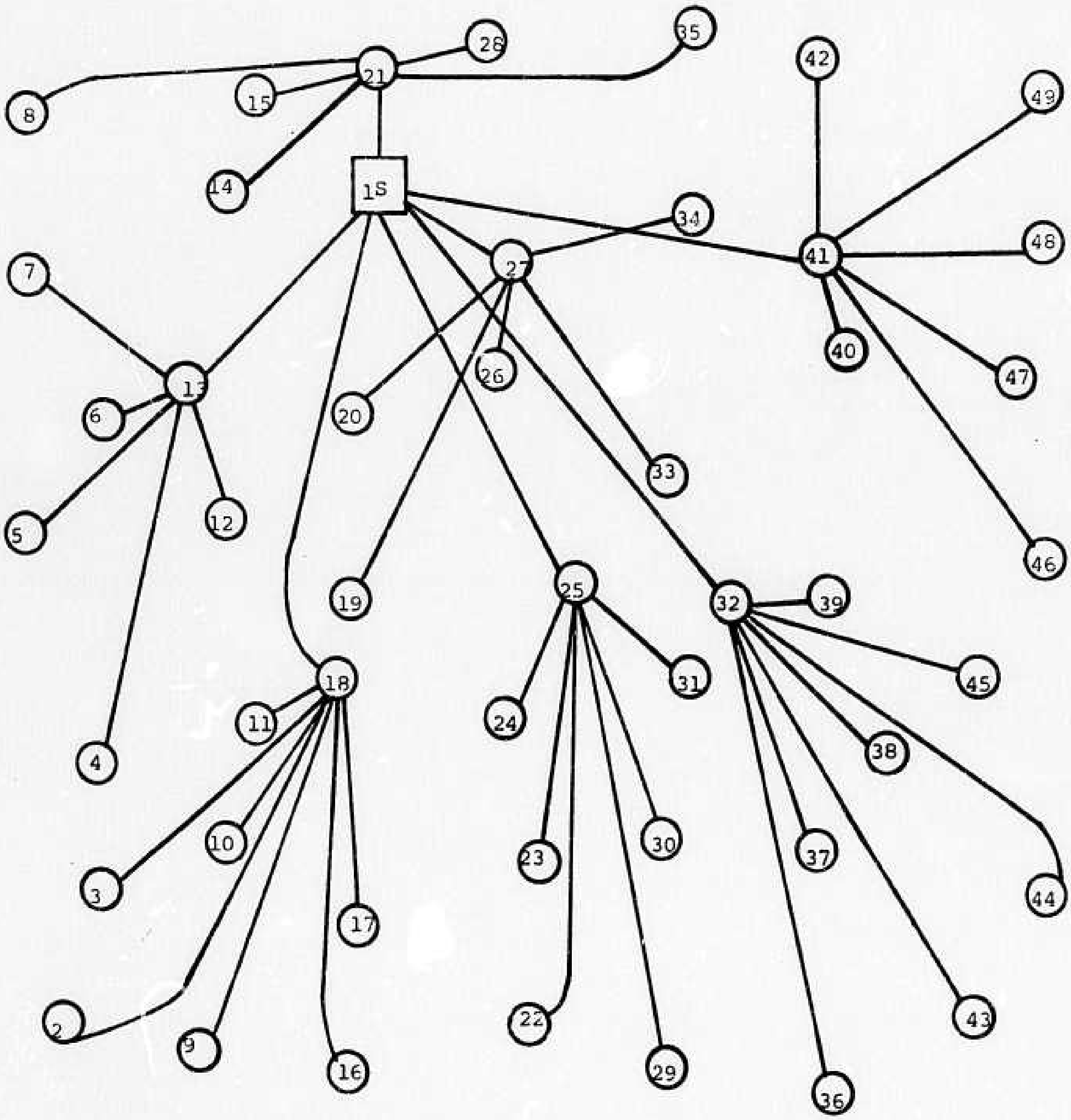


FIGURE 11

CCSDR (HI) SYSTEM WITH IMPROVED LABELLING

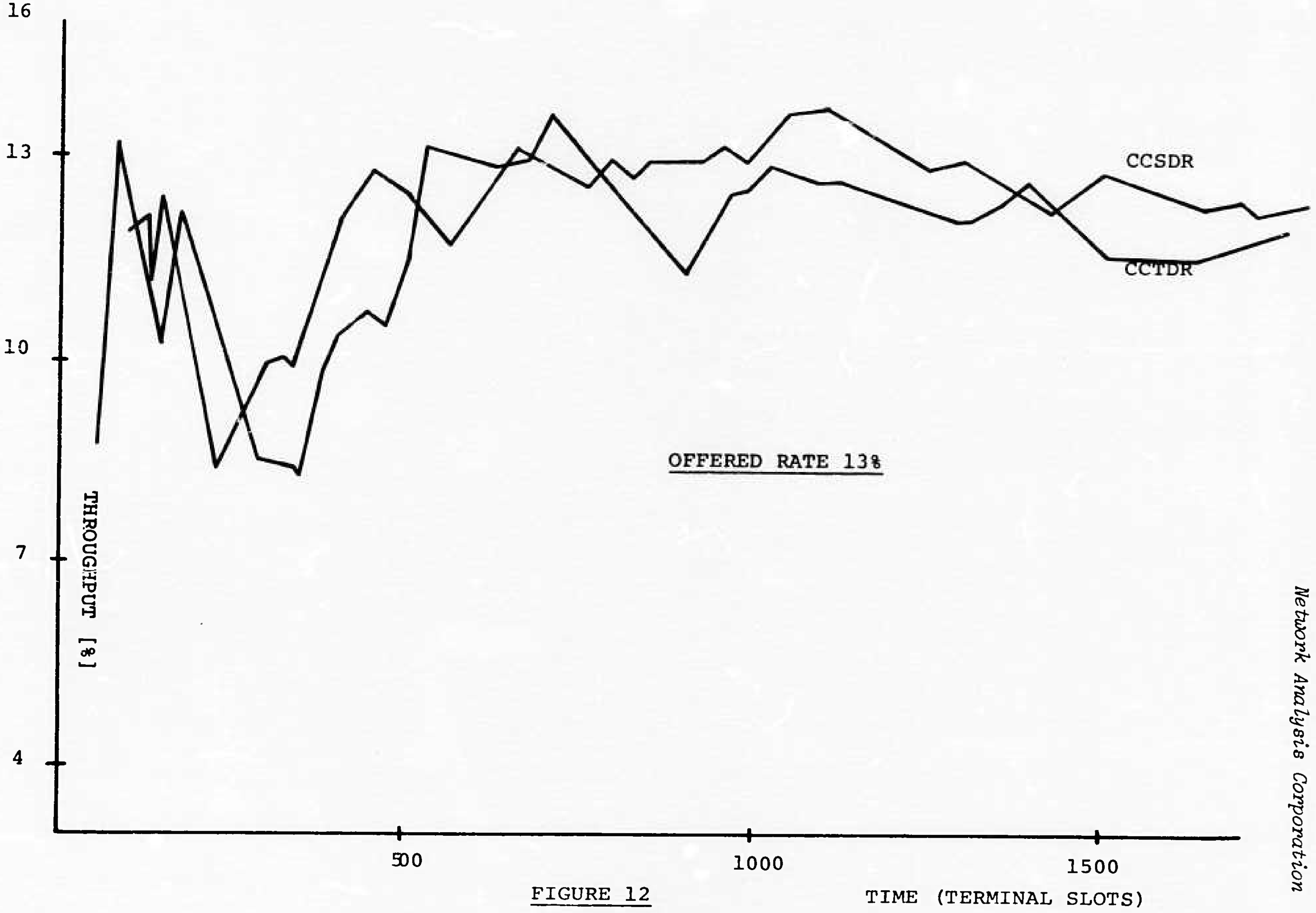


FIGURE 12

THROUGHPUT VS. TERMINAL SLOTS: 13% RATE

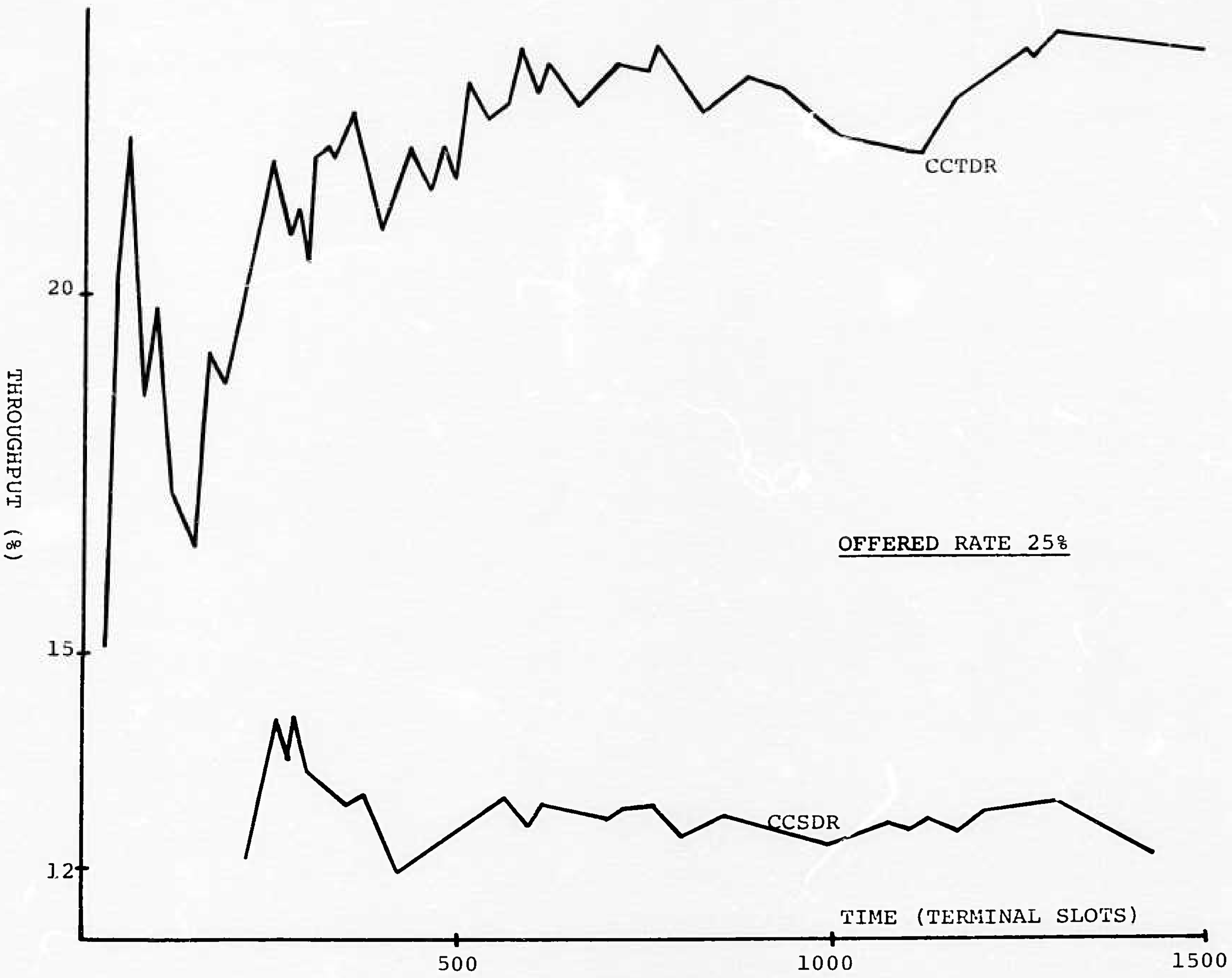


FIGURE 13

THROUGHPUT VS. TERMINAL SLOTS: 25% RATE

	OFFERED RATE [%]	THROUGHPUT [%]	DELAY OF IP TO STATION [Terminal Slots]	RATE OF STATION RESPONCE	PROB. STATION BUSY	% OF IP BLOCKED	TOTAL % OF IP LOSS	TERMINALS REMAINING
CCSDR	13	12.14	16.61	2.06	.50	10.63	11.41	10
	25	12.20	34.97	1.61	.48	29.50	32.95	23
CCTDR	13	12.39	4.91	1.99	.26	1.59	1.59	9
	25	23.33	11.51	1.97	.31	3.31	3.31	34

TABLE 2

XII. PRELIMINARY RESULTS OF MAXIMUM THROUGHPUT, LOSS, AND DELAY OF CCSDR AND CCTDR SYSTEMS

In the packet radio system there is an absolute maximum throughput (independent of loss and delay) because of the interference characteristics. Similar to curves of throughput versus channel traffic, when the relation is known analytically [7], we draw the curves of system throughput vs. offered rate for estimating the maximum throughput.Figure 14 shows the throughput versus offered rate for CCSDR and CCTDR systems. The curves are linear for low offered rates and saturate when the offered rate increases.

For the CCSDR system one can see that the throughput is practically the same when the offered rate is increased from 13% to 25%. This and the other measures (see Table 2), (for example, the rate of station response) show that the system is overloaded at a 25% offered rate. On the other hand, the system seems to operate at steady state at an offered rate of 13% (rate of station response 2.06). A rough estimate of maximum throughput for this system would be between 12% and 15%. Similar observations of the performance measures lead to an "estimate" of between 27% and 30% for the maximum throughput of the CCTDR system.

The average delay of the first Information Packet from terminal to station, and the Total Loss, as a function of offered rate are shown in Figure 15 and Figure 16, respectively.

Remark: There are many parameters in the simulation program which we have not experimented with (or tried to optimize) and which affect the quantities discussed above. One parameter which is significant in determining the maximum throughput is the average number of response packets from station to terminal. The affect of this parameter has been analyzed in [10] for a slotted ALOHA random access mode. It has been shown that the maximum throughput is increased in the Common Channel system when the rate of response increases, and the maximum throughput tends to 100% of the data rate when the rate of response tends to infinity. We expect that this parameter has a similar effect for the mode of access simulated. In the results reported here the rate of response is 2.0 which is small compared to usual estimates for terminals interacting with computers. Furthermore, the relatively short terminal interaction increases the traffic overhead of the search procedure, per information packet.

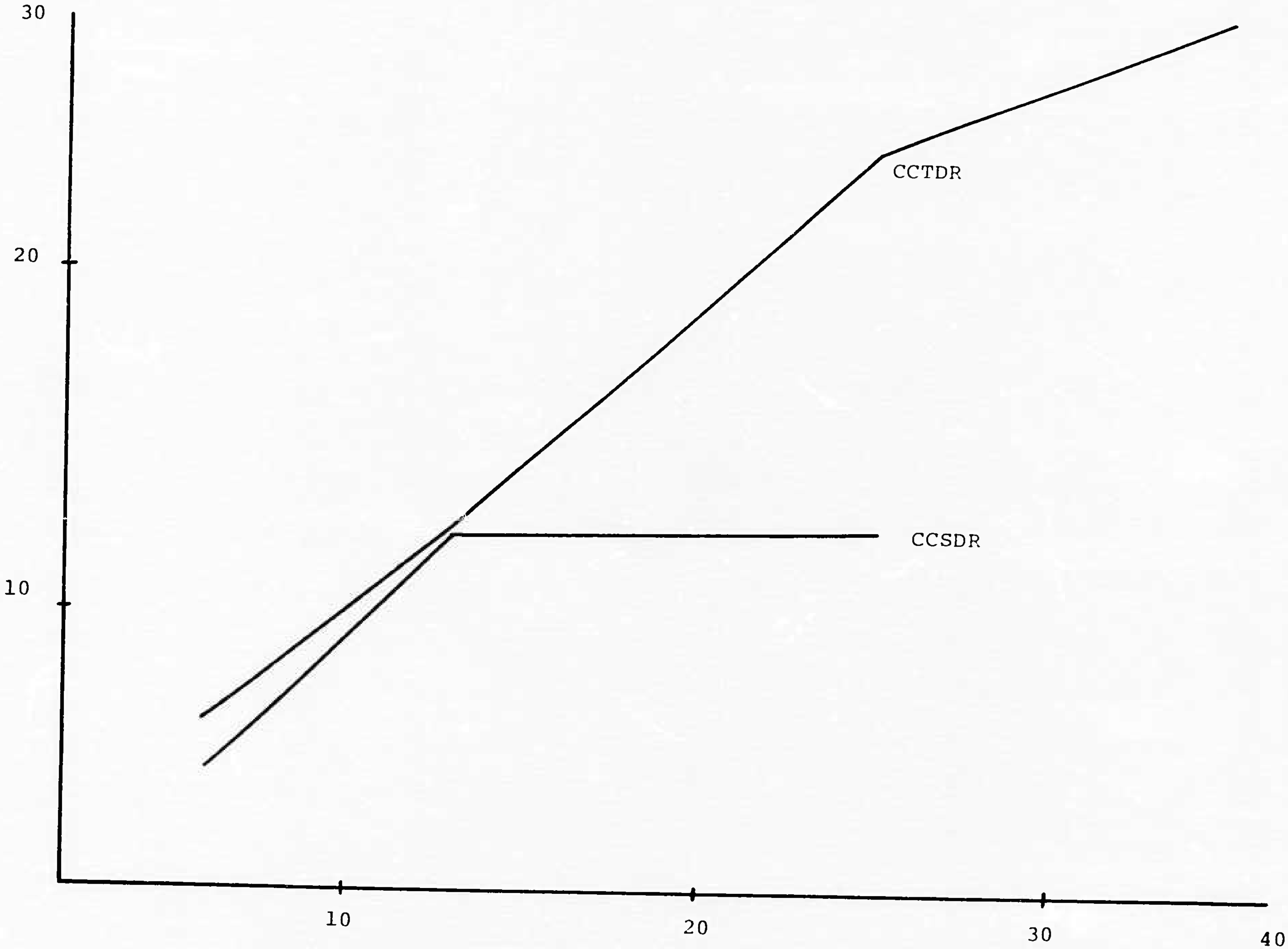


FIGURE 14

SYSTEM THROUGHPUT VS. OFFERED RATE

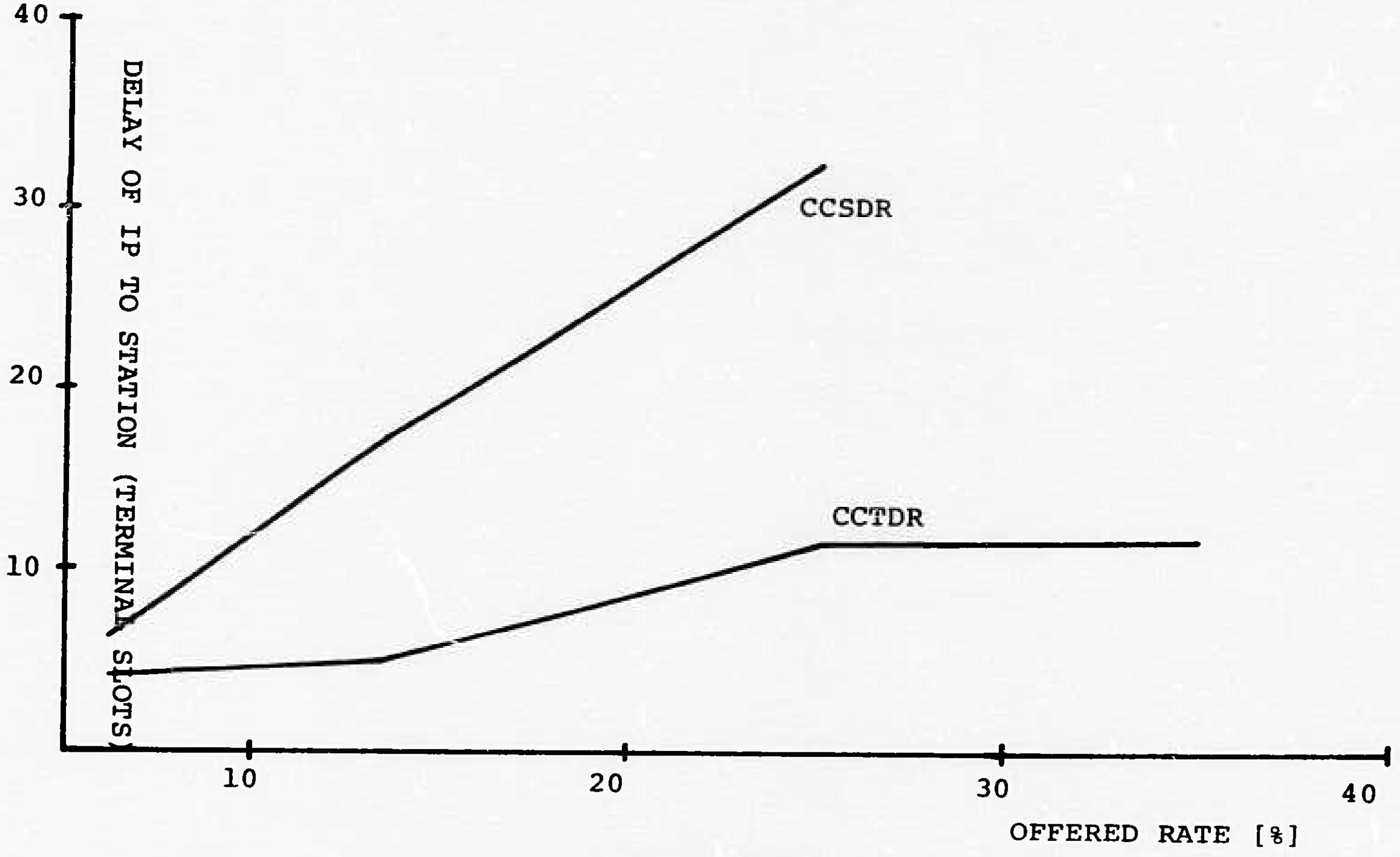
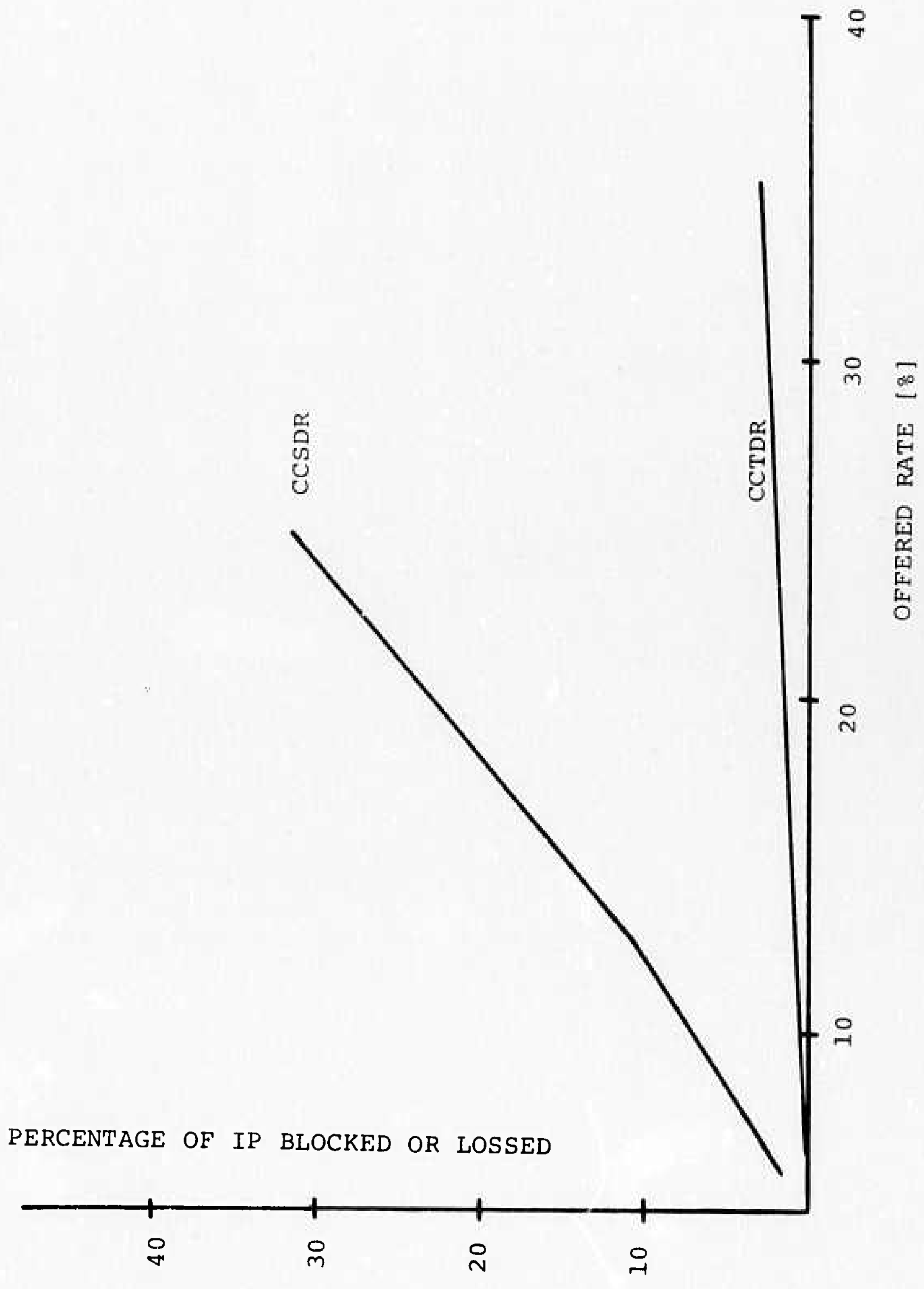


FIGURE 15

TERMINAL-STATION DELAY VS. OFFERED RATE



<u>FIGURE 16</u>

IP BLOCKING VS. OFFERED RATE

XIII. FUTURE DEVELOPMENT OF THE PACKET RADIO SIMULATOR

We outline several of the future developments for the simulator; some of these are in the implementation stage.

- Initialization and Labelling of Repeaters

Preliminary experiments have shown that the Hierarchical Labelling algorithm is much more efficient than the other two algorithms of Section VI, subsection B. Consequently, it is recommended for implementation in the Packet Radio System. In many cases however, the connectivity between devices in the network will not be known apriori. For example, in a military application one may wish to establish a network by distributing repeaters at random locations, and one may not have physical access to the repeaters. Furthermore, there may be changes in the "topology" of the network due to variations in transmission power of devices, or when some devices cease to operate. Thus, it is necessary to assign and reassign labels to repeaters in an operating network.

The approach that we adopted is to use the flooding routing algorithm to load repeaters with hierarchical labels. The flooding algorithm was selected because it does not require any knowledge of the topology of the network. A process for repeater initialization and labelling which has been detailed in [6] is currently under implementation. Initially, it is assumed that the station contains a set of fixed identifiers of repeaters which may possibly be connected into a network; three stages are then followed. In stage 1 the station transmits special control packets to the above repeaters, and repeaters respond with control packets from which a connectivity matrix between repeaters is established. The hierarchical labels are determined in stage 2 from the connectivity matrix. In stage 3 the station transmits the labels to repeaters and tests each path in both directions, from station to repeater and from repeater to station.

- <u>Flow Control</u>

Control packets for changing the operating parameters of devices, and algorithms for using these will be implemented. For example, turning repeaters "on" and "off", changing the parameter for the maximum number of transmissions, etc.

- <u>Access Modes</u>

In [7] it has been shown that one of the important parameters which affects the performance of the carrier sense access modes is the ratio of the propagation time between devices to the packet transmission time. Specifically, that the performance (relative) deteriorates when the above ratio increases; which is the case when the data signalling rate is increased and the number of bits in an information packet is kept constant. Thus for some operating parameters the carrier sense access modes may not show a much better performance than the more simple nonslotted ALOHA [11] random access scheme. These problems will be studied in a network environment by simulating the latter, for comparison, and by studying the carrier sense performance as a function of the data signalling rate.

- Directional Antenna at the Station

Analysis has shown [12] that directional antennas at the station may increase the system capacity. This can possibly be verified and quantified by simulating such an antenna.

- Capture

Currently the non-capture system is simulated. Caputre models which reflect the practical performance of hardware are under development and will be simulated.

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PACKET RADIO SYSTEM CONSIDERATIONS - NETWORK CAPACITY TRADEOFFS

I. INTRODUCTION

Packet Switching over radio channels with random access schemes is of current interest for local distribution systems and for satellite channels. This mode of operation is useful when the communicating devices are mobile and when the ratio of the peak to average data rate requirement of each device is high. Such systems have been analyzed for the case in which all communicating devices are within an effective transmission range of each other; either directly or through the satellite. These analyses were originally done for the ALOHA system [1] and for a satellite channel [4]. The models used do not sufficiently describe the Packet Radio System. The reason being that in the packet radio system there is a network of repeaters which separate an originating device from a destination device (terminals and stations).

In this chapter, we address broadcast networks in which orginating devices cannot directly reach the destination receiver. Thus, repeaters are introduced which receive these packets and repeat them to the destination. The capacity (maximum throughput) of such systems is determined, and design problems related to the number of repeating devices and the usefulness of directional antennas are resolved.

The model used in this chapter can describe systems other than the packet radio system and we discuss it in the more general context.

One way to categorize channel allocation schemes for data transmission is the following:

1. Fixed assignment (FDM, TDMA)

2. Dynamic assignment with centralized control (polling schemes, reservation upon request)

3. Dynamic assignment with distributed control (loop networks, random reservation schemes)

4. No assignment (random access schemes)

It has been recognized that a fixed allocation of channel capacity is extremely wasteful when the traffic of users is of a bursty nature; that is, when the traffic requirements of the users can be characterized as having a high peak to average data rate. Users can be so characterized in an inquiry response application. In fact, if one characterizes a set of users by the number and by the ratio of the peak to average data rate requirements of each user, one can conjecture that when the above number and ratio are increasing, one obtains a higher channel utilization when proceeding (along the categorization) from the fixed assignment to the no assignment allocation schemes. For example, when the time delay to make a reservation or the average time between two conseqtutive pollings of a user is large compared to the fraction of time that the user wishes to use the channel, then the dynamic assignment with centralized control, becomes inefficient (apart from the need for a system for polling or making reservations).

Roberts [7] has demonstrated the cost advantages of a random reservation scheme and a random access scheme over fixed assignment, when the number of users increases and the average traffic requirement per user is kept constant. A somewhat more formal justification for sharing a channel was given by Kleinrock and Lam [4], we quote: "Rather than provide channels on a user-pair basis, we much prefer to provide a single high-speed channel to a large number of users which can be shared in some fashion; this when allows us to take advantage of the powerful 'large number laws' which state that with very high probability, the demand at any instant will be approximately equal to the sum of the average demands of that population." Gitman, Van Slyke, and Frank [3], have addressed the problem of splitting a channel between two classes of users. It was shown that in almost all cases, sharing the channel results in a higher utilization of the total channel capacity.

In this chapter, we consider a packet switching network in which a single radio channel is shared by all communicating devices. Devices access the channel using the so called "slotted ALOHA" random access scheme. When a random access scheme is used, there is a possibility that more than one packet is simultaneously received by a receiver due to independent transmissions of several devices. In that event, it is assumed that none of the packets are correctly received, and the corresponding devices have to retransmit their packets. One can see that the number of packets transmitted is larger than the number of originating packets. That is, part of the channel capacity is used up by the wasteful collisions and are not considered as effective channel utilization since it does not contribute to the throughput. For example, if each packet is transmitted two times, on the average, before it is successfully received, then the maximum effective utilization of the channel (or the effective channel capacity) is one-half of the given capacity, the other one-half is used up by the non-successful transmissions. The first problem that one faces is to determine the maximum effective utilization (or system capacity) that can be obtained. This is one of the problems addressed in the chapter.

If the channel is offered a higher rate of traffic than its effective capacity, the system becomes unstable in the sense that the number of transmissions increases with time and the throughput decreases with time until zero throughput is obtained. This implies another problem in random access schemes and that is, the control of the offered rate and retransmission strategies so as to obtain the highest possible channel utilization [5], [8]. It is clear however, that retransmissions have to be randomized in time since otherwise once a collision between two devices occurred, it will persist.

The first packet radio channel system where devices use a random access scheme, known to us, was analyzed and implemented at the University of Hawaii [1]. The random access scheme used is the so called "pure" or "unslotted" ALOHA. In this scheme, every terminal transmits its packets independent of any other terminal or any specific time. That is, the terminal transmits the whole packet at a random point in time; the terminal then times out for receiving an acknowledgement. If an acknowledgement is not received, it is assumed that a collision occurred and the packet is retransmitted after an additional random waiting time. Abramson [1] obtained that the capacity (effective) of the channel is 1/2e of the given capacity when the number of terminals is very large and when the point process of the beginning of packet transmission onto the channel is Poisson.

It was realized that a gain in capacity can be obtained if the channel was slotted into segments of time whose duration is equal to the packet transmission time, and when terminals are required to begin the transmission at the beginning of a time slot. The access scheme is random in the sense that terminals transmit into a random slot in time and retransmit after waiting a random number of slots. This scheme is called "slotted ALOHA." Roberts [6] has shown that the capacity of this scheme is 1/e of the given capacity, using the same assumptions as Abramson. The slotted ALOHA random access scheme was further analyzed in [2], [3], [4], and [5] for the case of a small number of terminals and when there is a mixture of traffic from a small number of "big" users and a large number of "small" users.

II. PROBLEM DESCRIPTION

In a network context, the analyses of all the references address a "single hop network." That is, when considering a terrestrial system, it is implicitly assumed that all devices are within an effective distance of each other; the same is true when considering a satellite channel, since the satellite echoes the packets to the destination station. Such a network can be described as one in which there is a single receiver and many transmitters, all of which are within an effective transmission range to the receiver, and to each other.

In cases in which the transmission range of terminals is not sufficient to reach the destination receiver, it is necessary to introduce another device which will receive the packets from the terminals and repeat them to the final destination. Such a network can be used for local distribution and collection of traffic, in which case the station is a gateway to a point-to-point network. We particularly consider a 2-Hop network model in which there is a large number of terminals in the neighborhood of each repeater and that the transmission range of terminals is short so that a terminal can reach only one repeater, as shown in Figure 1. Our model can be useful as a distribution model for a suburban area, where instead of supplying each terminal with a powerful transmitter, we allocate repeaters which collect the traffic from terminals.

In a military application, one can think of a large unit which uses such a 2-Hop network, where each subunit (the terminals of which are relatively close to each other and may be mobile) has its own repeater, and the station is at the Headquarters. In fact, the network can be operative when the whole unit is moving, providing each subunit carries its repeater or station along with it, e.g., a fleet of ships.

This network can also model a satellite case as shown in Figure 2. In the analyses of a satellite channel reported, it was assumed that each terminal is a ground station which originates its traffic or which is connected to a point-to-point network. However, suppose there are clusters of (possibly mobile) terminals, where in each cluster, they are relatively close to each other, and wish to communicate with a remote central computer installation. Then, one can devise a Ground Station Repeater to communicate with the terminals and the ground station repeater may use a satellite channel to transmit (second hop) to the computer installation.

Before we specify the problems to be addressed, we comment on the operation of the repeater. We have indicated that a terminal retransmits its packet after a time out if it does not receive an acknowledgement. The repeater can operate in a very simple manner in which it repeats only once a packet that it correctly receives. In this case, the terminal has to time out for a longer period of time to wait for an acknowledtement from the station (end-to-end ack). Alternatively, the repeater can be made responsible for the successful transmission on the second hop, by acknowledging the terminal, storing the packet, and transmitting it until it receives an acknowledgement from the station (hop-by-hop acknowledgement). This problem has been considered in [9] where it was shown that a hop-by-hop acknowledgement operation is more efficient. Thus, we will assume a repeater of that type.

The first problem that we address is to determine the network capacity as a function of the number of repeaters and the interference between repeaters. Another problem of interest, is to determine the capacity bottleneck (critical hop); i.e., whether the capacity bottleneck is on the hop from terminals to repeaters or on the hop from repeaters to station. The system is assumed to have two channels; one channel is used for transmission from terminals via repeaters to the station and the second channel for transmission from the station via repeaters to the terminals. The traffic of acknowledgement packets is not considered; which implicitly assumes that there is a separate channel for acknowledgements. The two channels are analyzed separately and design questions such as the following are considered: Is it useful to have directional antennas at repeaters when transmitting to the station ? This question is relevant in the terrestrial system, since in the satellite system, the ground station repeaters will presumably be out of range from each other, and use directional antennas. Similarly, in a terrestrial system, one may ask about the usefulness of using a directional antenna at the station when transmitting to repeaters. Other questions relate to the possibility of using several transmitters and antennas at the station when transmitting to repeaters.

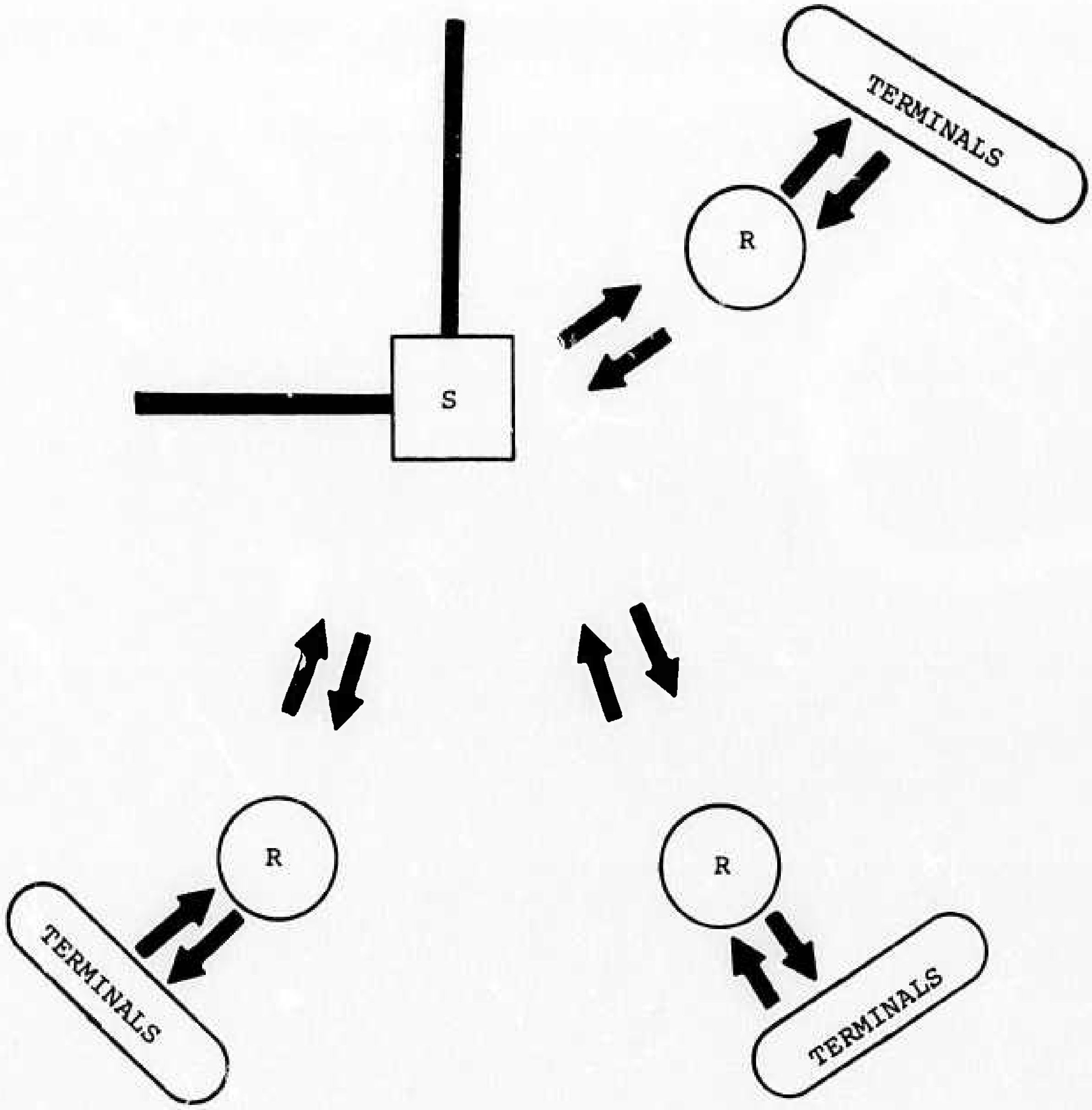


FIGURE 1: A TERRESTRIAL SYSTEM

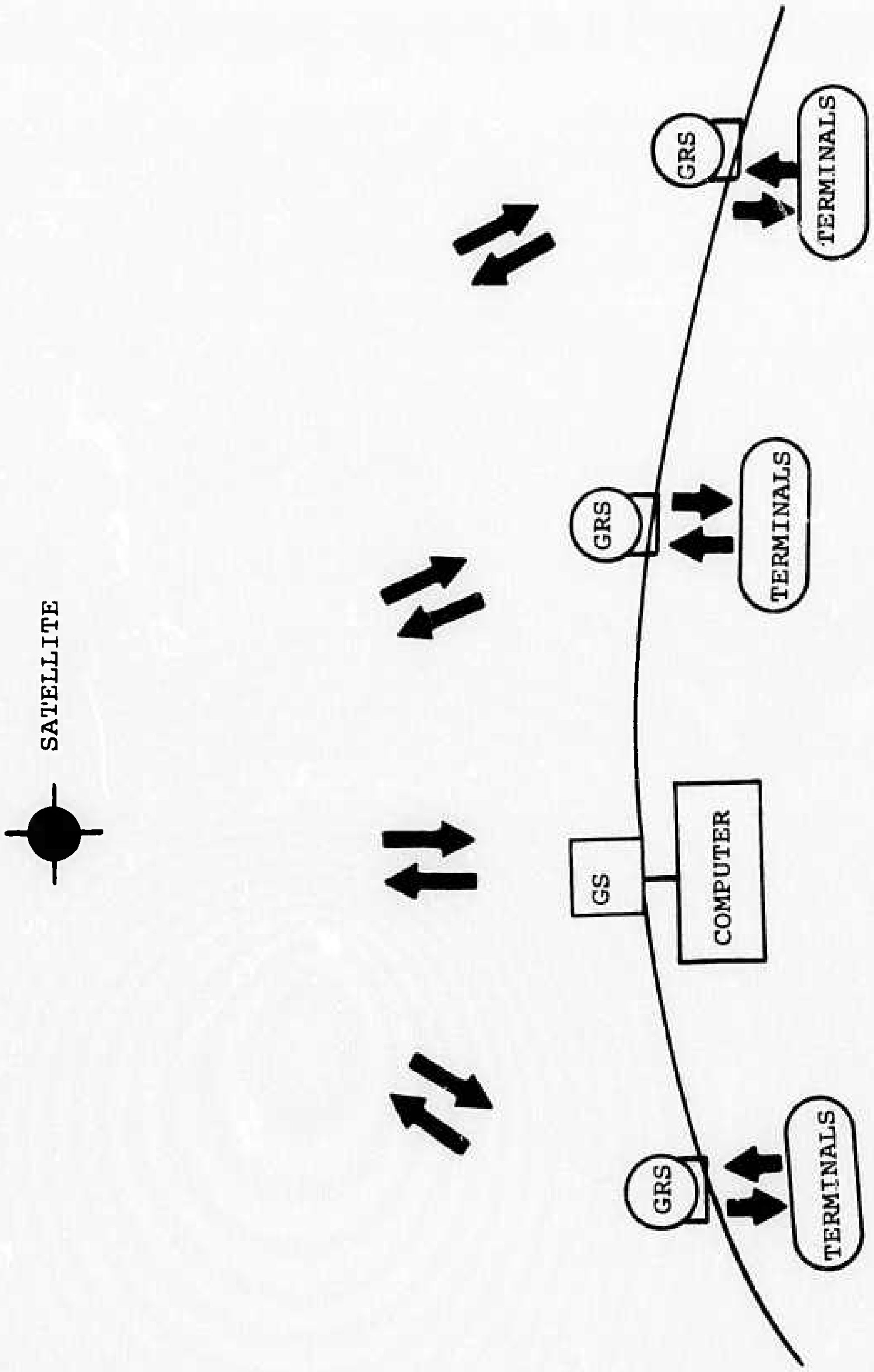


FIGURE 2: A SATELLITE SYSTEM

III. TRANSMISSION FROM TERMINALS TO STATION

Consider a system where m repeaters receive packets from terminals and repeat the packets to a single station, as shown in Figure 3. We donote by G and S the rate of packet transmission per slot and the rate of successful packet transmission per slot, respectively. Specifically, let G_{1i} and S_{1i} be the rates of transmission from terminals to repeater i, and G_{2i} and S_{2i} the rates from repeater i to the station. We wish to obtain the probability that a repeater is idle.

A single hop network is the case in which a set of terminals transmit to a repeater and the repeater is the final destination and does not repeat the packets. Thus, the probability that a transmission from a terminal to the repeater is successful is the probability that no other terminal transmits into that same slot. That is, if we assume that a packet is transmitted into the first slot after it becomes ready for transmission then the probability of success is the probability that no new packet has arrived and that no other packet has been scheduled for retransmission in the interval of time of the preceeding slot. In the network case however, a transmission from a terminal to a repeater will not be successful also in the case when the repeater uses the same slot for transmitting to the station, or if another repeater, within an effective transmission range of the first and which uses an omnidirectional antenna, uses the same slot for transmission to the station.

Throughout the chapter, we use the following assumptions. The combined point process of packet origination and packets scheduled for retransmission, from each set of terminals to a repeater, is Poisson. Thus, the probability of no arrival during a slot time τ is $e^{-G_{1i}\tau}$; we use $\tau \equiv 1$. The probabilities of transmission by a repeater into different slots are independent. The probability of

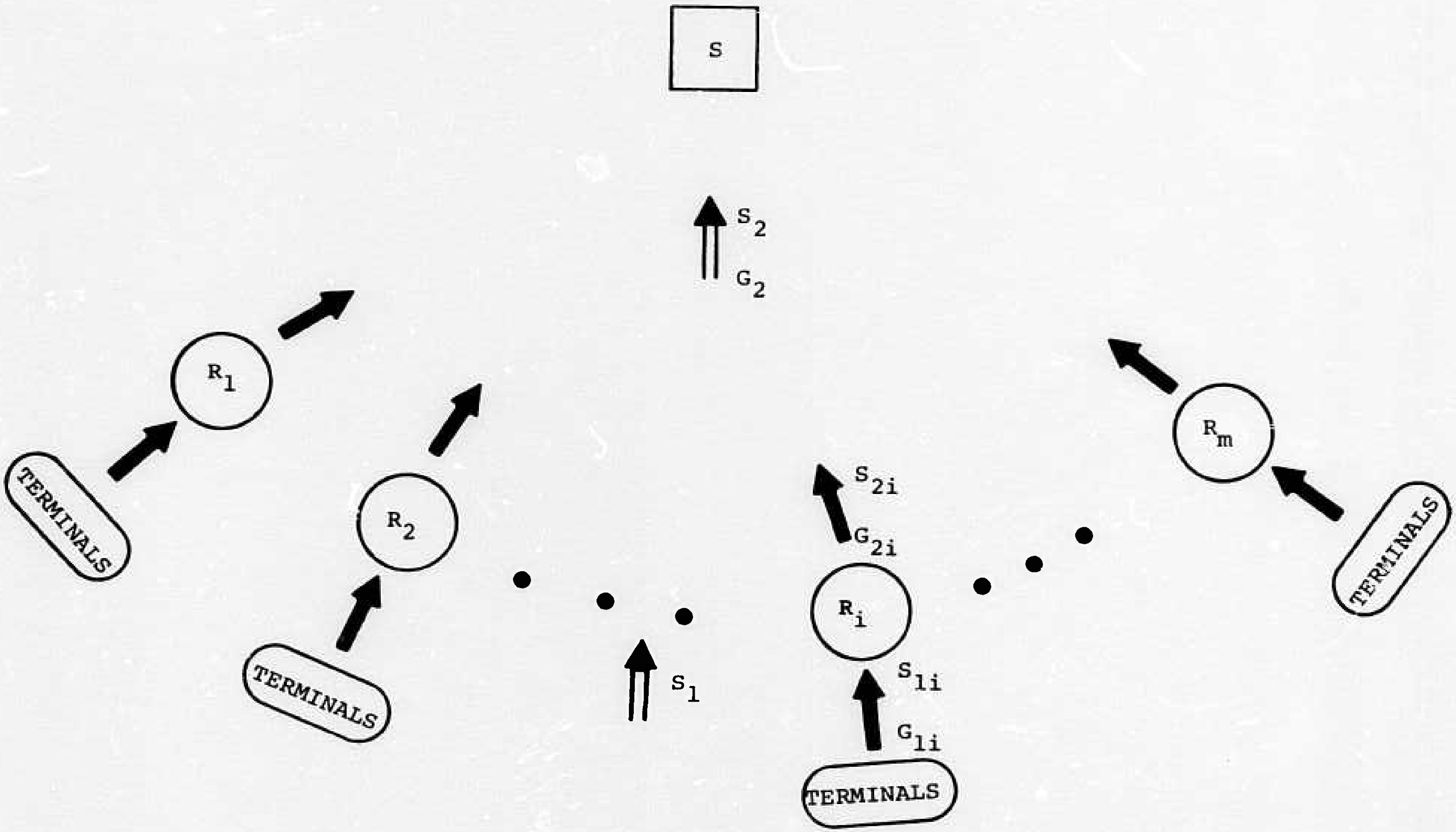


FIGURE 3: TRANSMISSION FROM TERMINALS TO STATION

transmission by two or more repeaters into a randomly chosen slot are mutually independent; and the probability of transmission into a random slot by a terminal and by a repeater are independent. The assumption of a Poisson distribution for the process of packet origination plus retransmission has been questioned in previous publications. The validity of this assumption is important when one considers packet delays, since if one assumes a Poisson point process for packet originations, then the assumption that the combined process, of origination plus retransmissions, is Poisson is valid only when one allows very large packet delays. In this chapter, we do not require finite delays and our interest is in the ultimate system capacity.

Finally, let Q_i denote the set of repeaters which have an effective transmission range to repeater i; Q_i includes repeater i. Under these assumptions we can write:

$$P_r[\text{repeater } i \text{ is idle}] = e^{-G_{1i}} \prod_{j \varepsilon Q_i} (1-G_{2j}) \qquad (1)$$

Similarly, the probability that the station is idle can be written as:

$$P_r\ [\text{station is idle}] = \prod_{j=1}^{m} (1-G_{2j}) \qquad (2)$$

We now make a few assumptions which simplity the computation but enable us to answer the questions of interest. Specifically, we assume that $G_{1i} = G_1$, $S_{1i} = S_1/m$ for all i, that repeaters share equally the load so that $G_{2i} = G_2/m$ and $S_{2i} = S_2/m$ for all i, and that all repeaters have the same number of interfering repeaters, $n\ (Q_i) = I$ for all i. We refer to I as the interference level.

We can now write the throughput equations for hop 1 and hop 2:

$$S_1 = \sum_1^m G_1 \left(1-\frac{G_2}{m}\right)^I e^{-G_1} = mG_1 \left(1-\frac{G_2}{m}\right)^I e^{-G_1} \qquad (3)$$

$$S_2 = \sum_1^m \frac{G_2}{m} \left(1-\frac{G_2}{m}\right)^{m-1} = G_2 \left(1-\frac{G_2}{m}\right)^{m-1} \qquad (4)$$

Note that when m=1 all transmissions from the repeater to the station are successful since it does not interfere with its own transmissions.

If a repeater were a traffic source and a traffic sink, then G_1 and G_2 could have been considered as independent variables. In our case, however, the intensity of the processes on the two hops are related. In particular, we consider traffic rates in which the system can operate at steady state. That is, if one observes the system for a long period of time, then all the packets which successfully arrive to repeaters also successfully arrive to the station. Thus we can use the conservation law at repeaters, namely $S_1 = S_2$. This results in the relation:

$$G_2 = mG_1 \left(1-\frac{G_2}{m}\right)^{I+1-m} e^{-G_1} \qquad (5)$$

One can now study the system performance as a function of parameters m and I, with one independent variable.

A. Complete Interference System, I = m

This case is applicable to terrestrial networks in which repeaters are either placed relatively close to each other or use powerful transmitters; either of which results in the interference among all repeaters. For this case we obtain:

$$G_2 = \frac{m\, G_1 e^{-G_1}}{1+G_1 e^{-G_1}} \qquad (6)$$

and

$$S_1 = \frac{m\ G_1\ e^{-G_1}}{(1+G_1\ e^{-G_1})^m} \tag{7}$$

The capacity of this system is given by the maximum of S_1.

$$\frac{d\ S_1}{d\ G_1} = \frac{m\ e^{-G_1}}{(1+G_1\ e^{-G_1})^{m+1}}(1-G_1)\ [1-(m-1)\ G_1\ e^{-G_1}] = 0 \tag{8}$$

By examining (8), one finds that there is one stationary point, a maximum at $G_1 = 1$, when $m<4$. When $m\geq4$ there are three stationary points; a minimum at $G_1 = 1$, and two maximum points of the same value at G which satisfies $1-(m-1)\ G\ e^{-G} = 0$. Substituting these values into (7), one obtains the capacity of this (complete interference) network as a function of m:

$$\text{Network Capacity} = S_1{}^* = \begin{cases} \dfrac{m}{e\ (1+\frac{1}{e})^m} & m<4 \\ (1-\frac{1}{m})^{m-1} & m\geq4 \end{cases} \tag{9}$$

Notice that the network capacity is lower than the capacity of a single hop network, 1/e, when m=1; it is higher than 1/e for m>1, and tends to 1/e in the limit when $m\to\infty$.

Figure 4 shows the network throughput S_1 as a function of the rate of transmission from terminals to a single repeater. It is interesting to observe the rate of change of S with respect to G. For example, when M is large (e.g. m=8) then the system becomes more sensitive to variations in the value of G. On the other hand, there are values of m, for example, m = 2 or 3 in the complete interference case, for which the maximum throughput region is quite flat.

B. The Critical Hop

To determine the critical hop, one has to obtain the capacity of the two hops of the network. The capacity of the hop from repeaters to the station is independent of G_1 and I (see Eq. (4)), and is given by $(1-1/m)^{m-1}$. The capacity of the hop from terminals to repeaters, on the other hand, depends on G_2 and I. Note that $0 \le G_2 \le m$; $G_2 > m$ is not realizable since we assume that a repeater has one transmitter and cannot transmit more than one packet per slot. For any G_2 in the above range, the capacity of hop 1 (see Eq. (3)) increases with m.

Thus, there exists an m_o for which the capacity is higher than that on the hop from repeaters to station, and the latter becomes the critical hop. Furthermore, m_o depends on I, and for $I_2 \ge I_1$, $m_o (I_2) \ge m_o (I_1)$. Thus, for $m > m_o$ the critical hop is that from repeaters to station, and for $m \le m_o$ the critical hop is from terminals to repeaters. For example, for I=m (see Eq.(9)) $m_o = 4$, and for $I = m-1$, $m_o = 3$.

Perhaps a more direct way to answer the question of the critical hop is to obtain the system capacity as a function of m and I. Then, whenever the capacity is smaller than $(1-1/m)^{m-1}$ the critical hop is that from terminals to repeaters and when the capacity equals this expression, the critical hop is from repeaters to station. However, it is difficult to obtain a closed form solution in the general case.

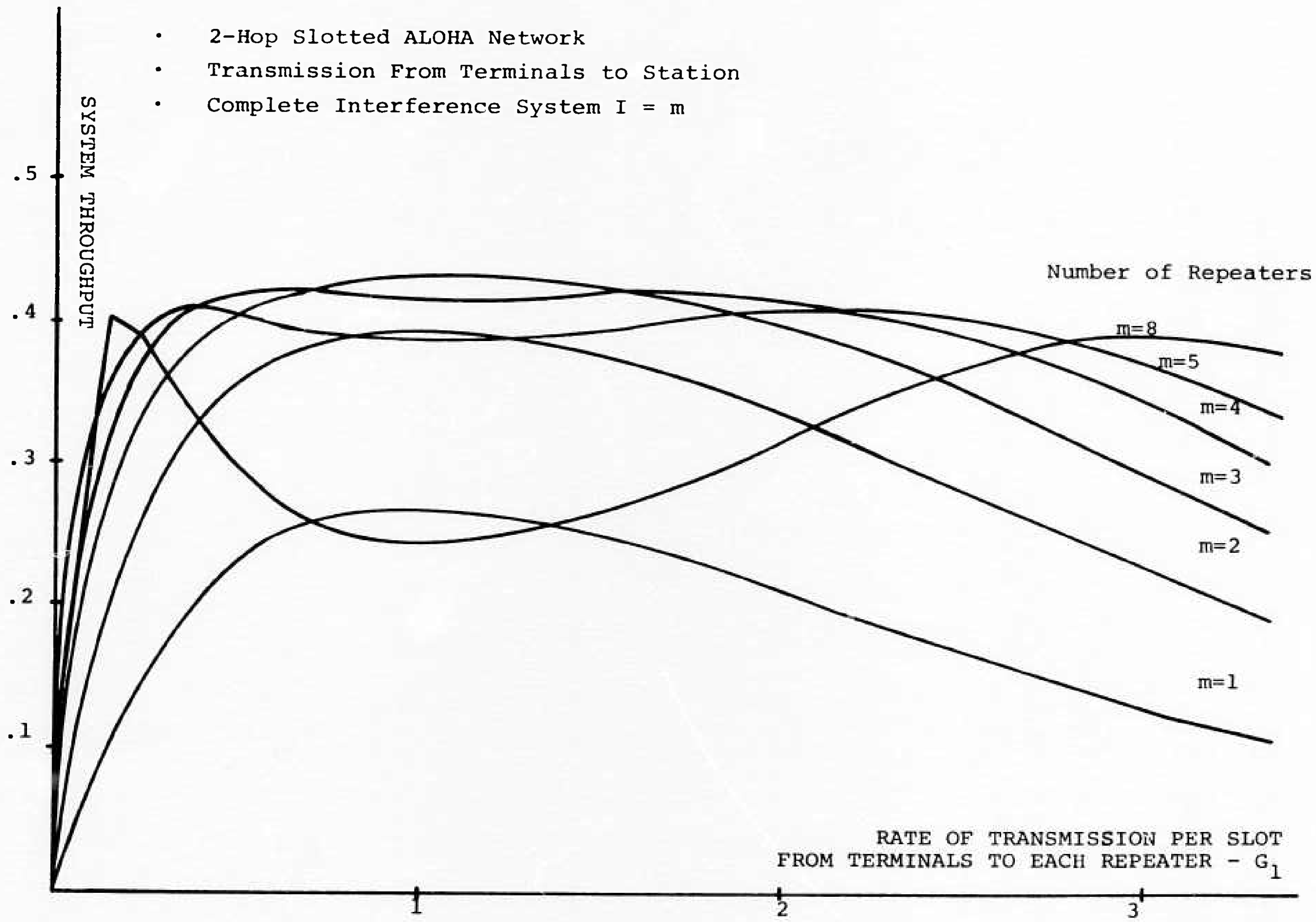
C. Number of Repeaters and Directional Antennas at Repeaters

The considerations of directional antennas at repeaters apply only to the terrestrial system. In the satellite system, ground station repeaters will use directional antennas when transmitting to the satellite and omnidirectional antennas when communicating with terminals.

The effect of directional antennas at repeaters in the terrestrial system is that the transmission from repeaters to the station is directed towards the station and does not interfere with the transmission of terminals to other repeaters. Thus, it is the special case with I=1. We notice, however, that directional antennas do not increase the capacity of the hop from repeaters to station because all antennas are directed towards the same physical location where the station is placed and where the conflicts may occur.

Figure 5 shows the capacity of the system as a function of m, for I=m and I=1, which is equivalent to omnidirectional and directional antennas (or satellite system) respectively. One can see that there is a gain in capacity when using directional antennas only when m=2, and a small gain for m=3. For $m \geq 4$ the capacity does not increase because the critical hop is between the repeaters and the station, so that it does not matter how much one can get through from terminals to repeaters.

As far as the number of repeaters is concerned, one can see from Figure 5 that the maximum system capacity is obtained when m=2 in the non-interference case and when m=3 in the complete interference case. Thus 2 or 3 repeaters would be a good design; any additional repeaters that may be added because of other considerations (such as area coverage) will result in a reduction in the system capacity.



- 2-Hop Slotted ALOHA Network
- Transmission From Terminals to Station
- Complete Interference System I = m

FIGURE 4

NETWORK THROUGHPUT VS. TERMINAL-REPEATER TRANSMISSION RATE

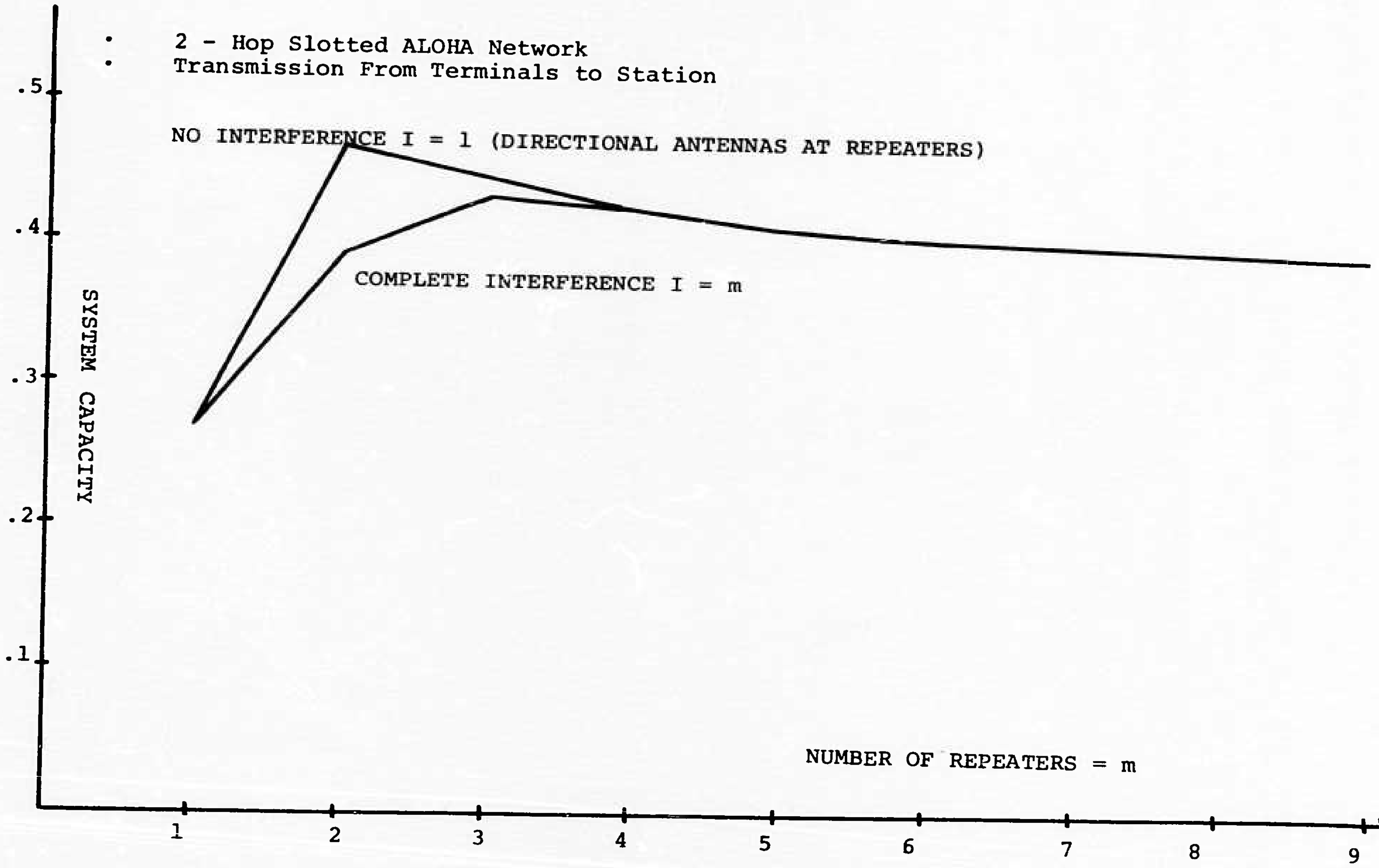


FIGURE 5

NETWORK THROUGHPUT VS. NO. OF REPEATERS: DIRECTIONAL AND NON-DIRECTIONAL ANTENNAS

IV. TRANSMISSION FROM STATION TO TERMINALS

In this section, we consider the second channel which is used for transmission from the station to terminals via repeaters. In the terrestrial system, it is assumed that the effective transmission range of the station is such that it interferes with the transmission from repeaters to terminals, as shown in Figure 6. However, we assume that terminals cannot directly receive from the station or from the satellite (otherwise, it becomes a single hop network and the capacity is 1). We use the notation shown in Figure 6, where the first hop is that from the station to repeaters and the second hop is from repeaters to terminals.

We use similar assumptions to the ones made in the previous section. Specifically, we assume that the probabilities of transmission by a repeater into different time slots are independent, that the probability of transmission of two or more repeaters into a randomly chosen slot are mutually independent, and that the probability of transmission by the station and by repeaters into a random slot are mutually independent. Finally, we simplify by assuming that repeaters share equally the load; by which we mean that $S_{2i} = S_2/m$ and $G_{2i} = G_2/m$, for all i. The equations which relate the rate of transmission to the rate of successful transmission on the two hops can now be written:

$$S_2 = \sum_{1}^{m} \frac{G_2}{m} (1 - G_1) (1 - \frac{G_2}{m})^{I - 1} = G_2 (1 - G_1) (1 - \frac{G_2}{m})^{I - 1} \quad (10)$$

$$S_1 = G_1 (1 - \frac{G_2}{m})^{I} \quad (11)$$

I is the interference level as in the previous section.

For consistency with the interference model of the previous section, we remark the following. Eqs. (10) and (11) are for the case in which the same energy-per-bit-to-noise-density is required

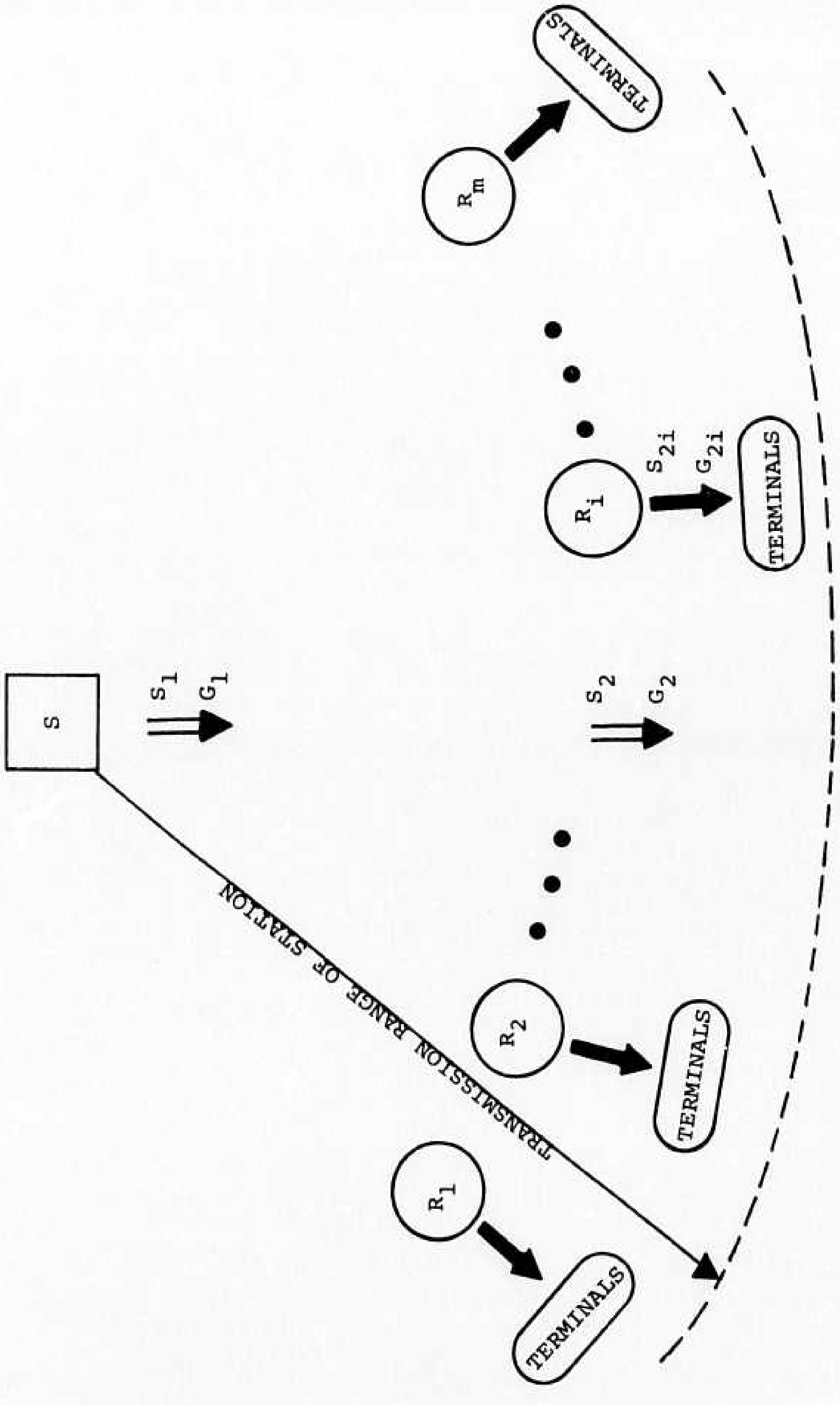


FIGURE 6: TRANSMISSION FROM STATION TO TERMINALS

for detection with equal error rates, by the repeater and by the terminal; and when the repeater used a higher transmitter power. In the more general case, one has to replace I in (10) by I_T which will designate the number of sets of terminals with which a transmission from a repeater may interfere; and replace I in (11) by I_R which will indicate the number of repeaters with which a transmission from a repeater may interfere.

We assume again steady state system operation and use the conservation law at repeaters, i.e., $S_1 = S_2$. This results in

$$G_2 = \frac{G_1}{1 - G_1\frac{m-1}{m}} \tag{12}$$

One can now substitute (12) into (10) and (11) to obtain the throughput as a function of one independent variable. For S_1, one obtains:

$$S_1 = G_1\left(\frac{1 - G_1}{1 - G_1\frac{m-1}{m}}\right)^I \tag{13}$$

To obtain the capacity one can maximize either S_1 or S_2. Equating dS_1/dG_1 to zero, we obtain that there are three stationary points. A minimum at $G_1^1 = 1$ and two maximum values at

$$G_1^{2,3} = \frac{(2m + I - 1) \pm \sqrt{4mI + (I - 1)^2}}{2\ (m - 1)} \tag{14}$$

We now examine the constraints. For the system to be realizable,

$$0 \leq G_1 \leq 1\ ;\ 0 \leq G_2 \leq m\ ;\ 1 \leq I \leq m \tag{15}$$

From Equation (14), one can see that the plus sign results in $G_1 > 1$, since

$$2m + I - 1 + \sqrt{4mI + (I - 1)^2} > 2m + 2(I - 1) > 2(m - 1) \tag{16}$$

Furthermore, from Equation (12), one can see that when $0 \leq G_1 \leq 1$, then $0 \leq G_2 \leq m$. Thus, the only realizable maximum is given by Equation (13) with G_1 as in Equation (14) when taking the minus sign of the square root.

Figure 7 shows the system throughput as a function of G_2 for $m = 6$ and I as a parameter. One can see that there is a high degradation in system performance when the interference level I increases. Figure 8 shows the capacity of the system as a function of the number of repeaters m for the non-interference $(I = 1)$ and the complete interference $(I = m)$ cases. When $1 < I < m$, the curve will be between the two shown. It can be seen that there is a large difference in system capacity between the non-interference and the complete interference systems, and that this difference increases with the number of repeaters m. For $I = 1$, the capacity of the system is $(m(\sqrt{m} - 1)^2)/(m - 1)^2$ which tends to 1 when m tends to infinity.

In the terrestrial system, the interference level depends on the transmission power of the repeaters when transmitting to terminals. Thus, it would be advantageous to use as low transmission power as possible sufficient to reach the terminals, or possibly an adaptive power mechanism.

Directional Antennas and Multiple Transmitters at the Station

This section is addressed only to a terrestrial system. When the station uses a directional antenna, then its transmission to repeater R_i does not interfere with R_j, $j \neq i$. Consequently, the average rate of transmission per slot to a single repeater is G_1/m (assuming equal share of load). The only change that would result in Equations (10) and (11) is the replacement of G_1 in Equation (10) by G_1/m. Doing so and equating S_1 and S_2 results in $G_1 = G_2$ and:

$$S_1 = G_1\left(1 - \frac{G_1}{m}\right)^I ; \quad 0 \leq G_1 \leq 1 ; \; 1 \leq I \leq m \qquad (17)$$

$$S_2 = G_2\left(1 - \frac{G_2}{m}\right)^I ; \quad 0 \leq G_2 \leq m ; \; 1 \leq I \leq m \qquad (18)$$

The capacity of the system is given by

$$S_1^* = \begin{cases} \frac{m}{I+1}\left(1 - \frac{1}{I+1}\right)^I ; \ m \le I + 1 & \text{(19a)} \\ \left(1 - \frac{1}{m}\right)^I ; \ m > I + 1 & \text{(19b)} \end{cases}$$

Figure 9 shows the capacity of the system as a function of m for I = 1 and I = m. The same curves for an omni-directional antenna are shown as a reference. It can be seen that the capacity of the system with a directional antenna is substantially higher; in particular, when the interference level is low.

We now address the question of multiple transmitters and antennas at the station. Consider the capacity of the system with a directional antenna at the station, Equation (19). The maximum given by (19a) is a stationary point whereas that given by (19b) is a boundary point at $G_1 = 1$. Also note that

$$\frac{m}{I+1}\left(1 - \frac{1}{I+1}\right)^I \ge \left(1 - \frac{1}{m}\right)^I, \text{ for } m \le I + 1 \qquad (20)$$

Thus, if one increases the domain of G_1, it would result in an increase in system capacity, which will then be given by Equation (19a). This is exactly what happens when adding additional transmitters and antennas to the station. It enables the station to transmit more than one packet into the same "slot in time" and direct the packets to different repeaters. In practice, if the station has several transmitters and directional antennas which enables it to transmit simultaneously in different directions, then one can devise an algorithm at the station, which will properly select the directions to which packets are simultaneously transmitted, so as to further reduce the interference level I (manage its transmissions).

Figure 10 shows the system throughput as a function of the rate of transmission from station to repeaters (or from repeaters to terminals, note $G_1 = G_2$). One can see that for $I = 1$ and $I = 3$, the maximum system capacity cannot be obtained with a single transmitter and the value obtained is at the boundary point at $G_1 = 1$ and given by (19b). Notice that when the number of repeaters is large and the interference level is low, then there is a large difference between the maximum capacity and the constrained maximum capacity.

We now determine the minimum number of transmitters and directional antennas needed at the station. If the interference level I is constant then the unconstrained capacity of Equation (19a) is increasing with the number of repeaters m. Moreover, the capacity increases also in the case that I is a linear function of m. For let $I = km$, $1/m \leq k \leq 1$, then the unconstrained capacity is given by

$$S^* = \frac{m}{km + 1} \left(1 - \frac{1}{km + 1}\right)^{km} \tag{21}$$

and

$$\frac{dS^*}{dm} = \frac{km}{(km + 1)^2} \left(1 - \frac{1}{km + 1}\right)^{km - 1} \geq 0; \text{ for } k \geq 0 \tag{22}$$

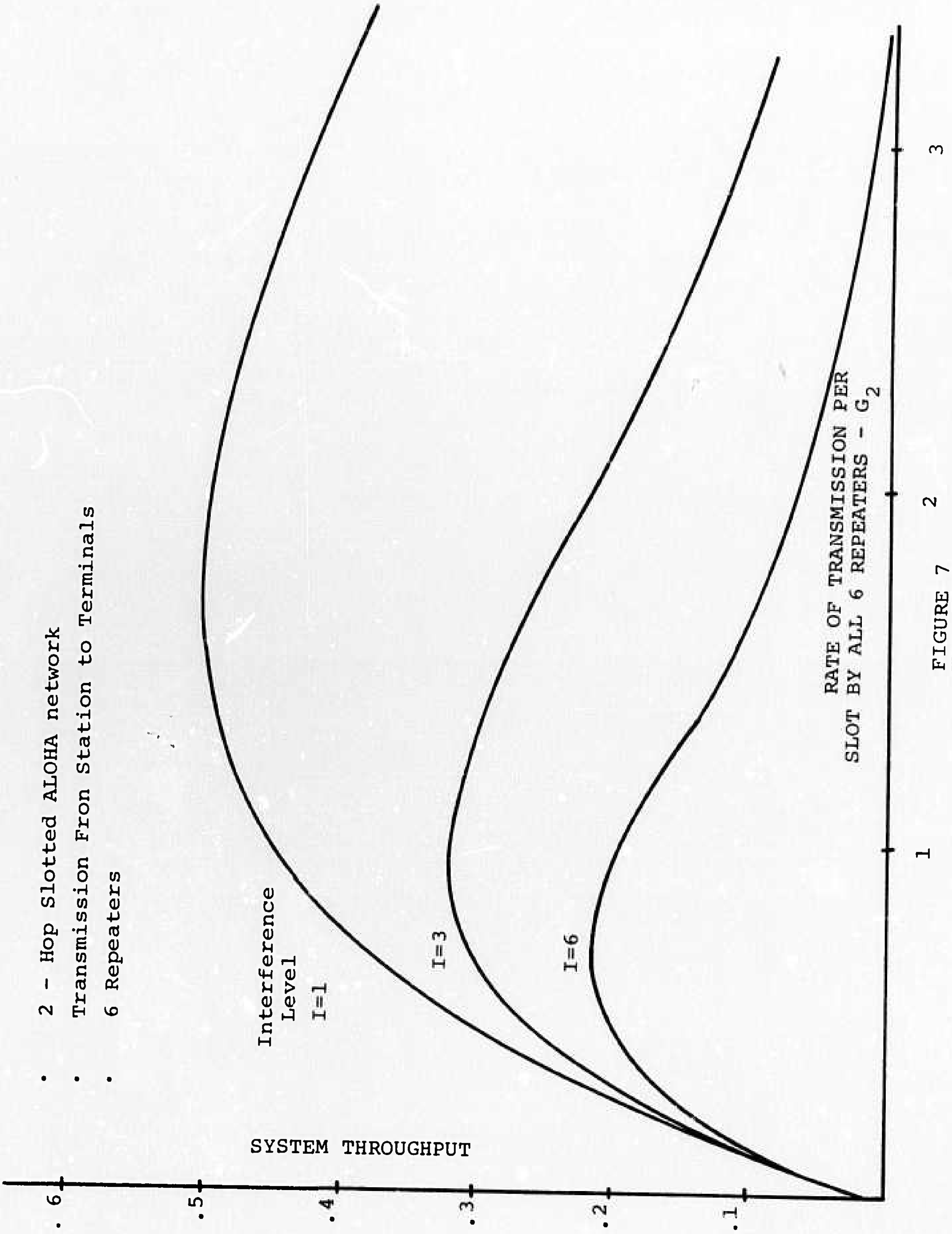
Since the capacity is increasing as a function of m, one can obtain a capacity greater than 1 (see for example Figure 10, I = 1). The minimum number of transmitters that can realize the unconstrained capacity is given by the rate of transmission from the station to repeaters at which the maximum utilization is obtained. That is:

$$\text{Minimum Number of Transmitters} = \left\lceil \frac{m}{km + 1} \right\rceil = \left\lceil \frac{m}{I + 1} \right\rceil \qquad (23)$$

where $\lceil x \rceil$ is the smallest integer greater than x. Equation (23) also implies that multiple transmitters will not result in an increase in system capacity when the interference level is high, i.e., $m \leq km + 1$. It is easy to verify that when the constraint on G_1 is satisfied, the constraint on G_2 is also satisfied.

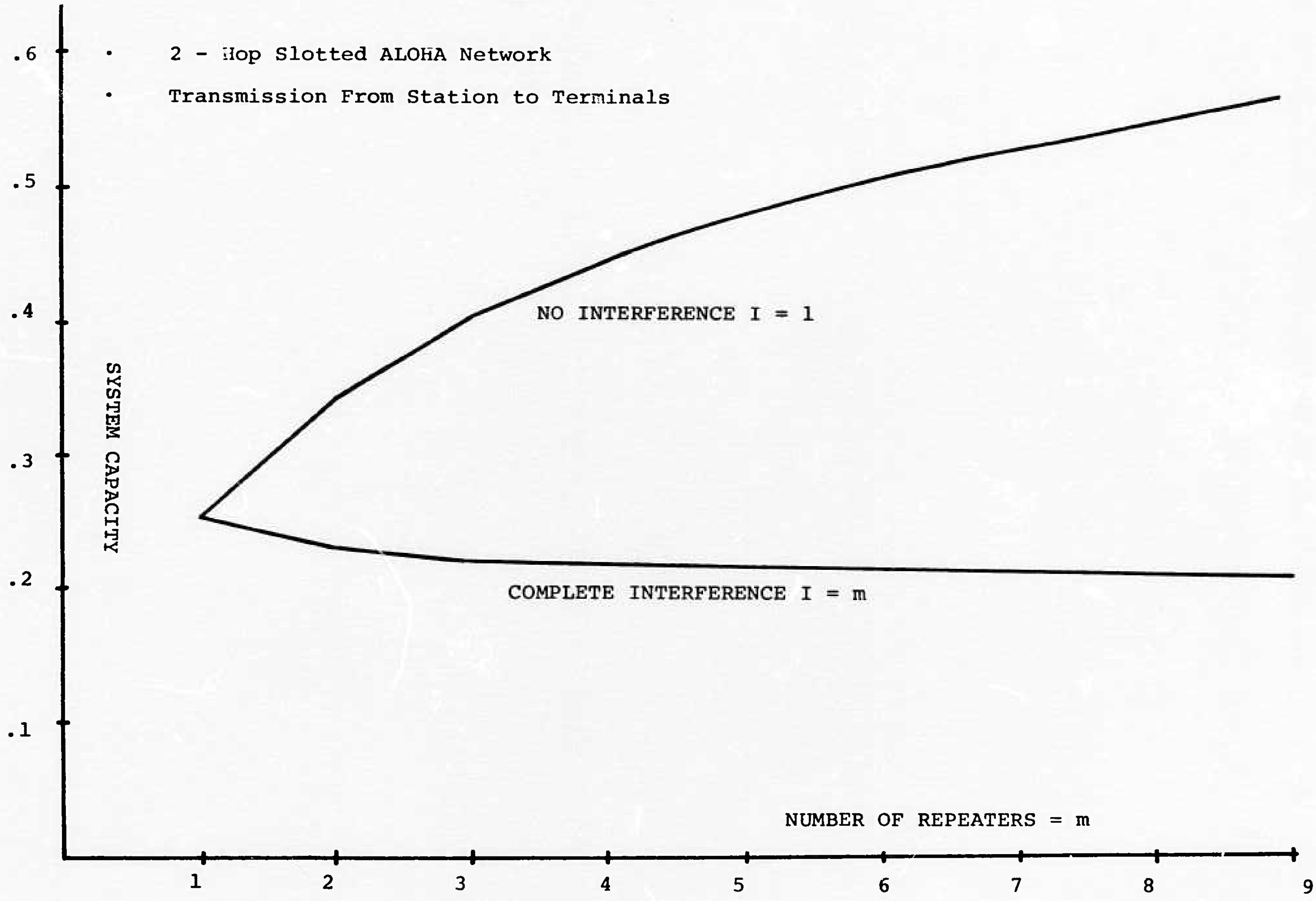
By associating a cost value with a repeater and with each additional transmitter at the station, one can formulate a design optimization problem in which the system cost is traded against the increase in capacity which it results.

The results of this section demonstrate that a single slotted ALOHA channel can be used (and reused) spatially to obtain channel utilization higher than 100%.



- 2 - Hop Slotted ALOHA network
- Transmission Fron Station to Terminals
- 6 Repeaters
Interference Level I=1
I=3
I=6
SYSTEM THROUGHPUT
.6
.5
.4
.3
.2
.1
1
2
3
RATE OF TRANSMISSION PER SLOT BY ALL 6 REPEATERS - G_2

FIGURE 7



.6
.5
.4
.3
.2
.1
• 2 - Hop Slotted ALOHA Network
• Transmission From Station to Terminals
SYSTEM CAPACITY
NO INTERFERENCE I = 1
COMPLETE INTERFERENCE I = m
NUMBER OF REPEATERS = m
1
2
3
4
5
6
7
8
9

FIGURE 8

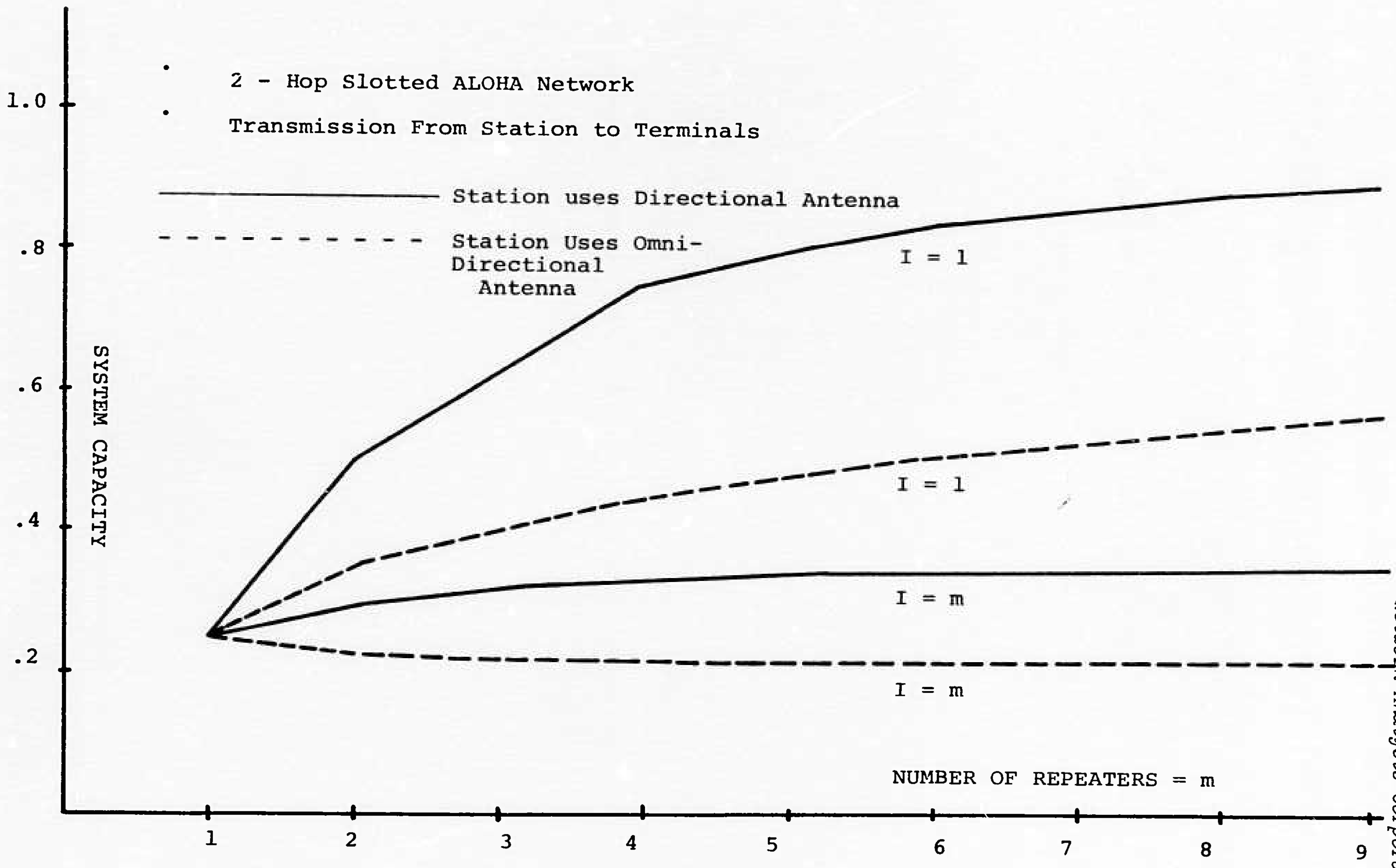

2 - Hop Slotted ALOHA Network
Transmission From Station to Terminals
Station uses Directional Antenna
Station Uses Omni-Directional Antenna
SYSTEM CAPACITY
1.0
.8
.6
.4
.2
I = 1
I = 1
I = m
I = m
NUMBER OF REPEATERS = m
1
2
3
4
5
6
7
8
9

FIGURE 9

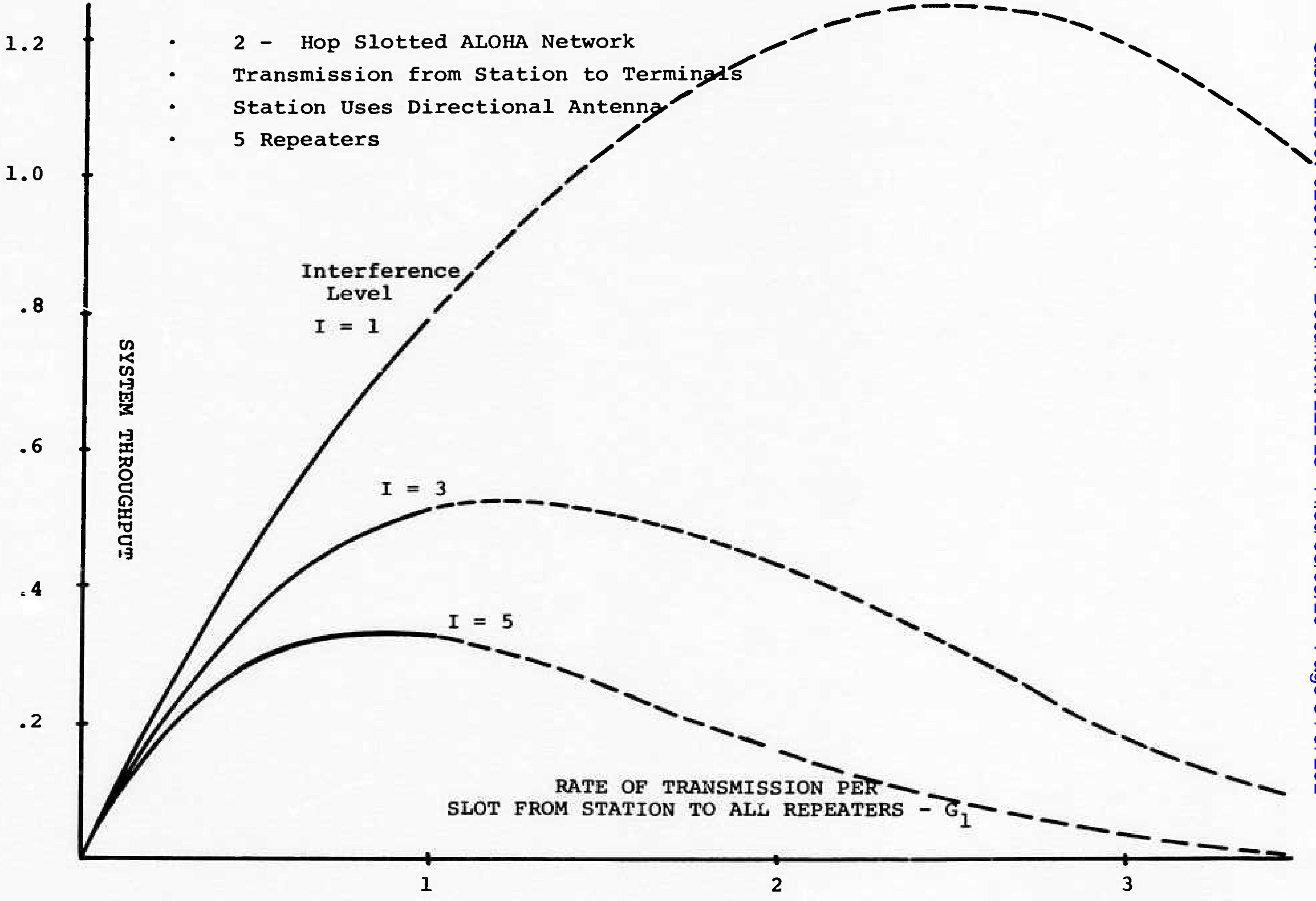

• 2 - Hop Slotted ALOHA Network
• Transmission from Station to Terminals
• Station Uses Directional Antenna
• 5 Repeaters
1.2
1.0
.8
.6
.4
.2
SYSTEM THROUGHPUT
Interference
Level
I = 1
I = 3
I = 5
1
2
3
RATE OF TRANSMISSION PER
SLOT FROM STATION TO ALL REPEATERS - G_1

FIGURE 10

V. CONCLUSIONS

Our conslusions from the analysis are outlined.

A. Transmission to Station

1. When the number of repeaters is small (the exact number depends on the interference level), then the critical hop in the network (the capacity bottleneck) is from terminals to repeaters.

2. When the number of repeaters is large, the critical hop is from repeaters to station.

3. The 2-Hop network design which maximizes the system capacity has 2 repeaters when the interference is minimum, and 3 repeaters when the interference is maximum. Additional repeaters reduce system capacity.

4. The capacity of a 2-Hop network is higher than that of a 1-Hop network when the number of repeaters is 2 or more, and is lower than the capacity of a 1-Hop network when there is one repeater.

5. Directional antennas at repeaters increase system capacity when the critical hop is from terminals to repeaters. The increase is significant only in the case when there are 2 repeaters, which do otherwise interfere with each other. In other cases, the increase in capacity is either insignificant or does not exist.

B. Transmission from Station

1. The interference of the station with the transmission of repeaters to terminals reduces significantly the system capacity. Thus, if possible, it is important to enable terminals to receive such transmissions.

2. The system capacity reduces substantially when the interference level between repeaters is increased. Note that this is not the case when transmitting to the station; compare Figures 5 and 8. Consequently, in a terrestrial system, it is important to reduce the interference factor by a mechanism such as adaptive power.

3. A directional antenna at the station in a terrestrial system increases significantly, the system capacity when the interference level between repeaters is low to moderate. This is not the case when the interference level is high, since the throughput on the hop from repeaters to terminals is limited due to this interference.

4. When the station has directional antennas, then multiple transmitters and antennas may further increase, significantly, system capacity. Note that in this case, one can obtain a capacity greater than 1.

5. An equation for the number of transmitters needed at the station is given. This number increases when the interference level decreases. When the interference level is high, then none of the devices in conslusions B.3, B.4, B.5, are desirable, since the capacity is limited by the throughput from repeaters to terminals.

REMARKS:

Conclusions A.5 and B.3 which relate to directional antennas, imply that directional antennas are generally not useful when devices which use them direct transmissions to a single location in space; because the interference at this location is not avoided. On the other hand, directional antennas are very useful when oriented to different azimuths, because one can take advantage of the spatial districution of receivers.

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PACKET RADIO SYSTEM CONSIDERATIONS - CHANNEL CONFIGURATION

I. INTRODUCTION

Consider a communication channel which is shared by two independent sources of traffic in a broadcast mode. Source 1 is generated by an infinite number of terminals, each with an infinitesimal traffic rate, and which collectively form a finite Poisson source. Source 2 is generated by a finite number of terminals each with a finite traffic rate. The terminals transmit fixed size packets and access the channel using the so called "slotted ALOHA" random access scheme. A terminal can transmit to any other terminal in the system.

This model can describe the ALOHA system at the University of Hawaii [1] or the Packet Radio System [6]. In the Packet Radio System there is a large number of terminals which communicate with a small number of stations. The terminals can be modelled by the terminals of Source 1 and the stations by the terminals of Source 2. The model is suitable for a Packet Radio System in an urban area where a terminal can directly transmit to a station and where any transmission from a terminal or station interferes with all other devices.

Roberts [5] has shown that if all the traffic is contributed by Source 1, then the maximum throughput which can be obtained is 1/e of the channel capacity. Abramson [2] and Kleinrock and Lam [3] have shown that the maximum throughput can be increased when the traffic is composed of contributions from both Source 1 and Source 2.

We approach the problem from a synthesis viewpoint. That is, given Source 1 and Source 2, the question is whether one should split the channel so that one part is used by Source 1 and the other part by Source 2; or alternatively, should one use the total capacity in common. The criteria for decision are maximum throughput and average delay. We consider a channel split in which the slots are partitioned between transmissions from Source 1 and Source 2, however, all terminals receive on all time slots. It is shown that the choice of channel configuration depends on the number of terminals in Source 2, n, and on the ratio of packet rate of Source 2 to Source 1, designated by α. Further, given n, there is an interval of α for which a higher maximum throughput can be obtained by splitting the channel.

The problem considered in this chapter was addressed in [7] for n=1 but for several random access schemes; specifically, for the nonslotted and slotted ALOHA and for the carrier sence [8] random access schemes. The qualitative conclusions of [7] are the same as in this chapter.

II. PRELIMINARY ANALYSIS

A. The Slotted ALOHA Channel

In the slotted ALOHA random access mode, a channel is partitioned into segments of time (slots) equal to a packet transmission time. Terminals transmit their packets into random slots in time. That is, there is no coordination among the terminals as far as the choice of the slot is concerned; however, there is a universal clock which enables each terminal to start the transmission of its packet at the beginning of a slot. If two or more packets are transmitted in the same slot, it is assumed that none of the packets are correctly received and each of the terminals will retransmit its packet at some randomly chosen future slot.

One can see that the number of packets transmitted (the channel traffic) is larger than the number of packets offered to the system due to the retransmissions of packets which collide. Let S denote the rate of packet originations per slot offered to the channel, and G the rate per slot of packets plus retransmissions. Assume that the two origination processes are Poisson. Further assume that the probability that a packet is blocked, given the packet is new, equals the probability that a packet is blocked given it is a retransmission [3].

B. The Single Source Case

If all the traffic is contributed from Source 1, Roberts [5] has shown that the relation between S_1 and G_1 is given by:

$$S_1 = G_1 e^{-G_1} \tag{1}$$

Abramson [2] considered the case where all the traffic is contributed from Source 2 of n identical terminals and has shown that

$$S_2 = G_2 \left(1-\frac{G_2}{n}\right)^{n-1} \tag{2}$$

One can see that Eq. (2) takes the form of Eq. (1) when n → ∞ . The maximum values of S_1 and S_2 are given by:

$$S_1^* = \frac{1}{e} \; ; \; S_2^* = \left(1-\frac{1}{n}\right)^{n-1} \tag{3}$$

The operation of the slotted ALOHA channel may become unstable [4]. In this paper we consider the steady state case, in which the offered rate is also the throughput per slot or the utilization of the channel.

C. Mixed Sources on a Common Channel

The performance of a channel which is shared by terminals from Sources 1 and 2 has been analyzed by Abramson [2], and Kleinrock and Lam [3]. The offered packet rate to the channnel is $S_1 + S_2$, and the channel traffic $G_1 + G_2$. The following equations hold:

$$S_1 = G_1 \left(1-\frac{G_2}{n}\right)^n e^{-G_1} \tag{4}$$

$$S_2 = G_2 \left(1-\frac{G_2}{n}\right)^{n-1} e^{-G_1} \tag{5}$$

III. MAXIMUM UTILIZATION OF THE SPLIT AND COMMON CHANNELS

Given Source 1 and Source 2, the question is whether one should split the channel so that one part is used by Source 1 and the other part by Source 2; or alternatively, should one use the total capacity in common. In this section we compare the channel configurations in terms of maximum utilization. To obtain an absolute comparison, we introduce the parameter of the ratio of packet rates of Source 2 to Source 1:

$$\alpha = \frac{S_2}{S_1} \qquad (6)$$

We shall use the subscripts c and s to denote a common channel and a split channel, respectively; and the superscript * to denote the optimum or maximum values. The total given capacity will be assumed as one unit and (C_1, C_2), $C_1 + C_2 = 1$, will denote a channel split where the fraction C_1 is assigned to Source 1 and the fraction C_2 to Source 2. Given an arbitrary split (C_1, C_2), the maximum utilization of the configuration is given by:

$$S_s^* = \begin{cases} C_1(1+\alpha)S_1^* & ;\alpha \le \dfrac{S_2^* C_2}{S_1^* C_1} \qquad (7a) \\[2ex] C_2(1+\frac{1}{\alpha})S_2^* & ;\alpha > \dfrac{S_2^* C_2}{S_1^* C_1} \qquad (7b) \end{cases}$$

Corresponding to C_1 and C_2 being saturated respectively.

If α is known one can split the channel optimally to obtain the highest maximum utilization of the total capacity. It can be shown that the optimum split (C_1^*, C_2^*) satisfies the following equation:

$$\frac{C_2^*}{C_1^*} = \frac{\alpha S_1^*}{S_2^*} \qquad (8)$$

Corresponding to both channels saturating simultaneously.

If the channel is optimally split, then the maximum utilization of the total capacity is given by:

$$S_s^* = C_1^* S_1^* + C_2^* S_2^* = \frac{(1+\alpha)(1-\frac{1}{n})^{n-1}}{e(1-\frac{1}{n})^{n-1} + \alpha} \qquad (9)$$

Note from Eq. (9) that when $\alpha \to 0$, $S_s^* \to \frac{1}{e}$, and when $\alpha \to \infty$, $S_s^* \to (1-\frac{1}{n})^{n-1}$.

The total utilization of the common channel configuration is given by the summation of Eqs. (4) and (5).

$$S_c = e^{-G_1} (1-\frac{G_2}{n})^{n-1} [G_2 + G_1 (1-\frac{G_2}{n})] \qquad (10)$$

To obtain the maximum of S_c, one has to maximize Eq. (10) subject to the constraint $S_2/S_1 = \alpha$. Alternatively, we can use the condition of the channel traffic at the maximum utilization obtained by Abramson [2]. Doing so, we obtain:

$$S_c^* = e^{-G_1^*} (1-\frac{G_2^*}{n})^{n-1} (1-\frac{G_1^* G_2^*}{n}) \qquad (11)$$

where,

$$G_2^* = \frac{n + \alpha(n+1) - \sqrt{[n+\alpha(n+1)]^2 - 4\alpha^2 n}}{2\alpha} \qquad (12)$$

and,

$$G_1^* = 1 - G_2^* \qquad (13)$$

Figure 1 shows the comparison between the maximum utilizations of the two configurations S_c^* and S_s^*. It is shown as a function of α with n as a parameter. The values of n used are 1, 3, and ∞, and the split $C_1 = C_2 = \frac{1}{2}$.

From Figure 1, one can see that for a given n, there is an interval of α such that within this interval $S_s^* \geq S_c^*$, and outside of the interval $S_s^* < S_c^*$. Furthermore, the interval discussed decreases when n increases and constitutes a single point when $n = \infty$.

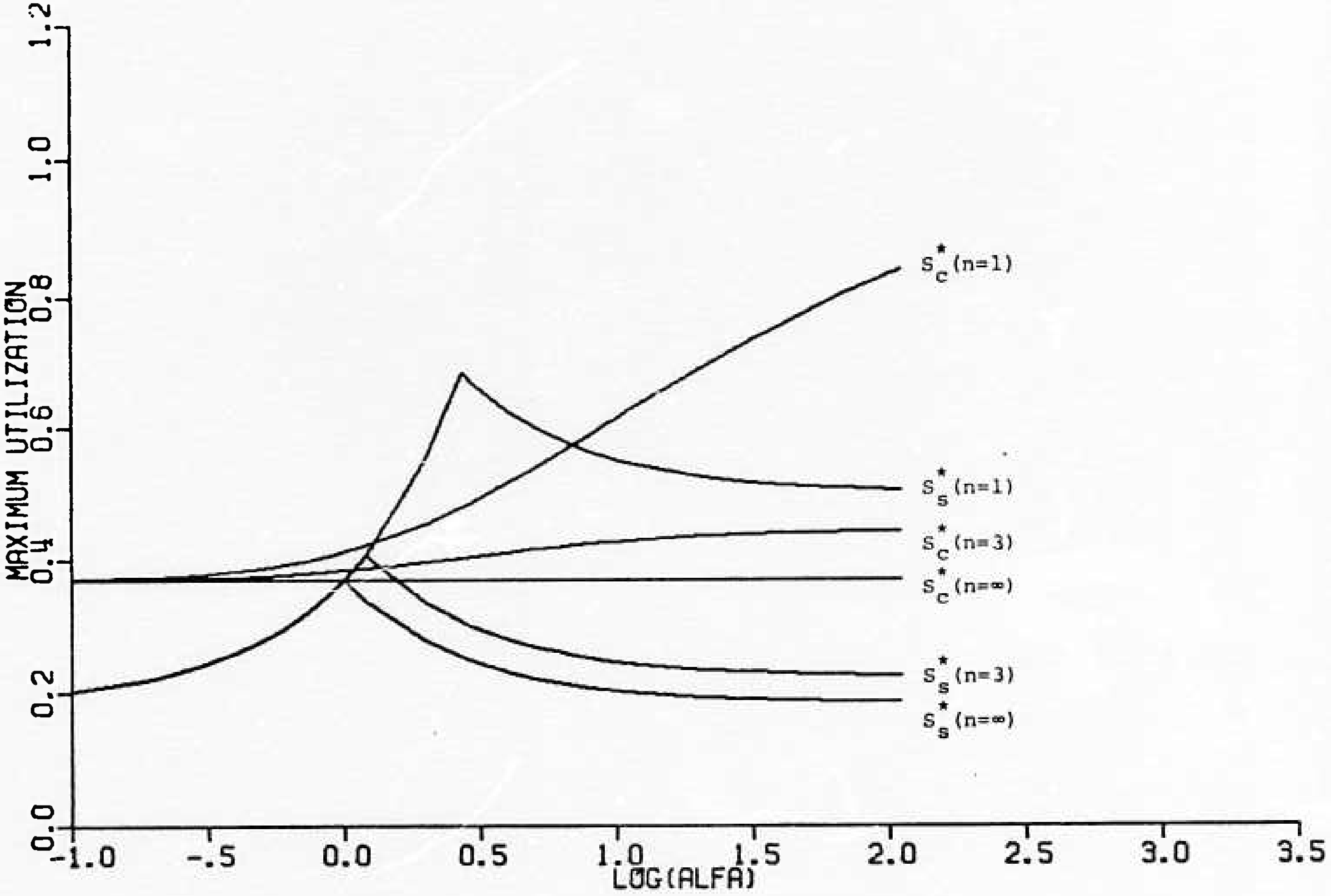

FIGURE 1

MAXIMUM UTILIZATION VS. SPLIT AND COMBINED CHANNEL PARAMETERS

IV. DELAY CONSIDERATIONS

It is clear that when α is within the interval discussed in the previous section, the split configuration is better than the common configuration in terms of maximum throughput and average delay, since an infinite delay is obtained in the common configuration before it reaches the maximum utilization of the split configuration. The average delay of a packet from Source 1 will usually be different from that of a packet from Source 2. In particular, in the split configuration when the maximum utilization is obtained, one channel is saturated (infinite delay) and the other is not; except for the optimum splitting α for which both channels saturate simultaneously.

The average delay in this system is composed of the delay when the first transmission is successful plus the average number of retransmissions times the average delay per retransmission. A terminal in Source 2 also encounters queueing delay.

We show curves of delay vs. throughput for several parameters α. We consider a terrestrial system in which propagation delay is ignored, and use all the assumptions given in Section 1. The average delay equations in units of slot times used are the following:

$$D_1 = 1.5 + \left(\frac{G_1 - S_1}{S_1}\right) \bar{k} \qquad (14)$$

$$D_2 = 1 + .5 \left(1 - \frac{G_2}{n}\right) + \frac{G_2/n}{2(1 - G_2/n)} + \left(\frac{G_2 - S_2}{S_2}\right) \bar{k} \qquad (15)$$

where $\bar{k}$ is the average waiting per retransmission. The value .5 is added to represent that when a packet is ready for transmission, the terminal will wait one half slot, on the average, until the beginning of a slot. The third term in Eq. (15) represents the queueing delay at terminals of Source 2. When writing the queuing delay we assume that packets arrive according to a Poisson distribution and require constant service time equal to the packet transmission time (an M/D/1 queueing system). Note that when n then Eq. (15) takes the same form as Eq. (14).

The delay equations hold for the split as well as the common channel configurations. However, if one assumes the same packet length in each case and the slot time on the common channel is one unit, then the slot time on the split channel will be $1/C_1$ and $1/C_2$ for the respective channels. This has been taken into account when showing the delay curves.

Figures 2, 3, and 4, show the delay as a function of throughput for $\alpha = .5$, $\alpha = 2.5$, $\alpha = 10.$, and for $n = 1, 3,$ and ∞. Other parameters are $C_1 = C_2$ and $\frac{1}{2}$ and $\bar{k} = 4$. From Figure 1, one can see that $\alpha = .5$ and $\alpha = 10.$ corresponds to the case where $S_c^* > S_s^*$; on the other hand $\alpha = 2.5$ corresponds to the case where $S_c^* < S_s^*$. The throughput shown is the sum of both sources and the delay is an average of D_1 and D_2 weighted by the throughputs of the sources.

One can see that when $S_c^* > S_s^*$ the common channel configuration results in lower average delay for all values of throughput (for the same value n). On the other hand, if $S_s^* > S_c^*$ (Figure 3), there are operating points, for example when the throughput is .55, where the split channel configuration is better both in throughput and delay. However, even in this case, when the system operates at low throughputs, the common channel configuration results in lower values of delay. This is due to the differences in the slot times.

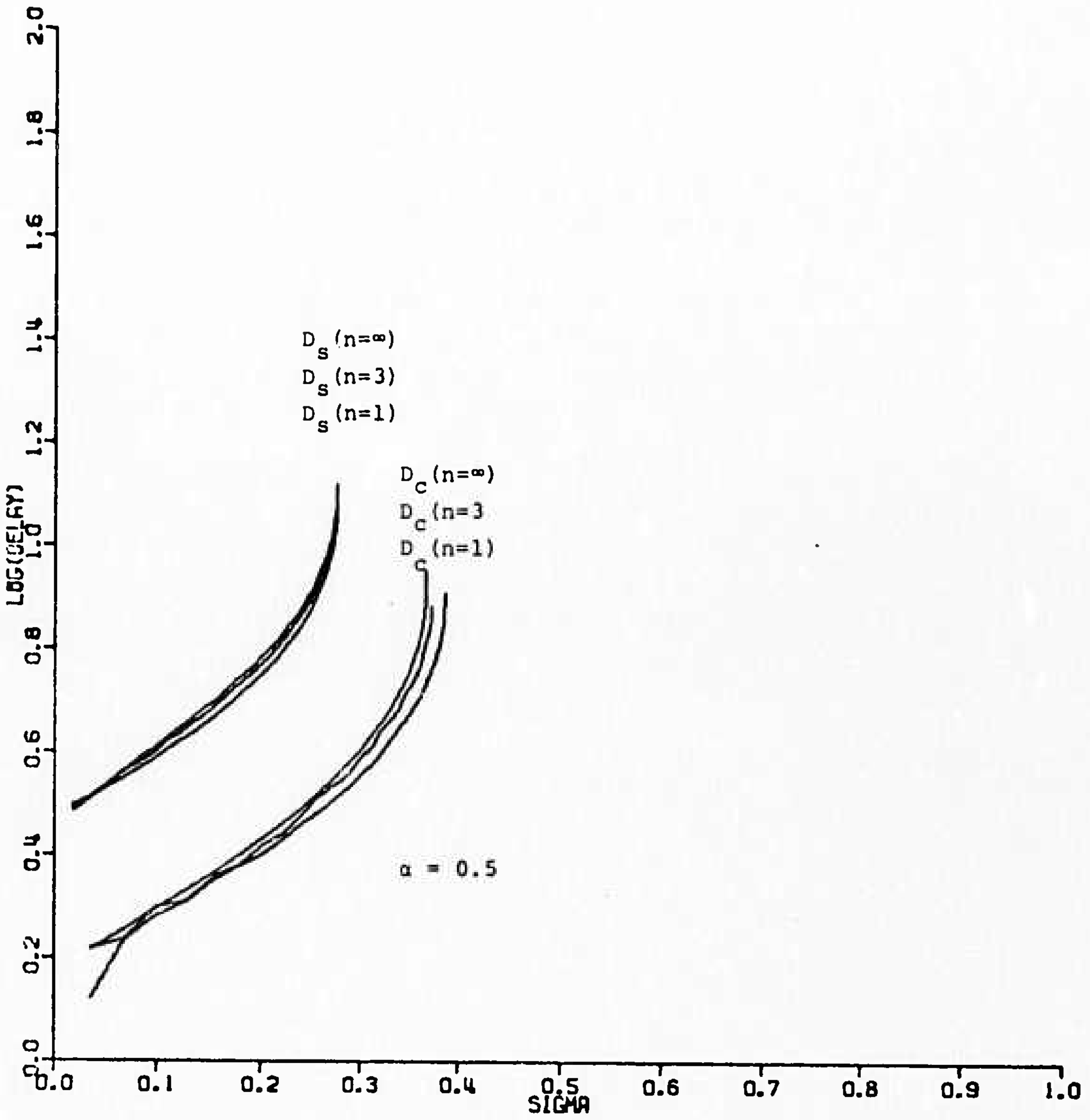

FIGURE 2

DELAY VS. THROUGHPUT, α = .5

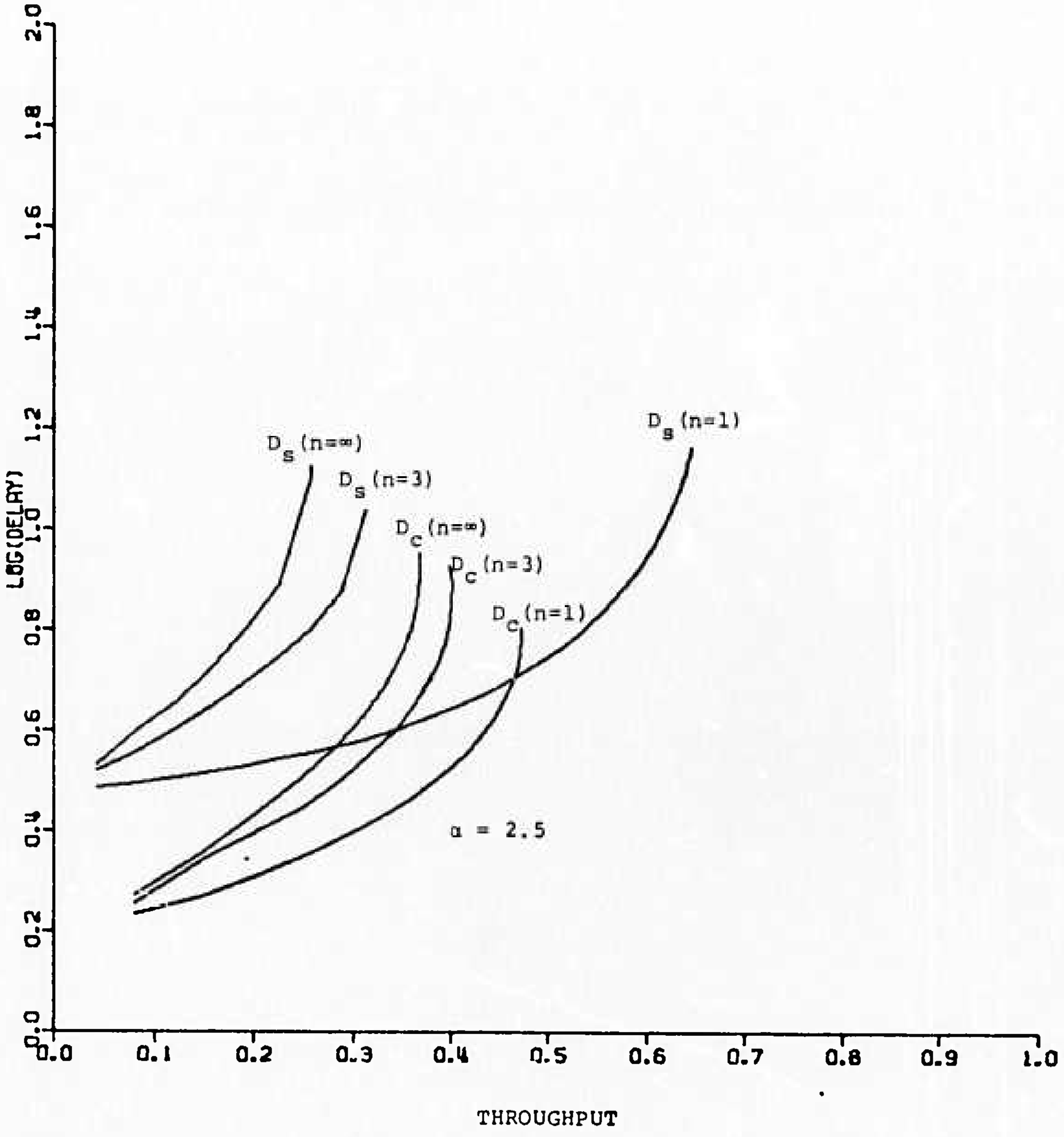

FIGURE 3

DEALY VS. THROUGHPUT, $\alpha = 2.5$

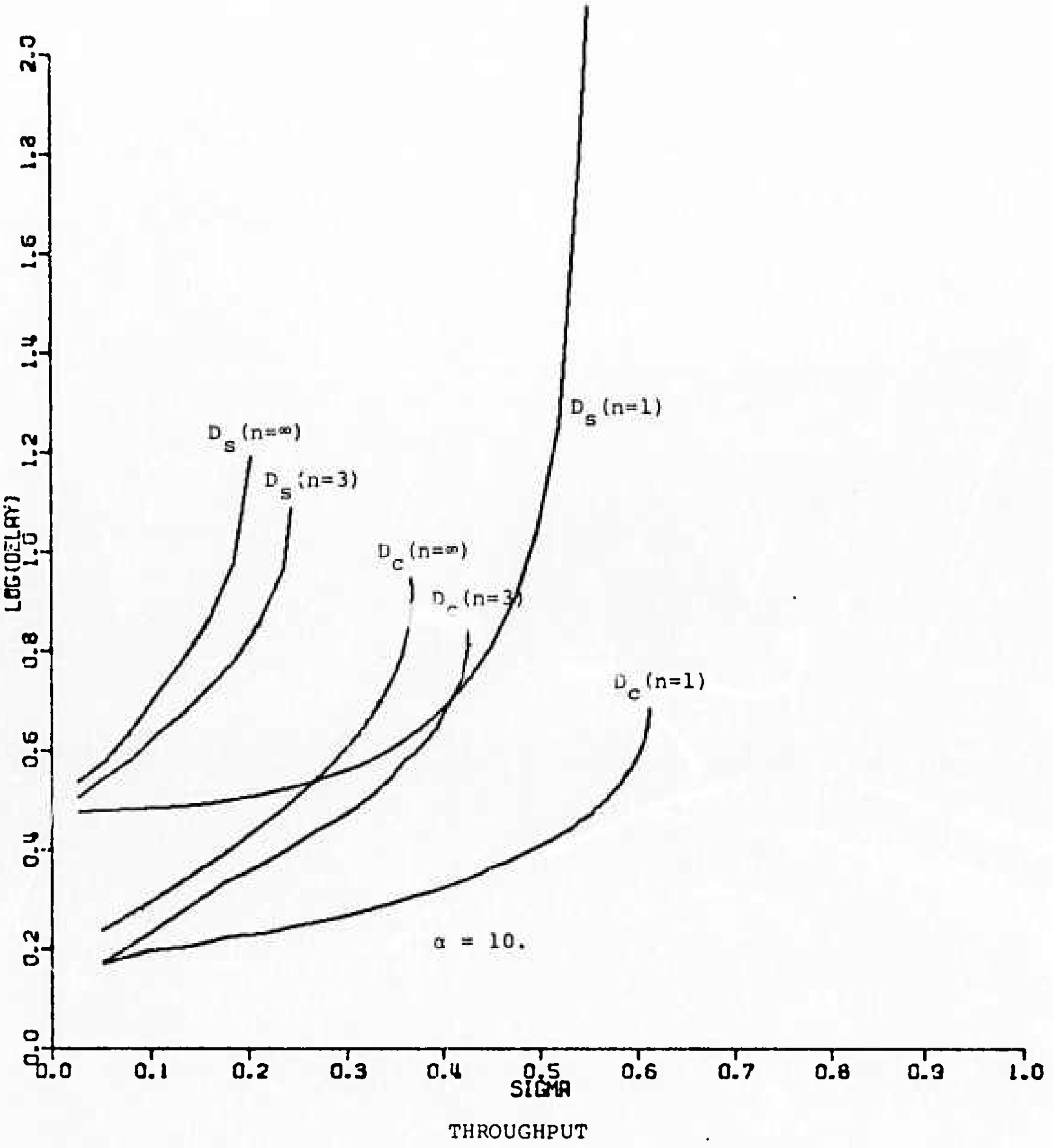

FIGURE 4

DELAY VS. THROUGHPUT, α = 2.5

V. CONCLUSIONS

We relate the results obtained to the Packet Radio System. If one assumes that a packet which originates from a terminal will always be directed towards a station then α has the practical meaning of the average number of response packets from station to terminal for every packet which is successfully transmitted from a terminal to a station.

It is demonstrated that if this ratio α is known, then one can split the channel into two to obtain a higher maximum utilization of the total channel capacity. It is also shown how to split the channel. However, if the channel is split and the system operates at low values of throughput, then the average delay is higher than would result from operating in a common channel configuration. Another conclusion is that if α is not known, or if it varies, then the common channel configuration will be better in terms of throughput and delay.

Finally, we note that the system analyzed models the Packet Radio System when there is no coordination among the stations in choosing the slot, so that packets from two stations can collide on the same slot. In practice, however, one may consider having signalling channels among the stations to obtain the above coordination dynamically. In this case, the set of stations has to be considered as a single source; that is, $n = 1$.

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AREA COVERAGE BY LINE-OF-SIGHT RADIO

I. PROBLEM FORMULATION

We are concerned with line-of-sight coverage of an area where mobile terminals or fixed terminals are transmitting by radio from unspecified locations. The problem is to locate repeaters so that any such terminal will be in line-of-sight of repeaters and that there be reliable connections between every pair of terminals (and repeaters). More precisely we wish to minimize the installation cost and maintenance cost of the repeaters subject to a constraint on the reliability of service.

In general, determining if line-of-sight micro-wave transmission between two points is possible, involves taking into account many factors including wave-length (Fresnel zones), weather conditions (effective earth radius), antenna design, height, topography, etc.. Nevertheless, we shall assume that there are known functions $\ell(r,t)$ and $L(r_1,r_2)$ that are 1 if a repeater at location r can communicate with a terminal at location t and if a repeater at location r_1 can communicate with a repeater at location r_2 respectively and are 0 otherwise.

From purely topographical considerations it is obvious that the "flat terrain" problem and the "hilly terrain" problem should be handled seperately. For "flat terrain" the problem is homogeneous, i.e. installation costs, maintenance costs can be assumed equal at all locations and the transmission properties are identical at all points. The "flat terrain" problem is discussed in Section 3; a model is suggested for which we compute an optimal solution.

The primary concern in the ensuing paragraphs is with the "hilly terrain" problem. By that one should understand hilly topographies as well as flat topographies that cannot be viewed as homogeneous from a cost or transmission viewpoint.

In "hilly terrain" it is impractical to consider all possible locations of repeaters and terminals, which theoretically are infinite in number. We shall limit ourselves to a finite set R of possible repeaters locations and a finite set T of possible terminal locations. How the set R and T are chosen will be of great computational importance and will probably be chosen adaptively. But for the time being, we assume R and T known and fixed.

The principal and immediate interest is in an appropraite mathematical model of the situation and some indications on how to solve the problem. The first problem is the proper choice of reliability measure or grade of service. We assume that the radio network is for local distribution-collection of terminal traffic with rates small compared to the channel capacity so that throughput capacity is not a constraint. That is, if any path through the network exists for a given pair of terminals we assume there is sufficient capacity for traffic between them. Possible measure of network reliability that have proved useful in the analysis of communication networks [13] are the probability that all terminal pairs can communicate and the average fraction of terminal pairs which can communicate. However, for network <u>synthesis</u> as distinguished from <u>analysis</u> these approaches appear too difficult both from computational and data collection points of view. This suggests the "deterministic" requirement that there exist k node disjoint paths between every terminal pair. This guarantees that at least k repeaters or line of sight links must fail before any terminal pair is disconnected. Let the cost of a repeater at location $r \in R$ be $c(r)$ and $c(R^{\circ}) = \Sigma \{c(r) | r \in R^{\circ}\}$ where $R^{\circ} \subset R$. Then we can formulate:

<u>Problem I</u>: Find $R^{*} \subseteq R$ minimizing $c(R^{*})$ subject to the constraint that for all t T and r R there exist k node disjoint

paths from t to r in the network N (t;R*) where

$$N(t;R^*) = [R^* \cup \{t\}, \{(r_1,r_2) | L(r_1,r_2)=1\} \cup \{(r,t) | \ell(r,t)=1\}]$$

One might demand only that there be k node disjoint paths between every pair of terminals instead of between each terminal-repeater pair, but we are assumming that communication always takes place through a "station" which could be any of the repeaters. The analysis of the terminal to terminal model is similar in any case.

The following two propositions motivate a new <u>Problem II</u>,

<u>Proposition 1</u>: $|\{r | \ell(r,t) = 1\} \cap R^*| \geq k$.

<u>Proposition 2</u>: For all $r \in R^*$, $|\{r_1 | L(r,r_1)=1\}| \geq k-1$.

If for each $r \in R$ there exists $t \in T$ such that $l(r,t)=0$ then $|\{r_1 | L(r,r_1)=1\}| \geq k$.

<u>Problem II</u>: Choose $R^* \subseteq R$ to minimize

C(R*) subject to:

1. For all $t \in T$, $|\{r | \ell(r,t)=1\} \cap R^*| \geq k$
   (the k-fold set covering problem).
2. For all $r_1, r_2 \in R^*, r_1 \neq r_2$, there exist k node disjoint paths connecting r_1 to r_2.
   (the minimum cost redundant network problem).

The virtue of II - as compared to I is that II is an amalgam of two well studied network problems, the set covering problem [7] and a problem closely related to the minimum cost redundant network problem [11]. We can then attack the problem using previously developed techniques.

Problem I and II are related by:

<u>Proposition 3</u>: Any R* satisfying the constraints of II also satisfies the constraints of I.

Proof: Suppose R* satisfies the constraints of Problem 2 and violates those of Problem I. Then there exists a terminal t_0 and a repeater r such that there are not k node disjoint paths connecting them in $N(t_0;R^*)$. According to the Menger Graph Theorem [8], there are k-1 repeaters $r_1,\ldots,r_{k-1}$ not including r_0 such that when these repeaters are removed from R* there is no path from t to r_0. But this leads to a contradiction. Removing k-1 repeaters cannot disconnect t_0 from the net by 1 of Problem II. Thus, there is an arc (t_0,r') with $(r',t^{\circ}) = 1$. Similarly, k-1 repeaters cannot disconnect r' from r_0 by 2. Thus, there is a path from t_0 to r.

The converse is not necessarily true; that is, there may be feasible solutions to I which are not feasible to II. Figure 1 depicts a counter-example to a possible converse for k=2.

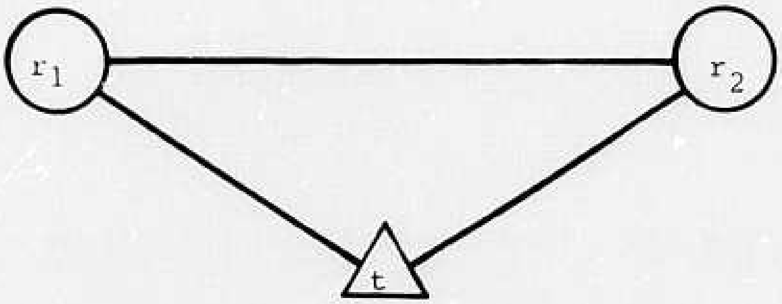

FIGURE 1

However, by Proposition 3, solving II is fail-safe in the sense that the solution obtained will always be feasible for Problem 1.

Feasible solutions to I which are not feasible in II will be quite rare. In order for this to happen, there would have to be two repeaters r_1 and r_2 in R* (where R* is an assumed solution to I) for which there are not k node disjoint paths joining them yet for which all terminals in the vicinity of r_1 can communicate by k node disjoint paths with r_2 and conversely. This does not happen unless there are very few terminals and many repeaters near r_1 and r_2 which from the physical nature of the problem is highly unlikely. On the other hand, as pointed out above, artificial counter-examples to a possible converse of Proposition 3 can easily be constructed.

II. COMPUTATIONAL TECHNIQUES

The following problem will be referred to as a k-cover problem. Let N be a bipartitie graph with edges E and nodes (R,T). The edges E connect nodes of R to nodes of T. Then the following is a k-cover problem:

Find $R^* \subset R$ such that valence of each T-node in N* is at least k and such that the cardinality of R* is minimum (number of elements of R*), where N* is the subgraph obtained by deleting all nodes $R \setminus R^*$ (where $\setminus$ indicates set difference) and all edges connected to these nodes. This is also known as the α-width problem [4], [10].

The 1-Cover Problem is the classical set covering problem. Extensive research has been and is being done on the 1-cover problem, see e.g., [7] and references mentioned therein, see also [1] (A special case of the 1-cover problem is the simple covering problem where each R-node is connected to exactly 2 T-nodes. For this problem algorithms are known that are "efficient" i.e., with known polynomial bounds on the number of operations [7]. So far, practice does not seem to have singled out a "best" algorithm to solve the general 1-cover problem. But in any case, those available seem to be much more efficient than solving the problem as a straight integer programming problem.

a. Every k-cover problem is equivalent to a 1-cover problem. To establish the assertion let us consider the case when k=2. The generalization to arbitrary is important but it is easier to see the proof for the case k=2.

Let us consider a bipartite graph constructed in the following manner: Start with a 2-covering problem. Let L be the number of elements in R. Take L copies of the T-nodes. Connect node r_1 to all copies of node t_j (if r_i, t_j) E of the 2-cover problem, except for the copy of t_j in the i^{th} copy of T. (See Figure 2.)

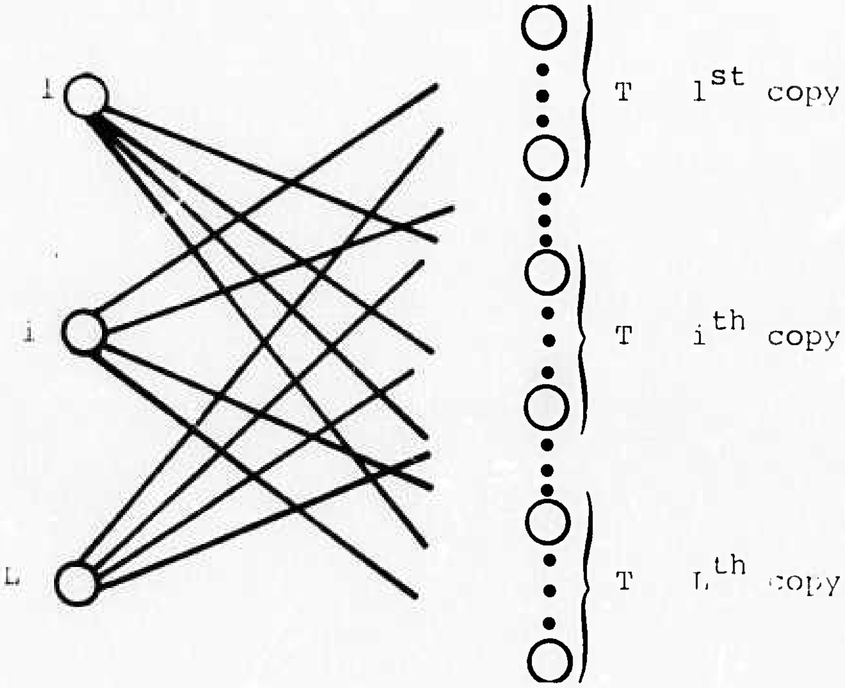

FIGURE 2

We call this problem the 1-cover problem generated by the 2-cover problem.

Proposition: Every solution to the 2-cover problem yields a solution to the generated 1-cover problem and conversely.

In particular, the optimal solution of the 2-cover problem yields an optimal solution to the generated 1-cover problem and conversely.

Proof: The first part (a 2-cover solution yields a solution to the generated 1-cover) is trivial. In the other direction, suppose the n^{th} copy of t_j is "covered" by node r_i the $r_i \varepsilon$ optimal 1-cover of the generated 1-cover problem, but in the i^{th} copy, t_j must be "covered" by another node in R. So every node is "covered" by at least 2-elements of R. The optimality statements follows readily.

For k-covers (k>2) a similar proposition can be found but then the number of copies of T needed (L in these case) go up exponentially. (We suspect that this type of transformation is similar to reducing general integer variables in integer programs to 0-1 variables.)

Important remarks: If the problem is formulated as a k-cover problem, one should realize that by the above results one might suspect that the k-cover problem is much more complicated than the 1-cover problem. This is corroborated by the experience with integer programming algorithms. The only class of problems which can be solved with any form of computational success are those involving only 0 and 1's.

b. The Covering Problem as an Integer Program. Let A be a (T,R) matrix (where the rows correspond to terminals and the columns to repeaters). The size of A is $|T| x |R|$. We have that a_{ij} = 1 or 0 depending on whether terminal i is "visible" or not to repeater j . In terms of the bipartite graph N, the entry a_{ij} = 1 if there is an edge between nodes r_i and t_j, the entry a_{ij} = 0 otherwise. We can then formulate the k-cover problem as follows:

$$\text{Min } \Sigma x_j$$

$$\text{(I.P) such that } \sum_j a_{ij} x_j \geq k \quad i = 1,\ldots,|T|$$

$$\text{with } \quad x_j = 1 \text{ or } 0$$

This is an integer program (sometimes called a program in boolean variables). A number of algorithms to solve integer programs are known. See [7] for a survey. These algorithms offer little hope in solving the location problem arising in the Packet Radio project in that $|R|$ and $|T|$, i.e., the number of variables and the number of constraints respectively, are much too large for existing methods for k-cover problems.

c. <u>Approximate to the k-cover Method by Linear Programming</u>:

The integer programming problem formulated above can be replaced by a linear program

$$\text{Minimize } \sum_j x_j$$

$$\text{(L.P.) such that } \sum_j a_{ij}\, x_j \geq k \quad i = 1,\ldots,|T|.$$

$$0 \leq x_j \leq 1$$

Where the constraint $x_j = 0$ or 1 has been replaced by the constraint $0 \leq x_j \leq 1$. The difference is obvious, the solution to (L.P.) will contain fractional values, but we note that if a variable is "profitable", it will usually be made as large as possible. The constraints $\sum_j a_{ij}\, x_j \geq k$ do <u>not</u> generate upper bounds on the x_j's. Thus one might expect many 0 and 1's in the optimal solution of the L.P. problem.

An upper bound for the optimal solution to the (I.P.) integer program can be obtained by "pushing" all the fractional x_j's appearing in the optimal solution to 1. Obviously, that yields an upper bound on Min Σx_j. The optimal solution of (L.P.) yields a lower bound.

There are many ways to improve the above solution. (without actually going all the way to integer programming). Let us suggest two.

Scheme 1. Let x^{opt} be the optimal solution of the (L.P.) problem. Set $x^*_j = 1$ if $x^{opt} = 1$ and $x^*_j = 0$ otherwise.

Let $k^*_i = \text{Max}\ \{0, k - \Sigma a_{ij}\ x^*_j\}$. Construct A* as follows: Remove from A all columns j such that $x^*_j = 1$ and remove all rows with $k^*_i = 0$. Set $k^* = \{k^*_i | k^*_i \neq 0\}$. You have now a reduced problem.

It is again an integer program.

$$\begin{aligned} & \text{Min } \Sigma x_j \\ \text{(I.P.S.1)}\quad & \text{s.t } A^* x \geq k^* \\ & \qquad x_j = 0 \text{ or } 1 \end{aligned}$$

The problem is *substantially* reduced in size. We can now use an integer programming algorithm.

Scheme 2. Once the solution to the L.P. is obtained. A branch-and-bound algorithm can be developed to be continued until a sufficiently small difference exists between the best solution and the smallest upper bound obtained so far. Note that by the remarks preceeding Scheme 1 it is always very easy to obtain upper and lower bounds.

There is also the possibility to use "integer cuts", see [7]. This *is* integer programming. However, there is a possibility that for k-cover problem, good cuts can actually be constructed and recognized. That is one direction of research (that might become imperative) but which has not been pursued. Success in characterizing cuts has been achieved in the past for highly structured problems, see [2] and [5].

d. *A Network Flow Problem with Concave Cost*: For the sake of completeness we record one other formulation. The problem is to find a feasible flow (in the network to be described below) at minimum cost. One possible advantage is the potential use of algorithms specifically developed for fixed charge problems or branch-and-bound methods on nets, see e.g., [9] and [14]. We consider the bipartitie graph N as described in the beginning of this section (2), Figure 3 to which we add a source s and a link S. Let f (x,y) be the blow between nodes x and y, we have the following constraints:

$f(x,y) \geq 0$ for all x,y

$f(s,y) \leq c(s,y)$ for all $y \in R$

$f(x,S) \geq p(x,S)$ For all $x \in T$

and $f(x,y)$ integer

The upper bound c (s,y) is set equal to the valence of $y \in R$ minus 1 and for all $x \in T, p(x,S) = k$. The cost $a[f(x,y)] = 0$ for all arcs except for arcs (s,y), $y \in R$ where.

$a[f(s,y)] = 0$ if $f(s,y) = 0$

$a[f(s,y)] = 1$ otherwise.

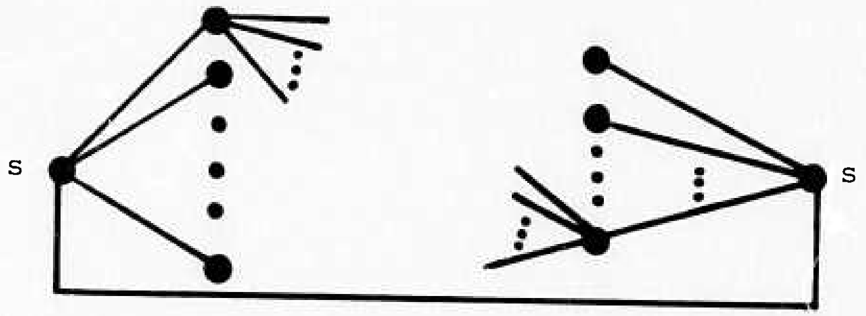

FIGURE 3

BIPARTITE GRAPH WITH SOURCE AND LINK

e. A Heuristic Approach to the k-cover Problem. Given the limited success of integer programming algorithms in solving large scale problems, we have been led to consider heuristic methods to find good solutions to the k-cover (of terminals by repeaters) problem which is typically large scale. It is intuitively appealing to consider a terminal as particularly critical if it is adjacent to few repeaters. (In the extreme cases, if a terminal has fewer than k adjacent repeaters, the problem is infeasible and if it has exactly k adjacent repeaters, all of them must be chosen for any feasible solution.) Similarly, a repeater is desirable if it is adjacent to a large number of terminals, especially if the terminals are highly critical. The heuristic algorithms described below systemizes these intuitive notions in the search of a "good" solution.

Again, consider the problem in matrix form, but this time we imbed the problem in a more general class where we can require a different cover multiple for each terminal. The more general problem is:

(I.P.H.) Min $\Sigma_j x_j$,

such that $\Sigma_j \; a_{ij} \; x_j \geq k_i, \; i = 1,\ldots,|T|$,

with $x_j = 0$ or 1,

where k_i represents the cover multiple required for terminal i. If $k_i \leq 0$, then no repeater is needed to cover terminal i.

One iteration of the heuristic method consists in passing from a problem of the type (I.P.H.) to an "equivalent" problem by fixing one of the variables, say x_s, as its upper bound 1, and putting the index s in an index set J. This implies the selection of the corresponding repeater. The new problem, denoted by $(I.P.H.)_n$ is obtained from (I.P.H.) by adjusting the matrix $A = [a_{ij}]$ and the cover requirement vector $k = [k_i]$ as follows: the column $A^s = [a_{is}]$ is deleted from A and the adjusted vector $\bar{k}$ is given by $\bar{k}_i = k_i - a_{is}$. The variable x_s no longer appears in $(I.P.H.)_n$. The algorithm terminates if at any iteration it is recognized that for the adjusted A and k, for some i, $\Sigma_j\, a_{ij} < k_i$, the problem is then <u>infeasible</u>; or as soon as all the adjusted k_i, $i = 1,\ldots,T$, become non-positive. In the latter case, the problem is <u>feasible</u>, the adjusted problem is optimized by setting all remaining $x_j = 0$, (for $j \notin J$). A feasible solution $\bar{x}$ to the original is obtained by setting $\bar{x}_j = 0$ if $j \notin J$ and $\bar{x}_j = 1$ if $j \in J$. The vector $\bar{x}$ is called the <u>heuristic solution</u>.

The equivalence of the new problem $(I.P.H.)_n$ and the earlier version (I.P.H.) depends naturally on the choice of the variable x_s. If x_s is 1 in an optimal solution to (I.P.H.), then the two problems are equivalent in the sense that;

$$\text{Min of (I.P.H.)} = [\text{Min of } (I.P.H.)_n] + 1$$

Equivalence of the set of optimal solutions is guaranteed only if $x_s = 1$ in every optimal solution of (I.P.H.) (we must ignore x_s when considering optimal solutions to (I.P.H.)).

The selection criterion to choose a variable at each iteration x_s (or equivalently the index s) to be fixed at value 1 can be viewed as a function, called σ, of the adjusted matrix A and vector k with values in the index set $\{j\}$, i.e.,

$$\sigma: (A,k) \mapsto s \ , \ s \in \{j\}.$$

There obviously exists a function σ - a selection criterion which will guarantee the equivalence of $(I.P.H.)_n$ and (I.P.H.) at each iteration and consequently, the optimality of the resulting heuristic solution. However, applying this function σ to $[A,k]$ might involve no less than solving (I.P.H.) We use a heuristic motivated by the considerations mentioned at the beginning of this section. The adjusted A and k are used to compute the "probability" that a given x_j belongs to the optimal solution, by this we mean that to each column of the adjusted matrix A we associated a nonnegative number ω_j such that;

$$P_j = \omega_j . (\Sigma\ \omega_j | j \not\in J)^{-1}$$

represents very loosely speaking - the probability that x_j belongs to the optimal solution. We select x_s, $s \not\in J$ if $P_s \geq P_j$ for $j \not\in J$ or equivalently if $\omega_s \geq \omega_j$ for $j \not\in J$. The selection criterion σ will be completely determined if we specify a method to compute the ω_j.

In the selection of these weights ω_j, we must take into consideration the effort involved in the computation as well as the reliability of the resultant selection. We have used four such weights:

We first define;

$$k_i^* = 1/2\ [|k_i| + k_i]$$

Observe that $k_i^* = 0$ if $k_i \quad 0$ and $k_i^* = k_i$ otherwise.
We set:

$$\omega_j^{(1)}\ (A,k) = \Sigma_i \left(\frac{k_i^*}{\Sigma_\ell\, a_{i\ell} - k_i^*}\right) a_{ij} \qquad j \not\in J$$

$$\omega_j^{(2)}\ (A,k) = \sum_i \left(\frac{k_i^*}{\sum_\ell a_{i\ell}}\right) a_{ij} \qquad j \not\in J$$

$$\omega_j^{(3)}\ (A,k) = \sum_i \left(\frac{1}{\sum_\ell a_{i\ell} - k_i^*}\right) a_{ij} \qquad j \not\in J$$

$$\omega_j^{(4)}\ (A,k) = \sum_i \left(\frac{1}{\sum_\ell a_{i\ell}}\right) a_{ij} \qquad j \not\in J$$

The entries between parentheses ( ) in the definition of weights ω_j, measures how critical terminal i is. If $k_i^* = 0$, the terminal does not need covering then $k_i^*/(\sum_{j\varepsilon J} a_{ij} - k_i^*) = 0$ (for our purposes $0/\infty = 0$). On the other hand, if $\sum_j a_{ij} = k_i$, namely there are exactly enough repeaters to cover the terminal then the weight is infinite. A repeater will be preferred to another one if it covers more critical terminals.

<u>Computational Experience</u>: The size of test problems solved varies from problems with as few as 5 repeaters and 5 terminals to problems with as many as 400 repeaters and 400 terminals. Roughly speaking, the computation time was directly proportional to the size of the adjacency matrix A and the cover multiple required. The computer used was a PDP-10 (time sharing). The larger problems (400 repeaters, 400 terminals, 2 - cover) were solved in 70 sec. or less. The time, as may be expected, is dependent on the density of 1's in the incidence matrix A. Thus, the maximum time recorded arose from terminals - repeaters configuration where each repeater covers many repeaters. The running time is of the order of $|T|x|R|^2$ where $|T|$and $|R|$ and the number of terminals and repeaters respectively.

We ran a number of problems with the heuristic code and for comparison with the Ophelie mixed integer programming code running on a CDC 6600 computer. The Ophelie code uses the branch-and-bound method. In the case of very simple problems (8 repeaters, 9 terminals, 2 - cover) there was essentially no difference in running time (presumably most of the time, less than .5 sec, was spent in setting up the problem). Running experience with the Steiner triples' problem described in the next section, yields a ratio of 500 to 1 between the Ophelie time and the heuristic code time when solving the smaller problem A_{27} (117 terminals, 30 repeaters, 1 - cover), and no comparison is available for the larger problem A_{45} (330 terminals, 45 repeaters) since for example the MPSX code failed to reach a solution in more than one half hour on a IBM 360-91.*

Comparison in running time is naturally not completely valid, since most of the computation time in the Ophelie code can be spent just checking if a given solution is optimal. The heuristic method does not try to check the optimality of its solution. However, in general, results with the heuristic code have been extremely good When the heuristic solution deviated from the optimal solution, the problem usually involved numerous ties for the maximum $\omega_j^{(\ell)}$ ℓ=1,2,3,4, such as in the Steiner triples' problems. In all problems that were generated to resemble the packet radio terminal - repeater problem, the heuristic algorithm reached the optimal solution (in those problems for which we are able to determine the optimal solution).

The running time was unaffected by the choice of any of the selection criteria but for "hard" problems, we obtained consistently better solutions when using $\omega_j^{(1)}$ and $\omega_j^{(2)}$ rather than $\omega_j^{(3)}$ and $\omega_j^{(4)}$.

<u>The Steiner Triples' Problem</u>: Fulkerson, Nemhauser and Trotter [6] report on two covering problems which they characterize as computationally difficult. In each problem, the matrix A is the incidence matrix of a Steiner triple system. The first problem, labelled A_{27} is a 1-cover problem with 117 terminals and 30 repeaters.

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* Private Communication, R. Fulkerson, June, 1974.

The second problem, labelled A_{45}, has 330 terminals and 45 repeaters and is also a 1-cover problem. Data for both problems can be found on pages 9 and 10 of [6]. The problems are considered to be difficult because the large number of verifications (branching in branch-and-bound, cuts in cutting methods) required to establish that a given solution is in fact optimal.

In our runs, the variable to be fixed at 1 (the selected repeater) at each iteration was selected by using the criterion resulting from using weights $\omega_j^{(2)}$. In the case of ties for the maximum weight ω_j, the variable with smallest index was chosen. Due to the inherent symmetries present in these problems, numerous ties did occur. For example, all weights are equal in the first iteration. Thus, the tie breaking rule plays a relatively important role in the selection of a solution. We solved both problems 100 times breaking ties by random selection among all tied variables.

The frequency of the values generated by the heuristic solutions is recorded in the table below. In all cases the heuristic obtained the optimum solution for the smaller problem A27.

Heuristic Minima	30	31	32	34
A_{45}	3	44	29	24

The total running for 100 solutions for A_{27} (including the generation of random numbers to break ties) required less than 1/5 of the time required to solve A_{27} by branch-and-bound (even giving the optimal solution as a starting solution as recorded in [6]. Approximately 5 sec. were necessary to obtain a heuristic solution to the larger problem A_{45}. (The branch-and-bound algorithm failed to produce a solution to A_{45}. H. Ryser has conjectured that the optimal solution to A_{45} has 30 repeaters [10].

How Accurate is the Heuristic Method: Unfortunately, we can not obtain significant bounds on the error for the heuristic method using any of the weights $\omega_j^{(\ell)}$ $\ell=1,\ldots,4$(that determine the selection criterion.) We give here an example, developed in collaboration with Professor Robert Bixby which shows that the error can be arbitrarily large. The example is a 1 - cover problem. Let $T_o,\ldots,T_n$ be disjoint sets of indices with the cordinality of $|T_i| = 2^i$. Set $T = \cup T_i$. Terminals are all pairs of indices (t, 1) and (t, 2) with $t \varepsilon T$. There are n + 3 repeaters. Repeaters R_i, i=0,1,...,n are connected to all terminals with indices (t,1) and (t,2) with $t \varepsilon T_i$. Repeaters R_{n+1} and R_{n+2} are connected to terminals with indices $\{(t, 1)|t \varepsilon T\}$ and $\{(t, 2)|t \varepsilon T\}$, respectively. For n = 3, the matrix A appears on the next page.

Observe that each row contains exactly 2 nonzero entries, thus $\sum_j a_{ij} = 2$ for all i. It is easy to verify that

$$\omega_j^{(1)} = \sum_j a_{ij} \left(\frac{1}{2-1}\right) = 2|T_j| = 2^{j+1} \qquad j = 0,\ldots,n$$

and

$$\omega_{n+1}^{(1)} = \omega_{n+2}^{(1)} = \sum_j a_{ij}(1) = \sum_{j=0}^{n} |T_j| = 2^{n+1} - 1.$$

Selecting the variable (index) with maximal weight implies that the choice will be repeater R_n. Eliminating x_n and the corresponding column as well as the rows corresponding to terminals covered by R_n, we obtain a new problem of exactly the same type as the original problem. The previous argument is independent of the value

	R_0	R_1	R_2	R_3		R_4	R_5
$(T_0,1)$	1					1	
$(T_1,1)$		1				1	
		1				1	
$(T_2,1)$			1			1	
			1			1	
			1			1	
			1			1	
$(T_3,1)$				1		1	
				1		1	
				1		1	
				1		1	
				1		1	
				1		1	
				1		1	
				1		1	
$(T_3,2)$				1			1
				1			1
				1			1
				1			1
				1			1
				1			1
				1			1
				1			1
$(T_2,2)$			1				1
			1				1
			1				1
			1				1
$(T_1,2)$		1					1
		1					1
$(T_0,2)$	1						1

= A

MATRIX A

of n, which implies that repeater R_{n-1} will be selected next, and so on. The heuristic solution is thus given by

$$x_n = x_{n-1} = \ldots\ldots = x_o = 1 \text{ and } x_{n+1} = x_{n+2} = 0$$

It is, however, easy to see that the optimal solution is

$$x_n = x_{n+1} = \ldots\ldots = x_o = 0 \text{ and } x_{n+1} = x_{n+2} = 1$$

The value of the optimal solution is 2 whereas the value of the heuristic solution is n + 1. (The same arguments apply for all the weights $\omega_j^{(\ell)}$ $\ell=1,\ldots,4$, which we use to determine the selection criteria).

The example shows that there are problems for which this heuristic method fails to produce an optimal. By itself, that is not surprising, since it is a heuristic method with no guarantee to generate the optimal solution. But more interesting is the fact that it is possible to find problems for which the error is arbitrarily large.

Typically, however, the example is not in the category of problems that one expects to encounter in the location of repeaters for a packet radio system. There are no ties, at <u>any</u> iteration, but near-ties. At each step we almost choose R_{n+1} and R_{n+2}, in the sense that for all n, R_{n+1} and R_{n+2} would be the second choice.

f. <u>The PaAl Example</u>. One of the test problems used to compare various techniques to solve the k - covering problem, is the PaAl problem. This problem was generated by using real data obtained from a topographical map for the region of Palo Alto. This part of the U.S. was selected because it contained many interesting topographical attributes: a flat terrain (salt flats, the region surrounding the Bayshore Freeway), an urban center (Palo Alto and neighboring communities) on slightly sloping terrain and finally a hilly region (with valleys, small plateaus, etc.). Moreover, at this time, it appears that a reduced scale experiment of a packet radio network will be installed in the Palo Alto area. The purpose of this section is to give a description of the design of this model.

Location: We decided to limit the investigation to the area covered by the topographical map known as the Palo Alto Quadrangle, California, 7.5 minute series (topographic), U.S. Department of the Interior, Geological Survey or equivalently to the area lying between meridians 37° 22' 30" N. and 37° 30' N. and longitudes 122° 15' W. and 122° 07' 30" W, see Figure 6.

Terminals and Repeaters: The map was divided in 180 cells (squares) obtained by dividing the meridian direction (height) in 15 equal parts and the longitudinal direction (width) in 12 equal parts, see the map reproduced below. Each rectangular subregion measures .9356 km in height and .9356 km in width which yields a total surface area of .87533 km^2 (or approximately .35 $miles^2$). Forty two (42) locations were singled out as potential locations for repeaters. In the hilly part of the map, the Southwest region, the high-points were selected, such as top of hills, location of water towers, smaller but prominent points overlooking valleys, etc. In the city, a number of high rise constructions were singled out as potential location such as radio towers, high rise apartment or office buildings, etc.

Connections Between Terminals and Repeaters: By definition, a cell i was declared to be covered by repeater j if a terminal located at the worst possible location in that cell i was in line of sight (LOS) of the repeater j. (In some cases, it turned out that a repeater located in a given cell k covers cell j but a repeater located in cell j did not cover cell k).

LOS Computation: To determine if a terminal at location j can be seen from a repeater at location k, we proceeded as follows. It was assumed, that if no particular high construction (building, water tower, etc.) was available to install the repeater's antenna, it would be installed at 30 feet above the ground level (making use of a tree, telephone pole, etc.). The terminals were assumed to be 5 feet above ground level. The points were said to be in LOS if the first Fresnel Zone associated with transmission between these two points was free of any obstacle.

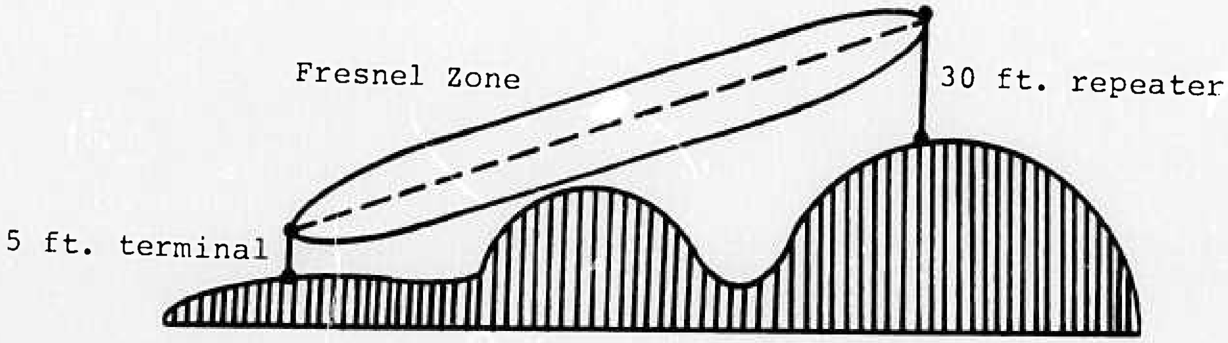

FIGURE 4

To compute the Fresnel Zones, we sssumed that transmission would occur at 1500 MHz corresponding to a wave length λ = .2m (7.87in.). We give here an example of such a Fresnel Zone, transmitter and repeater are assumed to be 5 km apart. In the figure on the next page, we give the radius for certain cross-sections of the Fresnel Zones (λ = .2m).

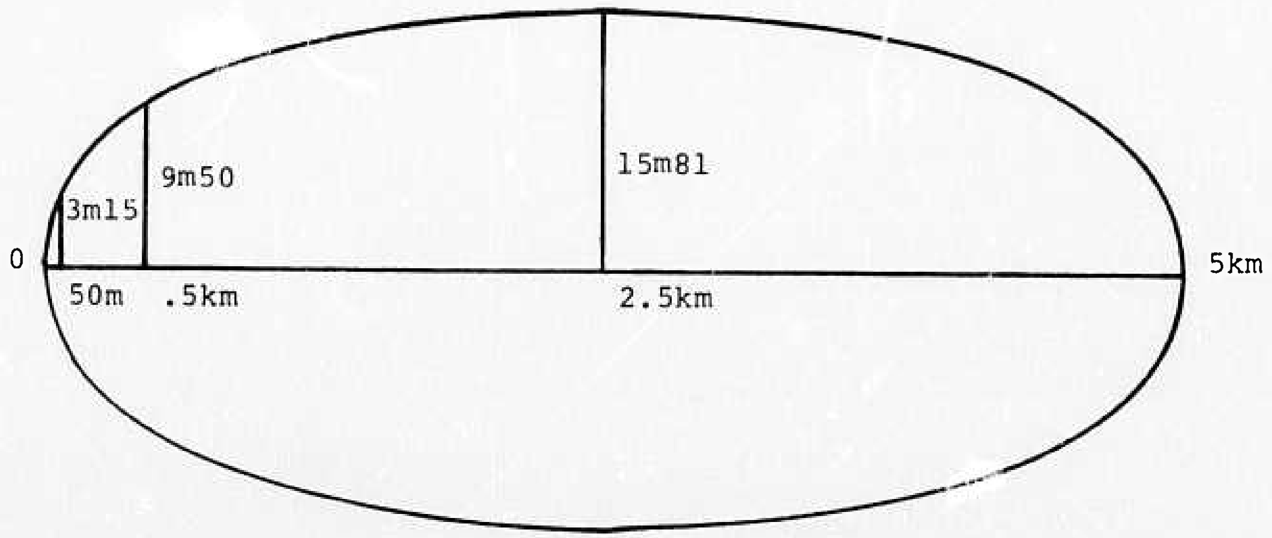

FIGURE 5

FRESNEL ZONE CROSS SECTIONS

Transmission radius is supposed to be less than 20 km (not an upper bound here since the greatest distance between any two points in this region is less than 18 km). In the urban area the maximum transmission radius was assumed to be 7 km.

Cover Multiple: In view of the fact that the region selected was a subregion of the area to be covered eventually by the packet radio network, we made some arbitrary decisions as to the boundary cells. Since they will probably also be covered by repeaters located outside the Palo Alto Quadrangle, we are requiring that these boundary cells be 1-covered rather than 2-covered as the other cells.

Computational Results: The PaAl problem, described above was solved by the heuristic algorithm, given the code name SETCOV, and by OPHELIE. (A rapid analysis of the terminal-repeater adjacency matrix shows that none of the optimal solutions would have been generated if one had used the more simplistic approach of selecting the repeater with highest adjacency degree. Such a selection yields quite different answers requiring a larger number of repeaters).

For the PaAl problem, the optimal solution requires the installation of 14 repeaters (different runs with SETCOV showed that there were in fact a number of optimal solutions with 14 repeaters). The total running time for OPHELIE was approximately 12 CPU sec. excluding set up time. The SETCOV required 3 sec. to produce a solution. The relative success of the OPHELIE code must, at least in part, be attributed to the fact that the linear programming solution (which is used to initiate the branch-and-bound part of the code) is actually the optimal solution. (If this is just an isolated phenomena to be associated with this particular problem or is actually a characteristic of this whole class of problems is not known). One optimal allocation of repeaters consists in selecting sites: 4, 5, 7, 11, 16, 17, 18, 27, 28, 30, 34, 36, 37, and 42.

We also solved a variant of the model described above. The presence of (small) ridges in cells 8, 23, and 54 combined with our model design rule - a cell is covered by a repeater if the worst location is that cell (subregion) can communicate with that repeater - renders these three cells "critical" in the sense that there are exactly one (for cell 8) and two (for cells 23 and 54) repeaters covering these cells. This results in the automatic selection of certain repeaters. To avoid this somewhat peculiar situation, we formulated a variant of the PaAl problem requiring no lower bound on the number of covers for cells 8, 23, and 54. This problem was also solved by OPHELIE and SETCOV. Running times were of the same order than before. The optimal solution only requires 12 repeaters this time. Both codes produced the optimal solution, with OPHELIE obtaining again the optimal solution in the LP part of the problem and the SETCOV using weights $\omega_j^{(3)}$. One optimal allocation of repeaters consists in selecting sites: 4, 7, 11, 17, 18, 26, 28, 30, 31, 36, 37, 42. An Ophelie solution is depicted in Figure 7.

g. <u>The Generalized k-covering Problem</u>. It is not always plausible to assume that the installation and maintenance costs associated with various repeaters at different locations is the same. This shortcoming of the previous model is overcomed by associating different cost to repeaters in the objective (of (I.H.P.)). An obvious adaptation of the heuristic method described in Section 3 replaces the weights $\omega_j^{(1)}$ (i) = 1, ..., 4 used in before by $\omega_j^{(i)}/c_j$ where c_j is the cost (≥ 0) associated with repeater j.

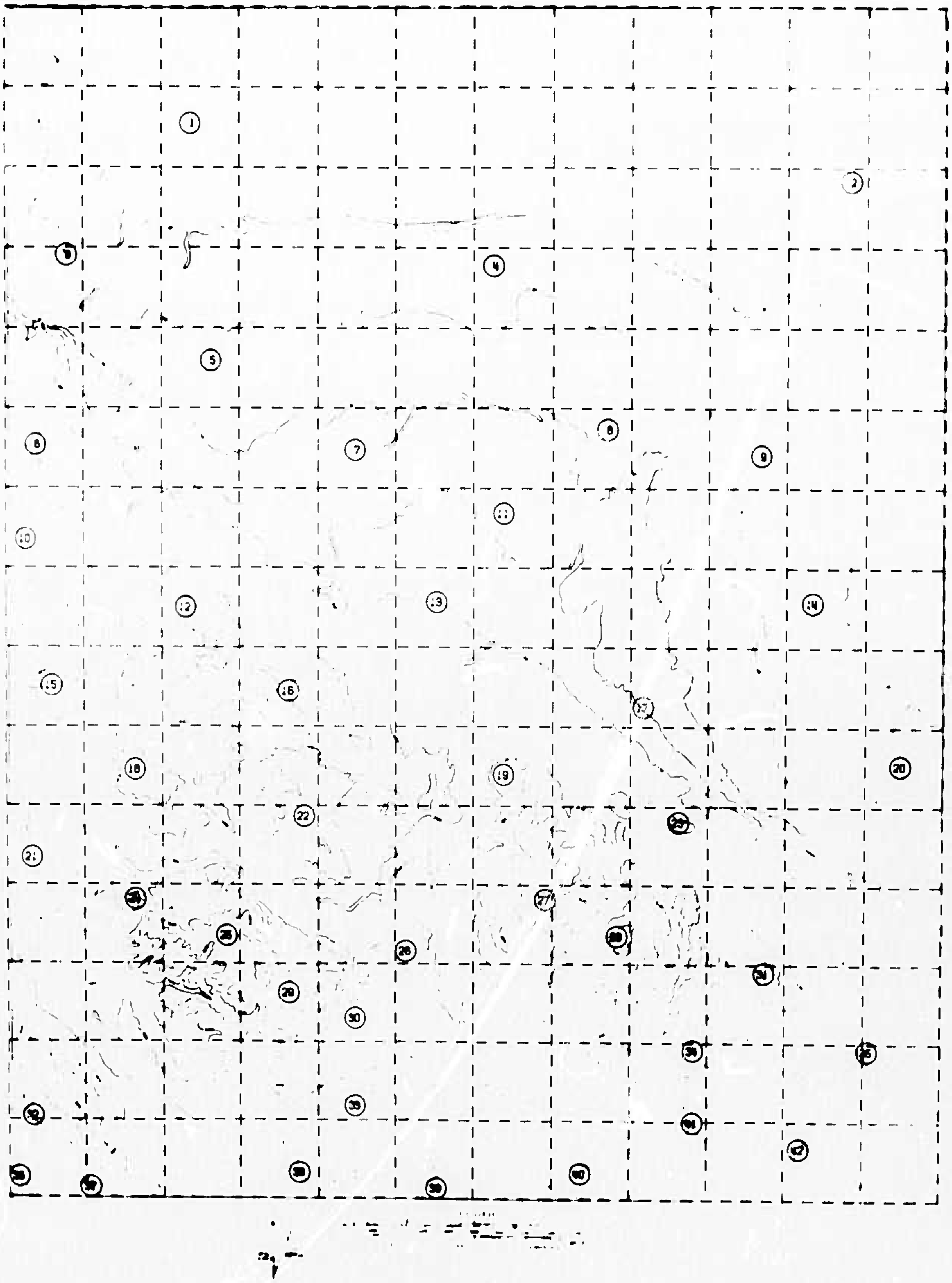

FIGURE 6

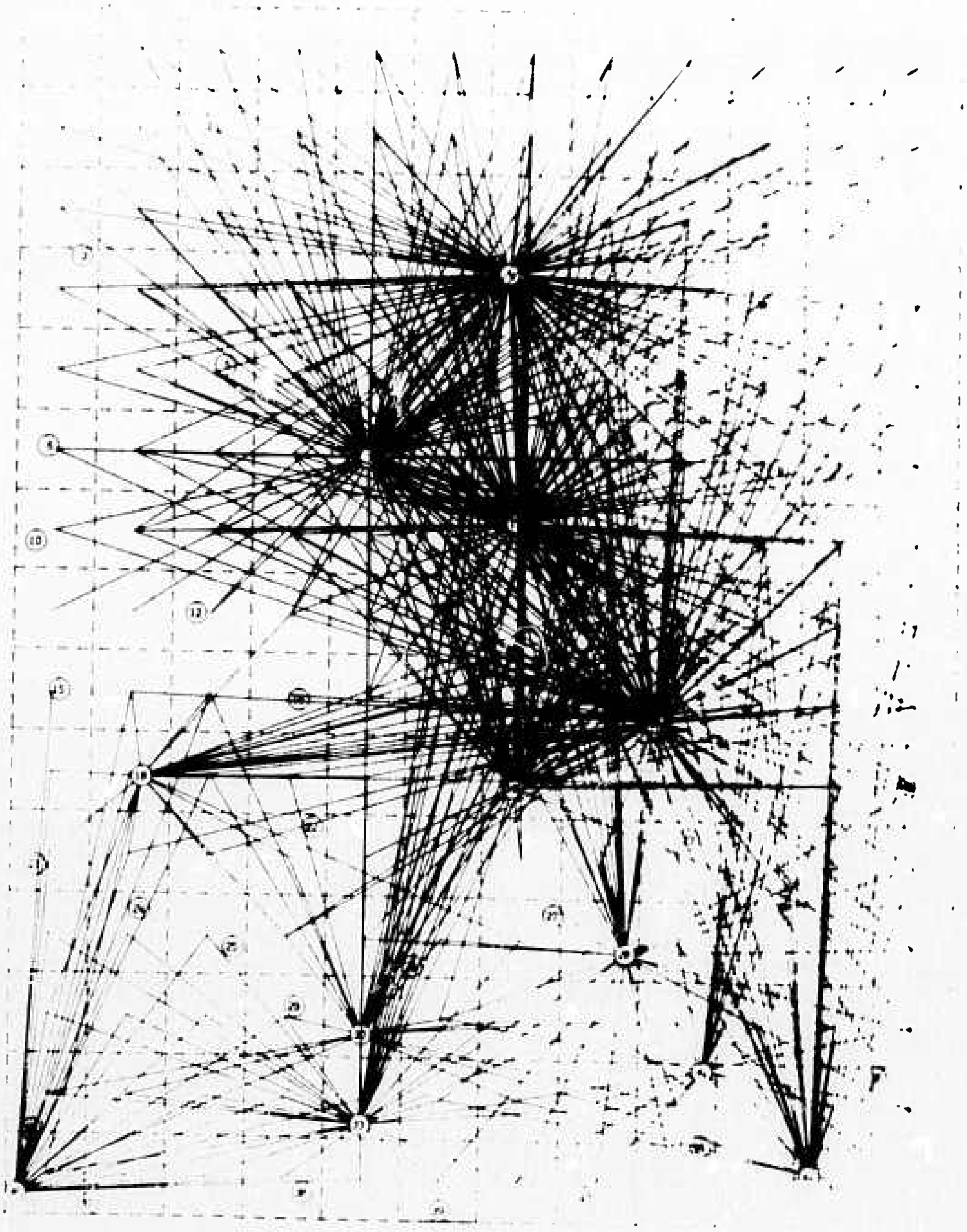

FIGURE 7

III. THE "FLAT TERRAIN" MODEL. A PROPOSED SOLUTION

a. Problem Formulation.

It is assumed that:

(i) The terrain is nearly flat

(ii) The installation (and maintenance) expenses for repeaters are independent of location.

(iii) Transmission characteristics are invariant with location. A repeater communicates with a terminal if and only if they are less than a fixed distance d_t apart. The maximum distance for communication between repeaters is assumed to be d_r (In practice d_r is substantially larger than d_t because repeater antennas are higher than terminal antennas). The area coverage by L.O.S. radio can again be separated into two parts, a covering problem and a connectedness problem (see Section 1 Problem II). Let P be the 2-dimensional plane.

Covering Problem. Find a minimal covering of P by discs of radius d_t such that every point of P is covered at least k times.

Connectedness Problem. Let G = (N,E) be the graph obtained as follows: The nodes N are the centers of the discs used in the minimal covering of P. The edges E of G are obtained by connecting two nodes if their distance is less or equal to d_r. The graph G is to be q-connected (reliability).

Since we are considering an infinite plane one can no longer define minimality of a cover in terms of its cardinality. There are various proceedures to define minimality, for example, the cover with the smallest percentage of area wasted or if $\lim_{r\to\infty} \frac{1}{r^2} \delta_c(r)$ is minimized over the space of all covers C of P that satisfy the covering and connectedness constraints, where $\delta_c(r)$ is the number of discs of C whose interior intersect a circle of radius r centered at the origin (an arbitary but fixed point of P).

b. Solutions, A conjecture. An optimal solution to the above problem is known for k = 1, $q \leq 6$ and $d_r/d_t \geq \sqrt{3}$ (that is the maximum distance for transmission between repeaters is at least 73.3% larger than that between terminals and repeaters). The problem is then the standard covering of the plane by discs of fixed radius and with least overlap. In [12], it is shown that the optimal solution is given by arrangement found below, which consists in placing a circle of radius d_t at each vertex of a regular triangular tessalation whose grid points (vertices) are $d_t \sqrt{3}$ apart. Since $d_r \geq d_t \sqrt{3}$ it follows that the resulting graph G contains as subgraph the regular triangular tesselation whose grid points (vertices) are $d_t \sqrt{3}$ apart which is 6-connected. In Figure 8 the area of the discs covered twice is shaded.

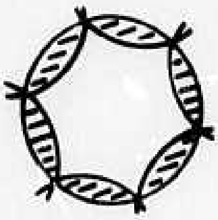

FIGURE 8

We do not consider any other cases for k = 1 since $q \leq 6$ and $d_r \geq d_t \sqrt{3}$ will always be satisfied in practice.

For k = 2, the optimal solution is not known (whatever be q and d_r/d_t). However, due to the inherent symmetry of the problem, we are ready to conjecture that the centers of the optimal solution produces a regular grid of points in P. Making this conjecture our working hypothesis, there are only three cases to consider, the three regular tesselations: (i) the tesselation of P by equilateral triangles, (ii) by squares, (iii) by hexagons [2].

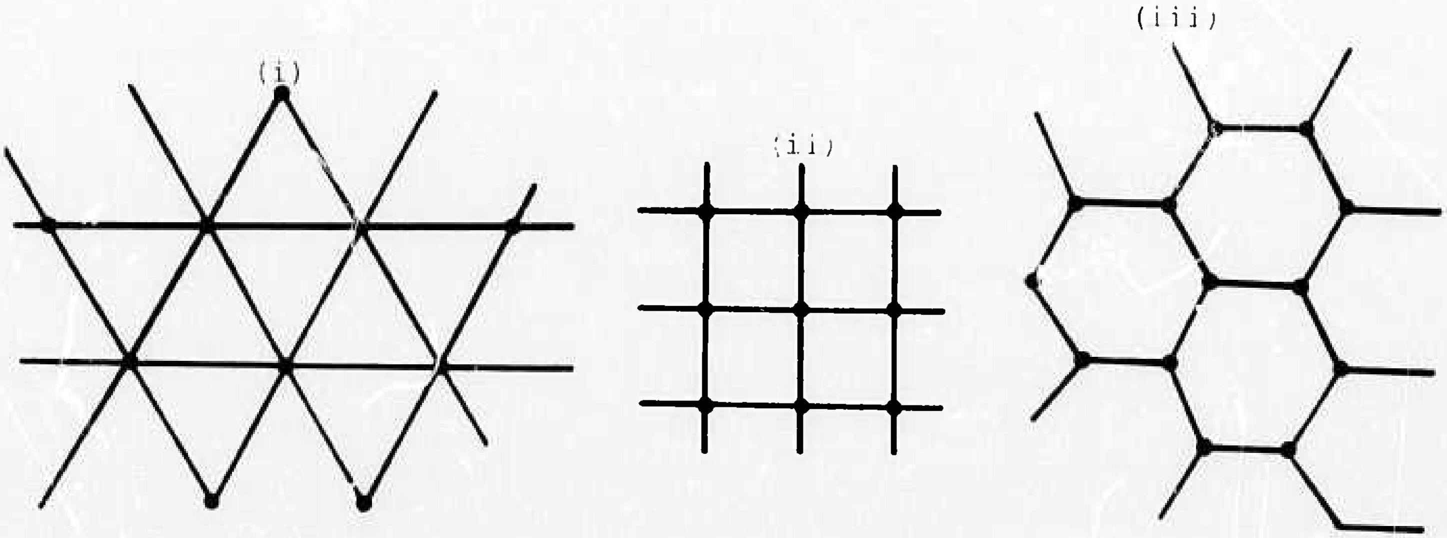

FIGURE 9 THE REGULAR TESSELATIONS

Each vertex of the tesselation corresponds to a repeater and if every point in P is to be covered twice, then the distance between two adjacent repeaters should not exceed d_t. Assuming that d_t is the distance between two adjacent vertices of the tesselations, then in the triangular tesselation one finds 6 repeaters at distance d_t, 6 repeaters at distance $\sqrt{3}d_t$, 6 repeaters at distance $2d_t$ ... from any given repeaters. In the square tesselation one finds 4 repeaters at distance d_t, 4 repeaters at distance $\sqrt{2}d_t$, 4 repeaters at distance $2d_t$ ... from any given repeaters. Finally, for the hexagonal tesselation, there are 3 repeaters at distance d_t, 6 repeaters at distance $\sqrt{3}d_t$, 3 repeaters at distance $2d_t$, ... from any repeater. With each tesselation one can associate a <u>repeater density</u>. It is easy to see that there is 1 node:

per $\sqrt{3}/2\ d_t^2$ units of area in case (i)

per d_t^2 units of area in case (ii)

per $\frac{3\sqrt{3}}{4}\ d_t^2$ units of area in case (iii)

Assuming that $\frac{3\sqrt{3}}{4}\ d_t^2$ is 1 unit of area it follows that <u>the repeater density</u> is

1.5 for triangular tesselations
1.293 for square tesselations
and 1 for hexagonal tesselations

In other words, design (i) requires 50% more equipment than (iii) and (ii) require ∿30% more equipment than (iii). Obviously, if a regular grid yields the optimal solution then the one created by the hexagonal tesselation (iii) is the optimal solution. If $q \leq 3$ and $d_t \leq d_r$ then the corresponding graph G contains the subgraph given by the tesselation (iii) which is obviously 3-connected. If $d_r \geq \sqrt{3}\ d_t$ and $q \leq 6$ again the connectedness constraint is satisfied.

One should observe some inefficiencies in this covering. In particular some areas are covered by three separate repeaters rather than 2 (as would ideally be the case). If we define the thickness of a cover as being the average "thickness" of the layer of discs covering the plane, then the optimal solution for k=1 has thickness 1.209 whereas the thickness of the conjectured optimal solution for k=2 is 2.418. Thus, in both cases, we have a 21% "waste". In Figure 10, a solution is shown with the area covered three times shaded. The question for k>2 is open.

<u>Remark</u>. The "flat terrain" results yield obvious lower bounds for the "hilly terrain" problem.

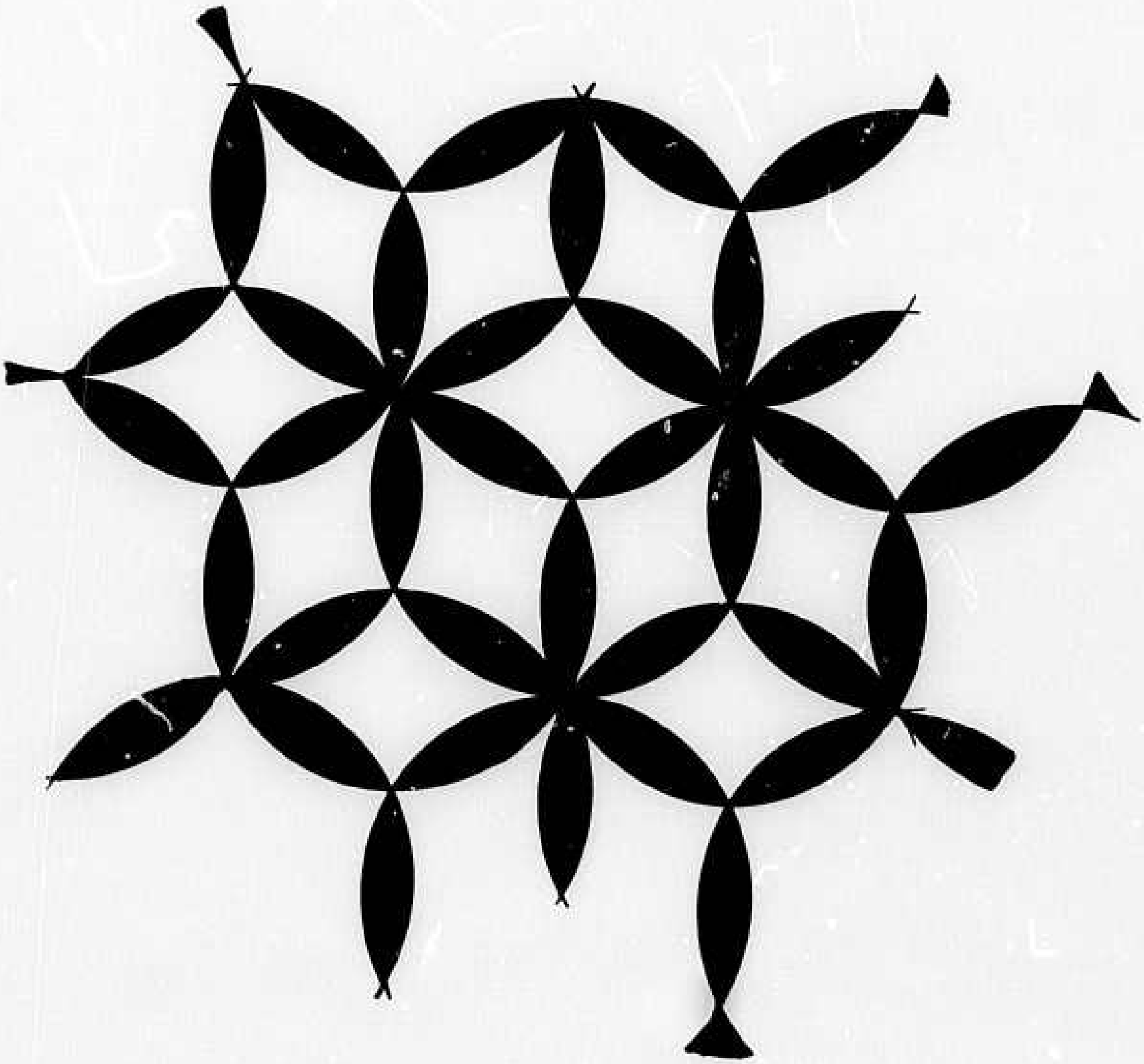

FIGURE 10

SOLUTION TO TESSELTATION OF P BY HEXAGONS

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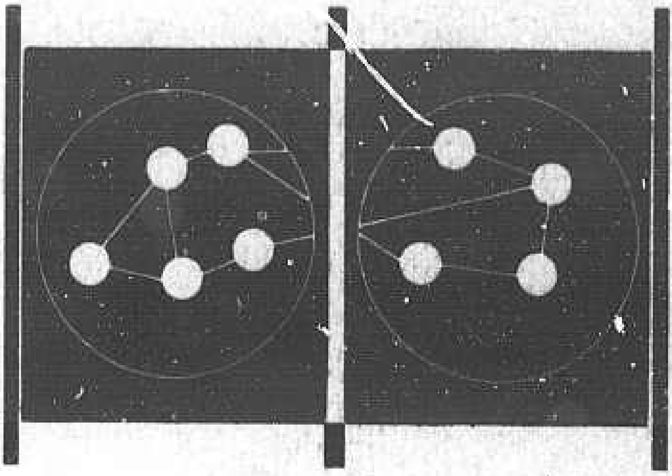
The Practical Impact of Recent Computer Advances on the Analysis and Design of Large Scale Networks

Fourth Semiannual Technical Report



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NETWORK ANALYSIS CORPORATION

Unclassified
Security Classification

DOCUMENT CONTROL DATA - R & D

(Security classification of title, body of abstract and indexing annotation must be entered when the overall report is classified)

1. ORIGINATING ACTIVITY (Corporate author) Network Analysis Corporation Beechwood, Old Tappan Road Glen Cove, New York 11542	2a. REPORT SECURITY CLASSIFICATION Unclassified 2b. GROUP None

3. REPORT TITLE
Fourth Semiannual Technical Report, December 1974, for the Project "The Practical Impact of the Recent Computer Advances on the Analysis and Design of Large Scale Networks"

4. DESCRIPTIVE NOTES (Type of report and inclusive dates)
Fourth Semiannual Report, December 1974.

5. AUTHOR(S) (First name, middle initial, last name)
Network Analysis Corporation

6. REPORT DATE December 1974	7a. TOTAL NO. OF PAGES 190	7b. NO. OF REFS 60

8a. CONTRACT OR GRANT NO. DAHC15-73-C-0135	9a. ORIGINATOR'S REPORT NUMBER(S) Semiannual Report 4B
b. PROJECT NO. ARPA Order No. 2286	9b. OTHER REPORT NO(S) (Any other numbers that may be assigned this report)

10. DISTRIBUTION STATEMENT
Approved for public release; distribution unlimited

11. SUPPLEMENTARY NOTES None	12. SPONSORING MILITARY ACTIVITY Advanced Research Projects Agency, Department of Defense

13. ABSTRACT

This report summarizes studies of local, regional, and large scale data communication network problems. Primary emphasis is placed on system issues and tradeoffs. Areas discussed are packet radio system studies, packet radio system algorithms and control, local and regional data network cost comparisons and alternatives, and integrated large scale packet switched network cost and performance. Studies of multidropped, point-to-point, and broad band cable and radio broadcast systems as well as the impact of satellites on network cost and performance are described.

PRICES SUBJECT TO CHANGE

14. KEY WORDS
Computer networks, throughput, cost, reliability, packet switching, packet radio, broadcast radio.

i.

DD FORM 1473 (PAGE 1) PLATE NO. 21856

S/N 0102-014-6600

Unclassified
Security Classification

Fourth Semiannual Technical Report
December 1974

For the Project

The Practical Impact of Recent Computer Advances on the Analysis and Design of Large Scale Networks

Principal Investigator
and Project Manager:

Howard Frank (516) 671-9580
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ARPA Order No. 2286

Contractor: Network Analysis Corporation
Contract No. DAHC15-73-C-0135
Effective Date: 13 October 1972
Expiration Date: 31 December 1974

Sponsored by

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SUMMARY

Technical Problem

Network Analysis Corporation's contract with the Advanced Research Projects Agency has the following objectives:

- To study the properties of packet switched computer communication networks for local, regional and large scale communications.
- To develop techniques for the analysis and design of large scale networks.
- To determine the cost/throughput/reliability characteristics of large packet-switched networks for application to Defense Department computer communication requirements.
- To apply recent computer advances, such as interactive display devices and distributed computing, to the analysis and design of large scale networks.

General Methodology

The approach to the solution of these problems has been the simultaneous

- Study of fundamental network analysis and design issues.
- Development of efficient algorithms for large scale network analysis and design.
- Development of an interactive distributed display and computational system to deal with large-scale problems.
- Application of the new analysis and design tools to study cost and performance tradeoffs for large systems.

Efforts have concentrated on the following areas:

- Packet Radio System Network Studies.
- Packet Radio System Network Algorithms and Controls.
- Local and Regional Data Network Performance and Cost Comparisons
- Integrated Large Scale Packet Switched Network Cost and Performance.
- Support Facility Development.

Technical Results

This document summarizes principal results obtained over the entire contract period with primary emphasis on the system issues and tradeoffs identified and examined. Algorithms, models, and design tools are discussed in prior semiannual reports. Accomplishments include:

- Base performance characteristics of a single station, fixed repeater location Packet Radio System were established.
- The effects on performance of a number of fundamental hardware design decisions including use of multiple or single detectors at repeaters and stations, tradeoffs between range, power and interference, single or dual rate repeaters, common versus split channel operation, and the use of omni versus directional antennas were evaluated.
- System delay, throughput and blocking under various routing alternatives, acknowledgement schemes, and repeater network organization were quantified.
- It was demonstrated that the Packet Radio System, using unoptimized operating parameters and algorithms, can provide reliable and efficient transportation of packets.
- Efficient routing algorithms were developed, simulated and tested, and based on these tests, recommended for implementation.
- Single station, multirepeater initialization, network mapping, and transmission algorithms were developed.
- Hop-by-hop and end-to-end acknowledgement schemes were developed, simulated and tested.
- Simple terminal search and local terminal control algorithms were developed and simulated.

- A repeater location optimization algorithm was developed, programmed, and tested.
- Low cost terminal access via hardware multiplexing at TIPs was demonstrated.
- The cost-effectiveness of multipoint lines for connecting low and medium speed terminals into ARPANET was shown. The use of software demultiplexing as a means of increasing the terminal handling capacity of a TIP by a factor of 10 was proven.
- The theoretical efficiency of incorporating broadcast packet radio techniques on a wideband coaxial cable local distribution network servicing a large terminal population was shown.
- Basic analysis and design algorithms for optimization of terminal processor location, topological optimization, throughput and delay analysis, and reliability analysis were completed.
- The cost-effectiveness of using satellites to increase ARPANET capacity was proven.
- The feasibility of a 1,000 IMP packet switched network using terrestrial was demonstrated.
- The cost-effectiveness of packet switching within an environment containing several thousand terminals was shown

Department of Defense Implications

The Department of Defense has vital need for highly reliable and economical communications. The results developed prove that packet switching can be used for massive DOD data communications problems. A major portion of the cost of implementing this technology will occur in providing local access to the networks. Hence, the development of local and regional communication techniques must be given high priority. The results on packet radio demonstrate that this technique can provide rapidly deployable, reliable and efficient local access networks. In addition, the initial results on the use of domestic satellites indicates that substantial savings can be achieved by their use in large scale DOD data communications.

Implications for Further Research

Further research must continue to refine tools to the study of large network problems. These tools must be used to investigate tradeoffs between terminal and computer density, traffic variations, the effects of improved local access schemes, the use of domestic satellites in broadcast mode for backbone networks, and the effect of link and computer hardware variations in reliability on overall network performance. The potential of these networks to the DOD establishes a high priority for these studies.

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Chapter 1
SUMMARY OF ACCOMPLISHMENTS

1.1 Introduction

Defense Department communications and computational requirements are global in scope and immensely complex. Communication problems range from the local collection and distribution of information between the components of a single unit in a small geographic area to the global transportation of strategic intelligence and command and control data. Defense Department network problems have become so complex that the unaided human cannot comprehend, let alone solve them effectively. The importance of these problems makes it imperative that effective analysis, design aids and new methods of communication be developed to meet the DOD's rapidly evolving requirements.

Network Analysis Corporation's (NAC) activities under the current contract have dealt with problems in local, regional, and large scale data communications, network analysis, design, cost, performance, and the application of computer techniques to support the analysis and design activities. NAC's efforts can be categorized into the following major areas:

- Packet Radio System Network Studies
- Packet Radio System Network Algorithms and Controls
- Local and Regional Data Network Performance and Cost Comparisons
- Integrated Large Scale Packet Switched Network Cost and Performance
- Support Facility Development

Results accomplished during the contract period in the above areas are summarized in the following paragraphs.

1.2 Packet Radio System Network Studies

Efforts during the past contract year were aimed at establishing base performance characteristics of a single station, fixed repeater location Packet Radio System and to evaluate the effects on performance of a number of fundamental hardware design decisions.

Analytic and simulation studies of throughput and delay were conducted to enable various design decisions including: use of multiple or single detectors at repeaters and stations; evaluation of tradeoffs between range, power and interference; incorporation of single or dual data rate repeaters; common versus split channel operation; and the use of directional versus non-directional antennas. In addition, numerous studies were performed to quantify system delay, throughput and blocking under various routing alternatives, acknowledgement schemes, repeater network organization and to insure that gross system performance using unoptimized operating parameters and algorithms was within a level that would justify further design efforts.

1.3 Packet Radio System Network Algorithms and Control

During the last project year, the main effort has been towards developing workable network algorithms, to insure order of magnitude performance and design robustness for a single station, multiple repeater, multiple terminal network. Preliminary designs of eleven routing algorithms were evaluated using combinatorial analysis. Three were selected for detailed design; two of these were simulated and tested, and based on these tests, recommended for implementation. Single station, multirepeater initialization, network mapping, and transmission algorithms were developed, but have not yet been simulated or tested. Hop-by-hop and end-to-end acknowledgement schemes were developed, simulated and tested. Simple terminal search and local terminal control algorithms were developed and simulated. In addition, a repeater location optimization algorithm was developed, programmed, and tested. The above family of algorithms provided a basis for demonstrating the reliable transmission of packets within the Packet Radio System, but further work is required to improve efficiency, to handle multiple stations, and to increase the number of types of terminals that can be handled by the system.

1.4 Local and Regional Data Network Performance and Cost Comparisons

During the previous contract years, a variety of tools have been developed to allow economical cost-performance tradeoff studies. Studies performed include: the practical demonstration that low cost terminal access can be achieved by hardware multiplexing at TIPs; the proof of the cost-effectiveness of multipoint lines for connecting low and medium speed terminals into ARPANET; the demonstration of the use of software demultiplexing as a means of increasing the terminal handling capacity of a TIP by a factor of 10; and the theoretical calculation of capacity, error rates and delay for a system incorporating broadcast packet radio techniques on a wideband coaxial cable local distribution network servicing a large suburban population.

1.5 Integrated Large Scale Packet-Switched Network Cost and Performance

During the contract period, the groundwork was laid to complete the study of cost and performance tradeoffs in large scale packet-switched networks. Basic analysis and design algorithms for optimization of terminal processor location, topological optimization, throughput and delay analysis, and reliability analysis have all been completed. These programs presently operate in stand alone mode and must be integrated in order to complete the large network studies. In addition, a number of cost performance studies have also been completed. These include studies of the impact of satellites on a 40 node ARPANET, the establishment of the feasibility of a 1,000 IMP packet switched network using terrestrial links, and studies of the cost-effectiveness of packet switching within an environment containing several thousand terminals. These studies are expected to lead to methods for handling large numbers of terminals and processors and various packet access methods implemented within different hierarchy levels of a large integrated C^3 network.

1.6 Support Facility Development

During the past year, a basic packet radio simulator was developed. The simulator handles a single station, up to 48 repeaters and several hundred terminals of the same type. Imbedded in the simulator are models of the repeater, station, and terminals, two routing algorithms, non-persistent carrier sense and unslotted ALOHA random access schemes, zero capture receivers, single and dual data rate channels, omnidirectional antennas, and an interactive terminal to station protocol. All device actions required to initiate, relay, and receive a packet are simulated in the same sequence of events that would occur in the actual packet radio system. Last year's experience showed that for systems like packet radio, interactive, graphical display can greatly reduce the time required to carry out certain forms of system studies such as repeater data rate, power, and operating parameter variations. During the contract period, the first phase of a graphical display system, specifically designed for handling network problems, was developed. In addition, several stand alone analysis and design algorithms and programs have been developed, including a repeater location algorithm and a basic network editor to serve as a front end to the network analysis and design programs. These have not yet been incorporated into the simulator.

The role of the packet radio simulator has been to quantify the network impact of basic device design decisions and to quantitatively demonstrate that the basic packet radio design provided technically effective and reliable packet transmission. The simulator was developed with a general data structure to allow capability for extension, but was used during the past year to deal only with a single station, multirepeater configuration. An important area of work for this year will be to extend the simulator to incorporate up to ten stations.

1.7 Emphasis of This Report

In this report, we summarize the principal results obtained during the contract period. Primary emphasis in this document is placed on the system issues and tradeoffs identified and examined, rather than on the specific algorithms and models developed to evaluate them. Algorithms, models, and computer aided design tools have been extensively described in NAC's Semiannual Reports 1-3 to ARPA.

Chapter 2
MODES FOR DIGITAL NETWORKING

The late 60's and the 70's have seen the proposal and development of an incredible array of digital services. The field is, of course, still dominated by the giants of the common carriers, AT&T and Western Union.

2.1 Common Carriers

AT&T offers a wide array of digital services, many with a long history of development and usage—others in the proposal stage. A short list is given to indicate the types of bandwidths and tariff structures available [34].

- Series 8000 offers up to 48 Kbps for data transmission. The total cost is determined by mileage and service terminal costs.
- Series 5000 (Telpak) type 5700 accommodates a 240 Kbps data rate, type 5800 accommodates a 1000 Kbps data rate. For data transmission, the total bandwidth can be divided into subchannels of the desired bandwidths. The charge consists of a mileage charge and a service terminal charge.
- High-Low (HiLo) density tariff was recently approved by the FCC. Approximately 370 locations are defined to be high density traffic points; the remaining are low density traffic points. For a half-duplex channel, the following interexchange mileage rates apply:

High point-high point:	.85 $/Mile	Mo.
High point-low point or low point-low point:	2.50 $/Mile	Mo.
Short haul (25 miles):	3.00 $/Mile	Mo.

Monthly channel terminal charges are $35 for High and $15 for Low; station terminal charges are $25 for both High and Low.

A low to low connection can be implemented either directly (in which case the low to low direct distance tariff applies) or through two intermediate high density points (in which case different tariffs apply to different segments).

- Digital Data Services (DDS). This is a proposed new data service based on the T1 digital carrier. DDS will initially interconnect 24 major cities and will be progressively extended to the 370 high density traffic cities. FCC approval has been granted for Boston, New York, Philadelphia, Washington, D.C., and Chicago. Cost is based on mileage and service terminal charges which depend on the bandwidth of 2.4, 4.8, 9.6, or 56 Kbps. The T1 carrier will handle up to 1.544 Mbps. The T2 system will operate at four times that rate and the T5 now in the experimental stage, will handle 564 Mpbs on coaxial cable or waveguide.

- Picturephone, although not intended for digital services, does have a bandwidth of 6.312 Mbps and could be a vehicle for the transmission and visual display of data.

Western Union [6] is introducing a hierarchy of time division multiplexing which can either operate over digital radio paths or use modems over analog radio systems. The 1.544 Mbps and 6.3 Mbps channels are designed to be compatible with Bell System T1 and T2 lines respectively. Similarly, the 2.4, 4.8, 9.6, and 56 Kbps speeds are designed to be compatible with anticipated telephone company offerings. Electronic data switches have and are being installed to form a common switching plant for Telex, TWX, and DATA Switching.

2.2 Specialized Common Carriers

A development of crucial interest to the computer industry is the growth of a new class of common carriers--the Specialized Common Carrier. Only a few are listed here. More are appraised by Gaines [25].

Microwave Communications, Inc. (MCI) was a pioneer among the specialized carriers, filing its first common carrier application in 1963. Channel bandwidths of 1,000 Kbps are available and will include 81 cities on both coasts as well as the initial configuration of Chicago, St. Louis, Cleveland, Detroit, Toledo, South Bend, and Pittsburgh. MCI will offer multi-address distribution capability and store-and-forward capability. The total line charge is the sum of the intercity mileage charge, a system access charge, and a channel termination charge, all depending upon bandwidth required.

Data Transmission Company (DATRAN) has innovated in the concept of nationwide digital transmission since 1968 [59]. In April, 1973, the company began construction of its network. The network will offer digital communications between 35 major cities. Users within 50 miles of each such city can be connected to the network by DATRAN facilities. Services will be provided from 2.4 Kbps through 1.344 Mbps. Circuit switching will be used with connection times of less than .5 seconds. Network rates will be similar to DDS in structure.

A number of regional common carriers originally constructed for video transmission are entering the picture as possible components of nationwide digital networks. As one example among several, Western Tele-Communications, Inc. (WTCI) began as a video microwave carrier carrying signals for CATV systems owned by its parent company, Tele-Communications, Inc. in the Western half of the U.S. It is one of the largest non-telephone microwave common carriers in the country, with over 13,000 route miles of microwave links in service. WTCI plans to convert some of its bandwidth to digital links with data circuits up to 1.544 Mbps operated on a store-and-forward basis.

2.3 Value Added Networks (VAN's)

Value added networks are communication service companies which lease transmission facilities from common carriers or specialized carriers and resell communication services not available from the original carrier. One of the services to be offered is packet communication with service comparable to that in the ARPANET. FCC approval has been granted to Graphnet, Packet Communications, Inc. and Telenet. The introduction of VAN's services may be quite rapid since they do not undertake the construction of transmission lines. Tariffs have not been determined yet, but at least for packet switching networks, the rates will be independent of distance and proportional to traffic volume.

2.4 Satellite Communications

A number of companies have applied for FCC approval to sell commercial satellite services including Amersat, CML, and WU. Some are launching their own satellites. On April 13, 1974, Western Union's Westar 1 became the first U.S. domestic communications satellite launched into orbit. Competing will be systems operated by GTE, RCA, and ATT. Their ground stations and planned operations are part of the layman's day to day information as in the map of planned ground stations from the April 15, 1974, edition of the *New York Times* (see Figure 2.1).

Others are leasing channels on existing satellites. For example, American Satellite Corporation, formed in 1972 as a joint venture of Fairchild Industries and Western Union International has leased three transponder channels on Telesat-Canada's ANIK-2 satellite with plans for earth station locations in New York, Los Angeles, Chicago, and Dallas at data rates up to 60 Mbps.

Two general characteristics of satellite rates are insensitivity to distance with a strong volume discount for use of satellite bandwidth.

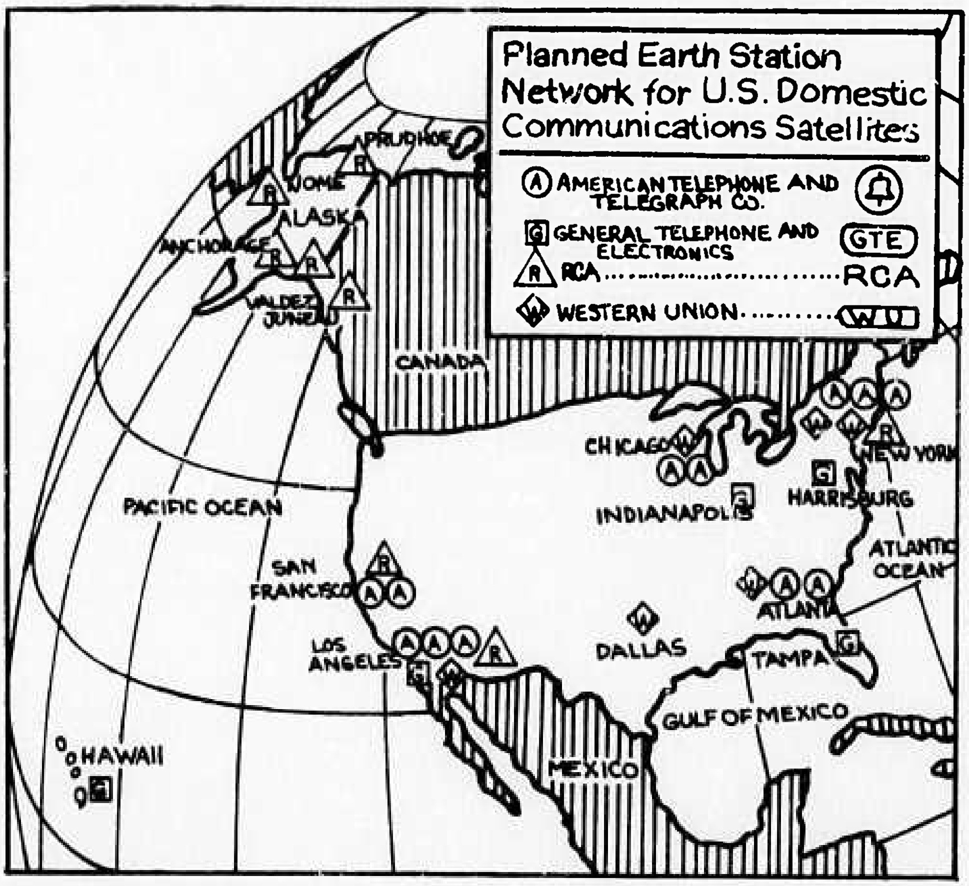


Figure 2.1: Satellite Map of U.S.

Chapter 3

NETWORK CONFIGURATION

The abundance and variety of services provided by the common carriers and specialized common carriers could mean low cost data transmission involving novel interconnection of facilities to match a broad range of user requirements, or it could mean failing networks and poorly spent dollars.

Let us first say that the main danger is not among the more classical problems of interconnecting facilities. There are, of course, challenging problems in interfacing terminal equipment from a range of manufacturers with a cascade of networking services. Hardware and software design for interfaces is required to assure proper synchronization, compatible message structure, electrical matching, and a host of other functions. These are important but, within the context of offered services, predictably solvable problems. The problem of servicing and maintaining a multilinked network with multiple owners is often a frustrating one with each vendor blaming a system outage on the others. But, this is also a solvable management problem.

The potential for difficulties is the complexity of configuring the best network to meet requirements and in designing the best method to locally access the network. If these tasks are improperly handled, the resulting networks may be more expensive than conventional approaches. Also, they might fail! Time delays in an interactive system might be so long that a programmer at a keyboard loses patience and abandons the system; times for bulk file transfers may become so long that the user opts for air express instead; users may receive so many busy signals that they lose business.

In both the national network and the local network, the user must consider the full gamut of constraints including throughput, delay [17, 12], reliability [14, 54, 37], and cost, among many others. These may affect not only network configuration, but also selection of mode of transmission and of the carrier.

For example, one of the properties of the ARPANET, and therefore, presumably of VAN's networks, is a high degree of reliability because of alternate routing and retransmission of lost packets. The varying requirements and performance profiles of different transmission schemes will dictate the use of sophisticated analysis and design aides. The characteristics of these aids are described below. The primary technical problem is the selection and configuration of the facilities to meet requirements, and to achieve savings. It is, at present, not feasible to develop a computer program which will optimize based on all commercial offerings. Nevertheless, it is clear to anyone who has tried to optimize even a simple problem involving only Telpak rate structures that network design must be computerized. Of necessity, the design programs must be characterized by an extreme degree of flexibility.

a. Programs must be modularized. Subnetwork designs, characterized by geographic locations, functions or traffic must correspond to modules of the program so that changes in tariffs and grade of service require only localized program modifications. Furthermore, new design methods can be achieved with rearrangement of modules.

b. Network design programs must be parametric. The designer must be able to evaluate requirements based on performance and cost. Traffic, delay, and throughput requirements can never be specified precisely *a priori,* but must be selected on the basis of tradeoffs of cost versus performance.

c. Programs must be capable of performance evaluation. From traffic statistics and measurements, the manager must be able to determine when a network must be upgraded.

d. Design programs must be interactive. It has been a common experience in many areas of operations research, combinatorics, and large scale network analysis and design that the human is an important element in the design loop.

e. Computational efficiency must be given paramount consideration. An improperly structured data base, a failure to consider sparsity, or an inefficient combinatorial subroutine can cause serious degradation in network performance and cost design.

Let us consider item a, modularized programs, as an example. In *Semiannual Report No. 2,* we gave a classification of general cost structures which includes most commercial tariffs and enables the development of efficient modules for each structure.

- Distance Dependent (DID) structures. The cost per channel between two points is a function only of the distance between the points; it is independent of the specific locations of the points or the bandwidth. Examples of DID structures are the AT&T type 8000 tariff and the Hi Lo density tariff, where both points are high density points.

- Location Dependent (LOD) structures. The cost per channel between two points depends on the location of the points, but not on the bandwidth. A typical example is the Hi-Lo density tariff if at least one point is a low density point. Another example is VAN's for non-network points.

- Volume Discount (VOD) structures. The cost of leasing additional bandwidth between two points decreases with the number of channels already leased. The cost also depends on the distance between the points, but not on their locations. Examples are the Telpak tariffs (see Figure 3.1), the DDS tariff for network points, the specialized carriers, satellite companies, and VAN networks. The case in which the line cost depends on terminal locations as well can often be reduced to the study of two separate LOD and VOD problems.

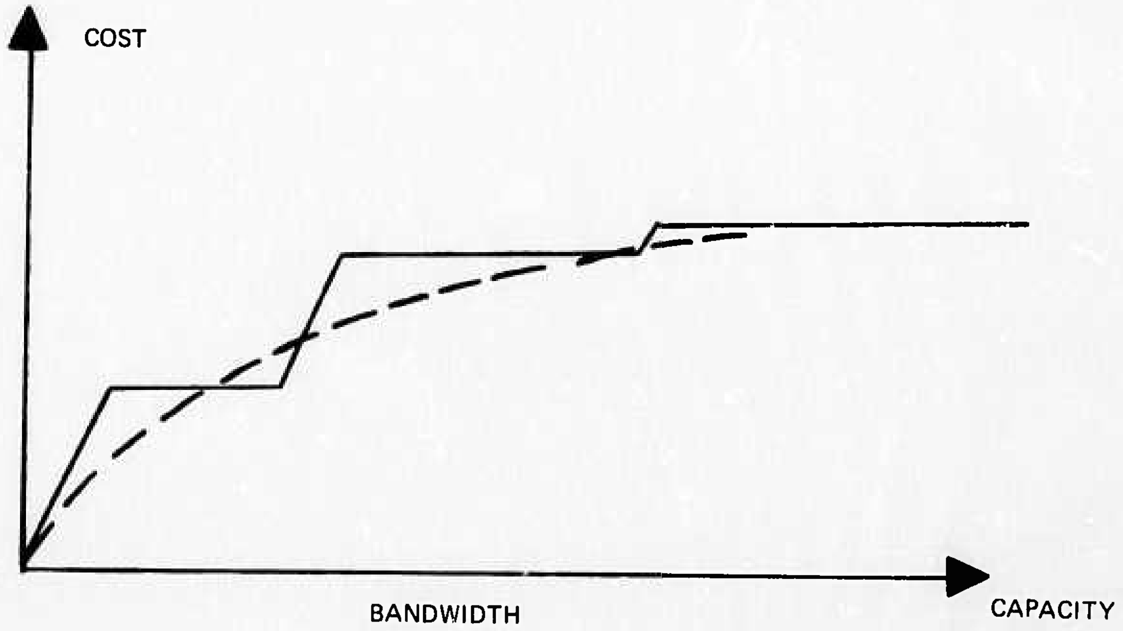


Figure 3.1: Telpak-Like Tariff

Each of the tariff structures is amenable to a different set of analytical and heuristic methods. For example, under VOD structures, network design tends to route traffic on links that provide the best economies of scale. Generally, cost structures with strong economies of scale lead to sparsely connected or tree topologies while structures with weak economies of scale lead to highly connected topologies. The network algorithms for VOD problems generally compute shortest routes according to appropriate link costs which vary with each iteration [59, 19, 13]. The specific algorithms and their efficiency depend upon the specific cost capacity function of the links [15]. Link functions which are neither convex nor concave and have large capacity jumps such as the Telpak-like tariff below, require very complex combinations of analytic and heuristic methods [47]. On the other hand, continuous concave cost-capacity functions can be handled exactly using mathematical programming techniques [13].

To illustrate how these modules are combined to solve a problem, we consider the design of a national network consisting of a terrestrial VAN and satellite links with private ground stations. Given the traffic requirements and tariffs, we wish to accommodate the traffic requirements with a specified grade of service at minimum cost. The network shows a strong economy of scale due to volume discount for satellite bandwidth and the high cost of ground stations. But, there is a complex tradeoff between terrestrial line cost, satellite bandwidth cost, and ground station cost.

A possible procedure for this problem begins with picking the number and location of ground stations (these may be varied for different designs). To connect non-network points to the VAN's terrestrial network we assign an LOD cost structure. We then satisfy node pair requirements along minimum cost routes relative to the LOD structure. From this assignment we determine national network requirements. Using a VOD structure, we then design the national network using the link costs for terrestrial and satellite links. Suboptimization of the terrestrial network is possible here. Once the nationwide network is obtained, the LOD cost structure is recomputed based on marginal costs and the procedure is iterated.

To solve such networks problems, which are quite large, will require special techniques that have not yet been developed. In the following chapters, we describe various network problems, ranging from cost and performance of large distributed networks to the control of traffic in a local broadcast packet network. The goal of the discussion will be to indicate the status of each research area and the open issues.

Chapter 4
LARGE SCALE DISTRIBUTED NETWORKS

4.1 Introduction

Performance of a distributed computer communication network is usually characterized by the parameters of cost, throughput, response time, and reliability. Analysis and design of large scale networks requires techniques substantially different from the ones used for smaller networks. Similarly, implementation in the actual network, of procedures such as routing and flow control are significantly impacted by network growth, and implementations suitable for a 50 node network may be totally inappropriate for a 500 node network required to perform the same functions.

Network design must be concerned with the properties of both the node's and the network's topological structure.

The outstanding design problem for large distributed networks is to specify their routing and topological structures. This specification must make full use of a wide variety of circuit options. Preliminary studies indicate that, initially, the most fruitful approaches will be based on the partitioning of the network into regions or, equivalently, constructing a large network by connecting a number of regional networks. In addition to reducing the computational complexity of the topological design problem, nodes may be clustered into regions for numerous reasons, such as:

- To partition status information for use in routing, flow control, and other decision processes within the operating network
- To determine regions of low-, medium-, and high-speed lines in hierarchical structures
- To find concentrator-multiplexer locations

To send a message in a distributed network constructed by connecting together a set of regional networks, a node might specify both the destination region and the destination node in that region. No detailed implementation of a large network has yet been specified, but an early study of their properties indicated that factors such as cost, throughput, delay, and reliability are similar to those of the present ARPA Network, if the ARPA technology is used [10].

One of the first questions that arises in the decomposition problem is how large to make the regions and how many regions to create. In a multilooped network, each loop would define a region. Once the number of regions has been determined, nodes must be assigned to each and interface nodes selected. This problem is extremely difficult, and little is presently known about this "clustering" problem.

Before any substantial progress can be made in applying existing clustering techniques to large computer networks, an appropriate "distance" or "nearness" measure must be selected, probably on the basis of intuition and experiment.

For the design of large computer networks, clustering requires the assignment of distance measures to take into account cost, capacity, traffic, delay, reliability, and routing. Almost no general theoretical results are presently known for this problem.

In this section, we report the results of a sequence of experimental network designs aimed at investigating the economic and performance tradeoffs of distributed computer communication networks as a function of size. Three sequences of designs were performed: 20-100 nodes, 200 nodes and 1000 nodes. The first series of networks were thoroughly optimized using the optimization techniques discussed [11]. The 200 node networks were partially optimized (within the limitations of a small finite computer time budget), while the 1000 node network designs represent workable network designs whose structure was chosen for both buildability and mathematical tractability.

4.2 The Network Model

The network model chosen for the study was the ARPANET [18]. This system, which employs packet switching is likely to be the prototype for most future distributed computer communication networks.

4.2.1 Message Handling

The message handling tasks at each node in the network are performed by Interface Message Processor (IMP) located at each computer center. The centers are interconnected through the IMP's by fully duplex communication lines. When a message is ready for transmission, it is broken up into a set of packets, each with appropriate header information. Each packet independently makes its way through the network to its destination. When a packet is transmitted between any pair of nodes, the transmission IMP must receive a positive acknowledgement from the receiving IMP within a given interval of time. If this acknowledgement is not received, the packet will be retransmitted, either over the same or a different channel depending on the network routing doctrine being employed.

4.2.2 Design Goals

A design goal of the system is to achieve a response time of less than 0.5 seconds for short messages. The final network must also be reliable, and it must be able to accommodate variations in traffic flow without significant degradation in performance. In order to achieve a reasonable level of reliability, the network must be designed so that *at least* two nodes or links must fail before it becomes disconnected. For small networks, this design goal ensures adequate reliability but for larger networks, additional provisions must be made [55]. Thus, the 1000 node design was specially handled with respect to its reliability considerations.

4.2.3 Routing Procedure

For the network designs containing 200 or fewer nodes, the *Minimum Node Routing* procedure described in [13] was employed. However, for the 1000 node network design, the *Cut Saturation Technique* described in [4] was used. This improved procedure was used in an attempt to compensate for the lack of optimization of the 1000 node network topology.

4.2.4 Cost Structure

Cost structures for lines and IMP's are shown in Tables 4.1 and 4.2. Note that the line costs are those available to the U.S. Government under the Telpak tariff.

Table 4.1: (all lines FDX)

Capacity (Kbps)	Data Set Cost/Month	Line Cost Per Mile/Month
9.6	$ 493	$ 0.42
19.2	$ 850	$ 2.50
50.0	$ 850	$ 5.00
230.4	$1300	$30.00
1544.0	$2000	$75.00

Table 4.2: Message Processor Cost

Description	Purchase Cost	Cost/Year*
DDP-316 IMP (Max throughput = 600 Kbps)	$ 50,000	$15,000
DDP-516 IMP (Max throughput = 800 Kbps)	$ 70,000	$21,000
DDP-316 TIP (Max throughput < 600 Kbps)	$100,000	$30,000
HSMIMP (Max throughput = 6,000 Kbps)	$250,000	$75,000

*Yearly cost is assumed 30% of purchase cost

4.2.5 Node Locations

Node Locations strongly influence network efficiency. For example, at one traffic level a 20-node network may be more efficient than the same network with two additional nodes while it may be less efficient at other traffic levels. Consequently, a systematic and hopefully "unbiased" procedure is required to generate realistic systems with differing numbers of nodes.

Many location algorithms are possible. For example, nodes could be located on the basis of any of the following requirements: industrial concentration; distribution of military bases; distribution of universities; population.

Since both industry and universities tend to be located at population centers, distributions produced by these factors are positively correlated. On the other hand, military bases are often located at a distance from population centers, and hence such nodal distributions would have a negative correlation with the others.

For the present study, nodes were located on the basis of population. Within each metropolitan area, IMP's were assigned, 20 miles apart, in proportion to the population of the area. As examples, the 20 node design connected 10 metropolitan areas, the 100 node design connected 40 metropolitan areas, the 200 node design connected 62 metropolitan areas, and the 1000 node design connected 238 metropolitan areas.

4.2.6 Traffic Assignment

A fundamental problem in all network design is the estimation of the traffic the network must accommodate. For some problems, accurate estimates of user requirements are known. However, complete studies are not yet available to predict the flow requirements in networks of the type being considered here. A number of basic questions are yet to be resolved. For example, it may be reasonable to assume that the flow out of a node will be proportional to the population assigned to that node. However, will the flow between two nodes be affected by the distance between these nodes? If so, how will the cost-throughput characteristics of the network be affected?

In order to investigate the effect of different traffic distributions on network economy, a sequence of experiments described in [10], was conducted in which traffic patterns were varied as a function of distance and low cost networks for these patterns generated. These experiments indicated that networks with comparable costs could be designed for widely varying traffic requirements. Hence, in the design experiments used to generate cost-performance tradeoffs, equal traffic requirements between all nodes was assumed.

4.2.7 Summary of System Parameters and Characteristics

The following is a summary of the factors utilized in the network design.

a. The system contains message processors located in the largest cities of the Continental United States. The number of message processors in each metropolitan area is proportional to the population of the town.

b. Required traffic between message processors is assumed uniform for all node pairs. Traffic levels in the range from 3 to 20 Kbps/node are considered.

c. Messages are assumed to have the same structure and formats as in the present ARPANET configuration. Message delay is evaluated for single packet messages.

d. The nominal traffic level is set at 80% of the saturation level in order to maintain within acceptable limits the queue size of packets awaiting transmission on each channel.

e. The link failure rate is assumed equal to 0.02. The node failure rate is assumed equal to 0.02 for IMP and TIP processors, and .0004 for redundant configurations (IMP or TIP plus backup, or redundant high speed modular IMP configurations).

f. The high throughput presented by a 1000 node network requires very high channel and message processor rates. Therefore, in the design, two high speed hardware options—the 1544 Kbps data channel and the HSMIMP (High Speed Modular IMP)—have been considered in addition to the options already available. Such high rate options are presently under development and will soon be operational offerings.

4.3 Regional Partitions and Hierarchical Network Structure

The network design is based on a regional decomposition principle. The system is divided into regions. Nodes are then uniquely assigned to each region. These nodes are classified as either exchange nodes or local nodes. There may be any number of regions, and the choice of local and exchange nodes is made by the designer. The distinction between nodes stems from restrictions on allowable connections.

a. Within a given region any connection is possible.

b. Connections between regions are only allowed between exchange nodes.

c. Any connection between exchange nodes is possible.

For the designs containing no more than 200 nodes, a two-level regionalization is used. For these networks, the network connecting the exchange nodes is called the national network while the networks connecting local nodes are called regional networks. Therefore, a flow from a local node in region I to a local node in region J must be routed first through the regional network to an exchange node in region I, then through the national network composed of exchange nodes to an exchange node in region J and finally through another regional network to the destination node in region J. Thus, in general, a hierarchical network is designed. If desired, structural constraints can be eliminated by declaring every node to be an exchange node. In this case, there are N(N-1)/2 possible links where N is the number of nodes in the network. If N is large, the computational time required for optimization may be prohibitive, and therefore, decomposition is essential. For the sizes of the networks being considered, the decomposition approach produces computation time savings ranging from a factor of ten to factors of more than 400.

The determination of the optimal topology in a 1000 node hierarchical network is a very complex problem since it requires the solution of a large number of subproblems, all related to one another. For example, one must optimally determine:

- The number of hierarchical levels
- The node partitions
- The topology within each partition
- The connections between networks in different levels of the hierarchy

Because of the complexity of design optimization, only feasible, reasonably low cost designs were considered in the 1000 node study. A feasible design in fact is sufficient for the determination of cost, throughput, delay and reliability trends with respect to network size, and for a comparison between hierarchical and non-hierarchical structures.

The hierarchical structure (see Figures 4.1, 4.2 and 4.3) here considered consists of *three* hierarchy levels: one 10 node national network, ten 10 node regional networks, and one hundred 10 node local networks. Each local network is considered as one "node" of the higher level regional net, and similarly, each regional net is one node of the national net. Various ways of connecting lower to higher level networks can be considered. In the cost/throughput study, we assume for simplicity that each subnetwork communicates with the higher level network only through one "exchange" node. In the reliability study, however, two and three exchange node configurations are also considered.

4.4 Reliability

Reliability constraints play increasingly more important roles with growth of the network [55]. For example, in the early stages of the ARPANET growth, adequate reliability was

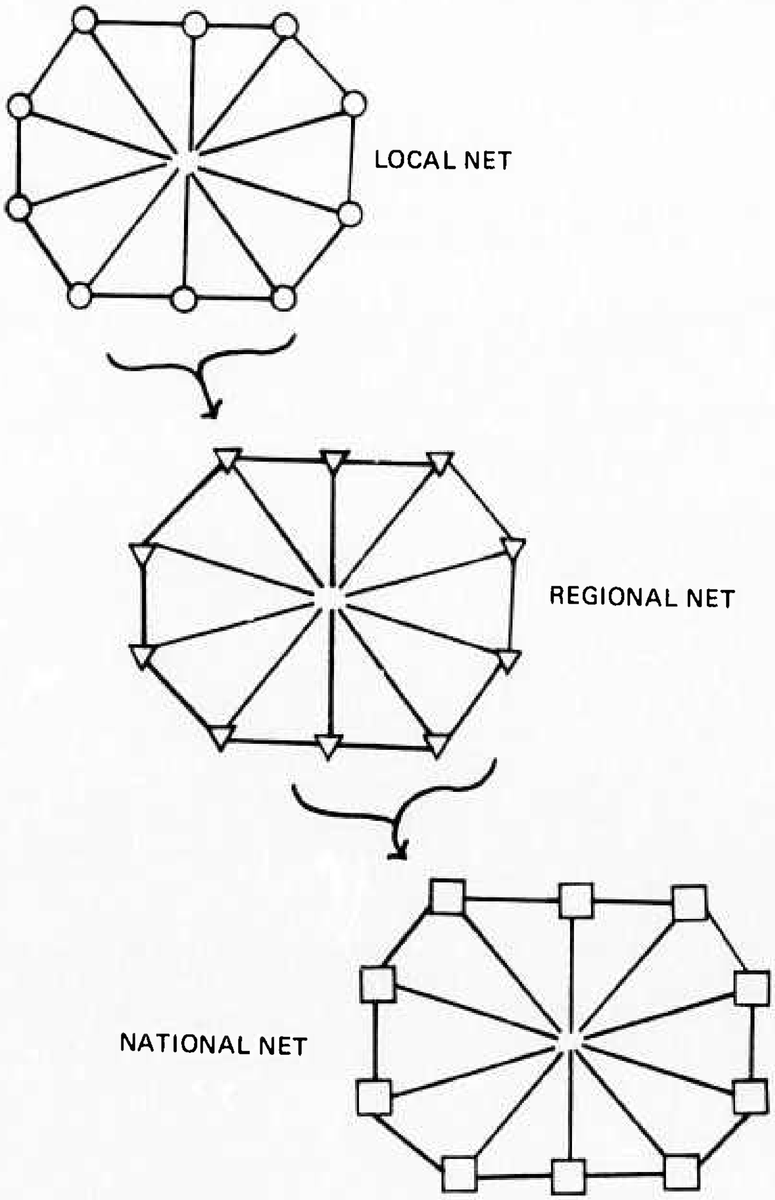


Figure 4.1: A 1000 Node Network Composed of 10 Ten-Node Regional Nets Each Containing 10 Ten-Node Local Nets

achieved by the provision of two node disjoint paths between all pairs of nodes. However, even at its present size (approximately 40 nodes) this is no longer a sufficient guarantee of adequate network reliability.

Network reliability analysis is concerned with the dependence of the reliability of the network on the reliability of its nodes and links. Element reliability is easily defined as, for example, the fraction of time the element is operable, or as by the mean time between failures and expected repair time. The proper measure of network reliability is not as

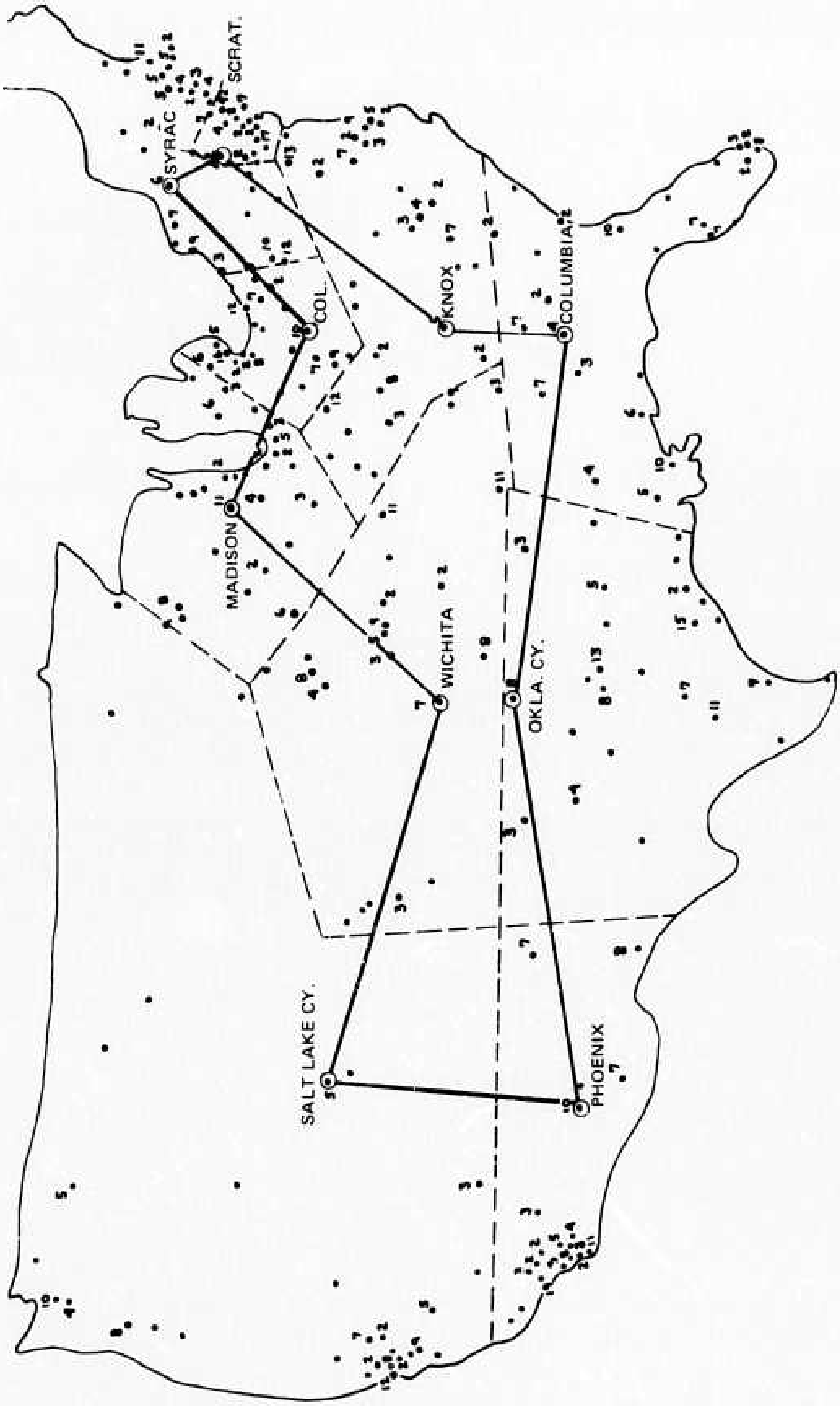


Figure 4.2: National Network and Regional Partitions

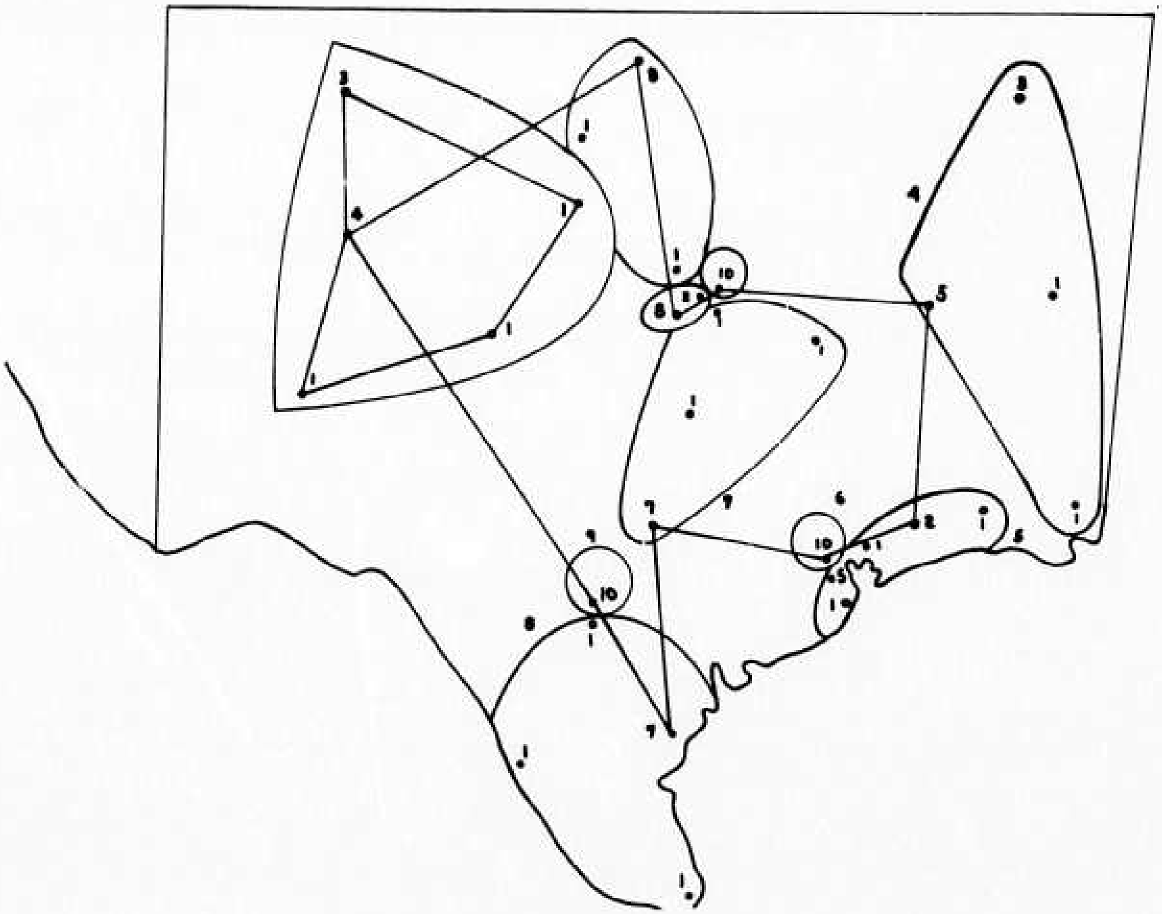
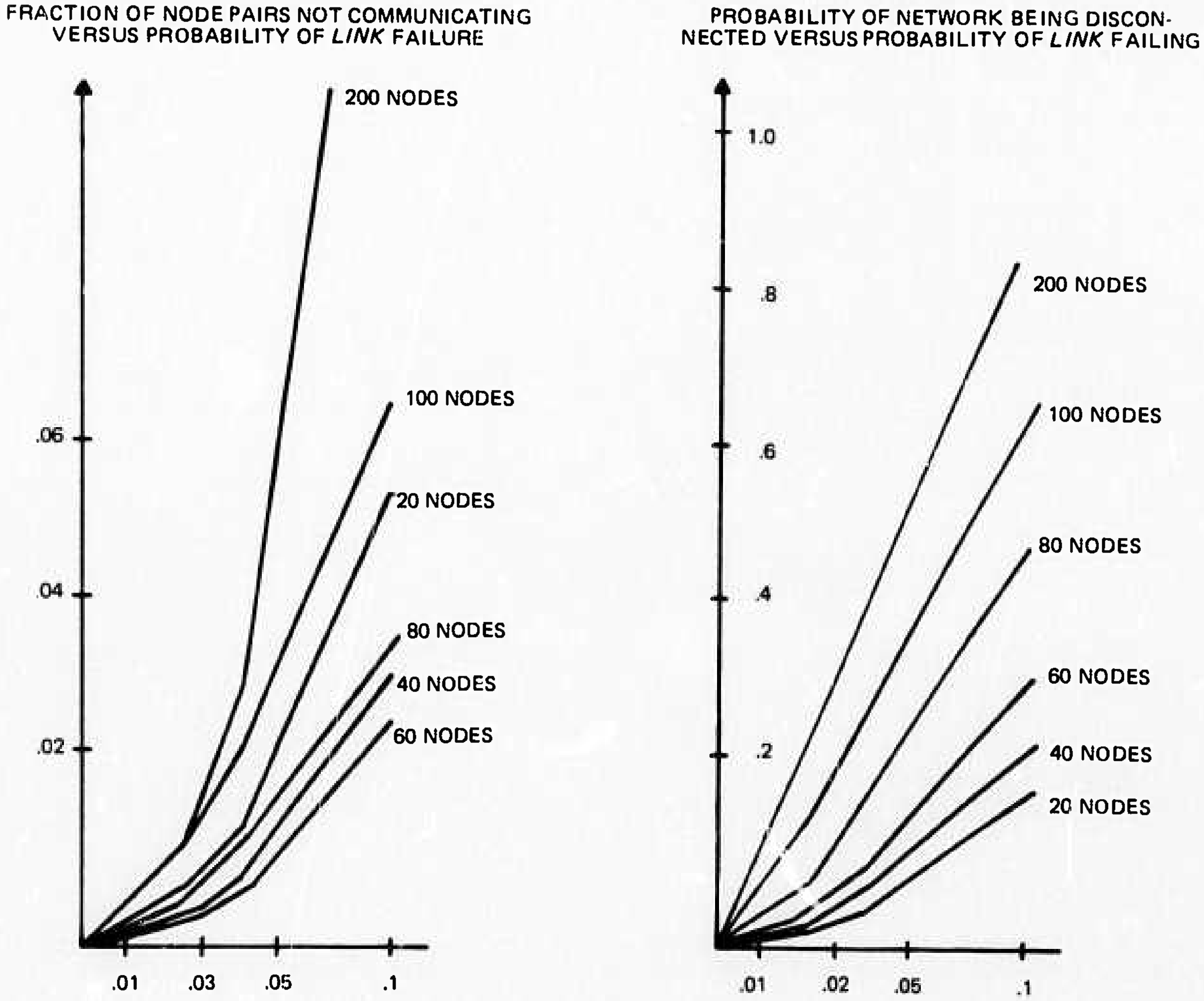


Figure 4.3: Regional Network; Local Partitions; Local Network

clear and simple. Three possible measures are: the number of elements which must be removed to disconnect the network, the probability that the network will be disconnected and the expected fraction of node pairs which can communicate through the network. Many other measures can and have been suggested.

Figure 4.4 shows the results of reliability analyses of a set of 20-100, and 200 node networks designed to meet throughput requirements of approximately 8 Kbps/node under the assumptions that nodes are perfectly reliable. These networks were designed with a "two connectivity" constraint and were optimized to provide the required throughput at least cost. As is evident, the larger networks are significantly less reliable than the smaller networks. Thus, extension of the same design principle to the 1000 node design would be likely to lead to a low reliability system. Therefore, the 1000 node network is considered under varying conditions of backup and structure. These changes are made at the *local* level. The regional and national levels are always three connected.

To evaluate the reliability of the hierarchical 1000 node network, we make the assumption that two nodes in the same subnetwork can communicate with each other only through


FRACTION OF NODE PAIRS NOT COMMUNICATING VERSUS PROBABILITY OF *LINK* FAILURE

PROBABILITY OF NETWORK BEING DISCONNECTED VERSUS PROBABILITY OF *LINK* FAILING

Figure 4.4: Network Reliability As A Function of Number of Nodes

paths entirely contained in the subnetwork. Therefore, two nodes of the same subnetwork can be disconnected even if there is a connection path through the higher level network. This assumption is very realistic because, in a hierarchical routing implementation, the capability of sending local or regional traffic along paths external to the corresponding local or regional net, can be achieved only with considerable increase in complexity and overhead of the routing algorithm.

With the above assumption, the probability P_{nt} of the total network being disconnected is given by:

$$\begin{aligned} 1 - P_{nt} = {} & (1-P_{n\ell})^{100} \times (1-P_{nr})^{10} \\ & \times (1-P_{nn}) \times (1-P_{ex})^{110} \end{aligned}$$

where $P_{n\ell}$ = probability of local net disconnected

P_{nr} = probability of regional net disconnected

P_{nn} = probability of national net disconnected

P_{ex} = probability of exchange node (or nodes) failure, which isolates the corresponding subnetwork

To evaluate F_{nt}, the fraction of disconnected node pairs, we make the simplifying (and conservative) assumption that whenever a subnetwork becomes disconnected, only one half of the nodes in the subnetwork can communicate, on the average, with the exchange node (or nodes). With such an assumption, if we let N be the number of nodes in the local net (in our case N = 10) and a_ℓ, a_r, a_n the number of noncommunicating node pairs resulting from the disconnection of a local, regional or national network respectively, we have:

$$a_\ell = \frac{N}{2}\left(N^3 - \frac{N}{2}\right) P_{n\ell} + N(N^3 - N)\, P_{ex}$$

$$a_r = \frac{N^2}{2}\left(N^3 - \frac{N^2}{2}\right) P_{nr} + N^2 (N^3 - N^2)\, P_{ex}$$

$$a_n = \frac{N^3}{4}\, P_{nn}$$

If we make the assumption that the above contributions are statistically independent of one another, then we can sum them up and obtain the following expression for F_{nt}:

$$F_{nt} = 2P_n(1-P_n) + \frac{2}{N^6}\left\{ a_\ell N^2 + a_r N + a_n \right\}$$

$$\cong 2P_n(1-P_n) + P_{n\ell} + P_{nr} + \frac{P_{nn}}{2} + 4P_{ex}$$

where P_n is the node failure rate, and $2P_n(1-P_n)$ is the fraction of disconnected node pairs resulting from source and/or destination failures.

To evaluate P_{nt} and F_{nt} as from expressions (1) and (5), we need to know the network disconnection probability P_{nc} for the basic, 3-connected 10 node structure. The following results were obtained using the reliability analysis programs described in [17, 12]:

$$P_{link} = .02;\ P_{node} = .02 \rightarrow P_{nc} = 7.10^{-4}$$

$$P_{link} = .02;\ P_{node} << .02 \rightarrow P_{nc} = 8.10^{-5}$$

To test the effect of various topological and back-up conditions, P_{nt} and F_{nt} are evaluated for a variety of network configurations which differ in:

1. Number of exchange nodes
2. Redundancy in the exchange nodes
3. Connectivity of the local network

Figure 4.5 illustrates the various configurations and Table 4.3 summarizes the results.

4.5 Cost-Throughput Trends

An important point which must be emphasized about the results to follow is that these results present a conservative picture of the relationship between cost and throughput. There are two major reasons for this.

Each point represents a *feasible* network obtained by either the computer network design program or by specification of the 1000 node network topology. Thus, to generate the specified throughput, no greater cost would be involved. However, because of the number of points needed to generate adequate curves, it is prohibitively costly to devote a large amount of computer time to optimize completely each design point. Therefore, if a specific throughput were to be required, a more intensive optimization would be warranted and a lower cost design would be probable.

In each design, only hardware and line options available now or in the near future have been allowed. Other developments could substantially reduce costs. For example, in [18] we demonstrated the economics created by using a 108 Kbps data set. Although this data set was once tested by AT&T, it is not a commercial offering. However, the costs involved in building a large computer network could justify the independent development of such a data set.

To illustrate the tradeoffs that occur, we first examine the 1000 node network.

For the 1000 node topology, total cost and delay for a given throughput can be obtained by analyzing 111 subnets and properly combining the results. Such an extensive analysis is too cumbersome in our case since we are interested in using throughput as a parameter. Therefore, to simplify the computation, only the costs of the national network shown in Figure 4.2 and the regional and local nets shown in Figure 4.3 were thoroughly computed, and the results interpreted as representative for all other regional and local nets. Notice that the above approach generates imprecision in the total cost, but provides the correct answers for both delay and throughput.

Figure 4.6 shows cost, throughput and delay of the *national* network for three different capacity allocations. The lowest cost configuration uses all 230.4 Kbps channel capacities. The intermediate configuration uses 1.544 Mbps channels for the outer loop, and 230.4 Kbps for the cross links. The highest cost configuration uses all 1.544 Mbps channels.

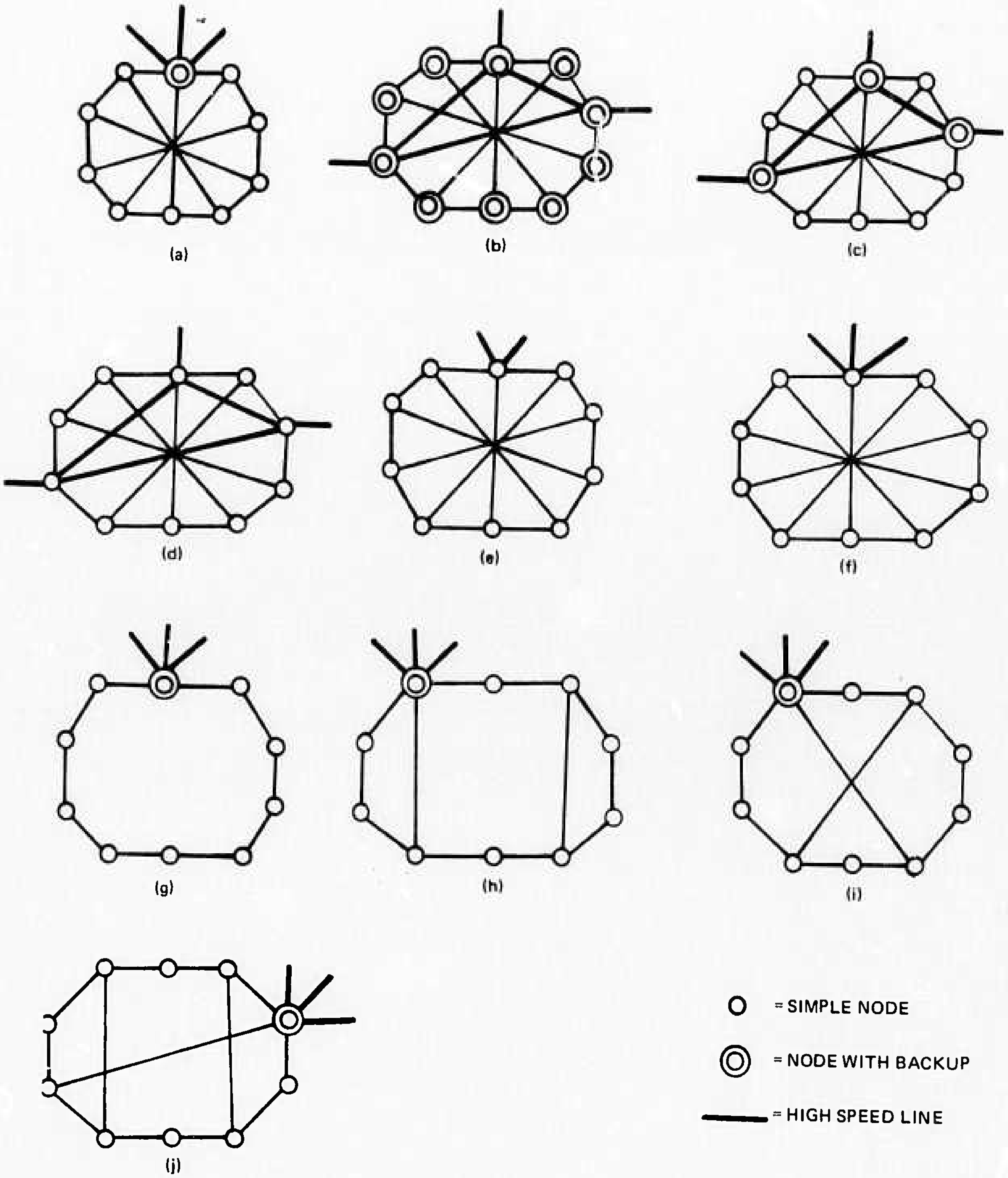


Figure 4.5: Various Local Network Configurations

Table 4.3: Failure Probabilities for Differing Networks of Figure 4.5

Network	**Number of Exchange Nodes**	**Location of Backup**	**Disconnection Probability**	**Average Fraction of Disconnected Node Pairs**
a	1	all exchanges	.011	.042
b	3	all nodes	.009	.001
c	3	all exchanges	.078	.028
d	3	none	.095	.044
e	2	none	.128	.044
f	1	none	.9	.12
g	1	all exchanges	.999	.114
h	1	all exchanges	.93	.066
i	1	all exchanges	.84	.058
j	1	all exchanges	.65	.05

The throughput, expressed in Kbps/node, refers to the local nodes; therefore, the throughput of each of the 10 "supernodes" in the national net is approximately 100 times higher. The cost in Figure 4.6 reflects line and data set costs. The additional message processor cost is now evaluated, assuming that each node has redundant processors:

a. Lower cost net:
20 x DDP-316 IMPs, cost = .3M$/Year

b. Intermediate cost net:
20 x HSMIMPs, cost = 1.5M$/year

c. Higher cost net:
20 x HSMIMPs, cost = 1.5M$/year

Figure 4.7 shows the results for the *regional* net. The lowest cost solution uses mostly 50 Kbps channels; the highest cost solution includes several 230.4 Kbps and 1.544 Mbps channels. The throughput refers to local nodes. Assuming that each node has redundant processors, the message processor cost is given below:

a. Lower cost net:
18 x DDP-316 IMPs, cost = 270K$/year

b. Intermediate cost net:
16 x DDP-316 IMPs, cost = 240K$/year
2 x HSMIMPs, cost = 150K$/year
Total cost = 390K$/year

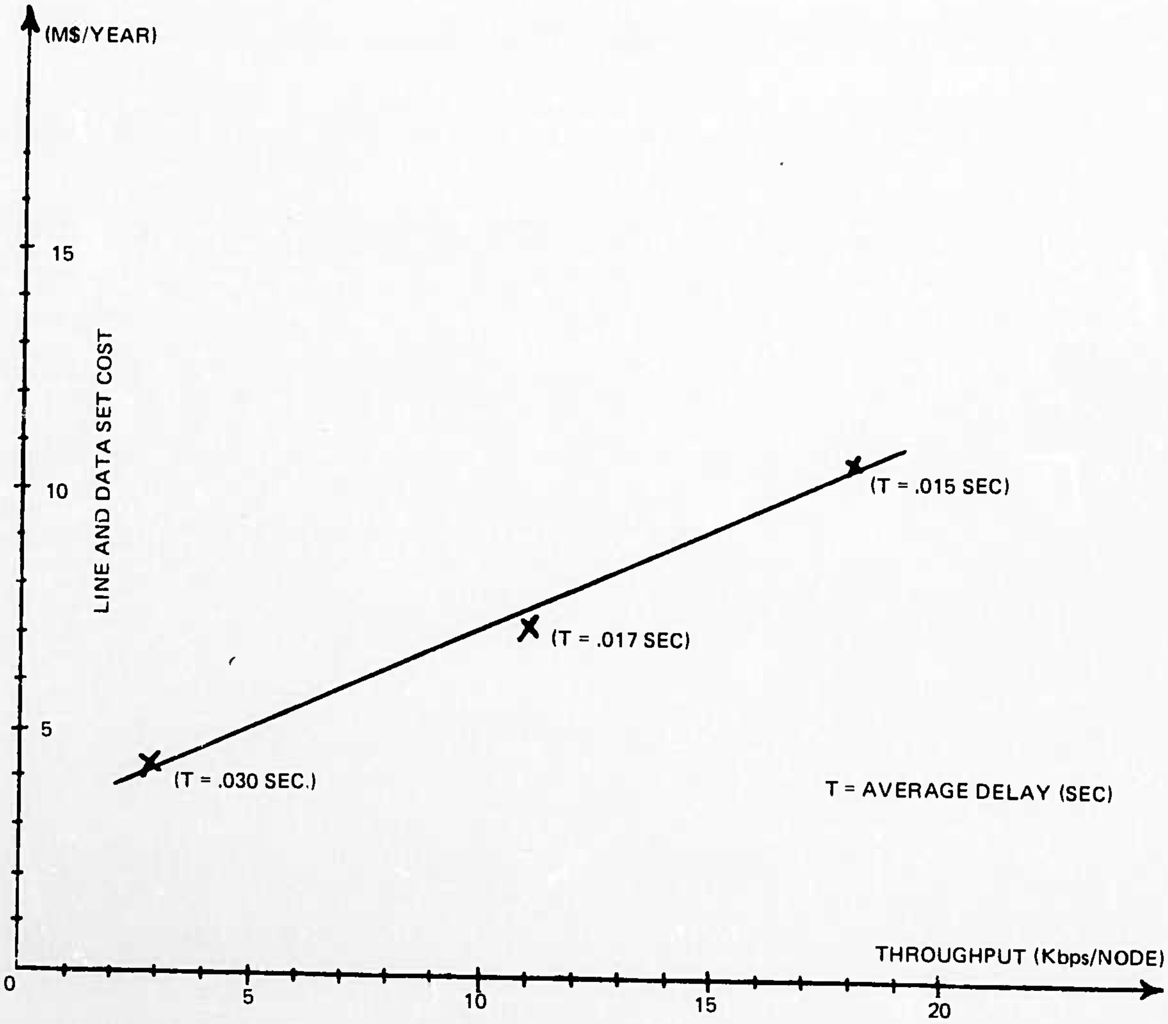


Figure 4.6: National Network

c. Highest cost net:
12 x DDP-316 IMPs, cost = 180K$/year
6 x HSMIMPs, cost = 450K$/year
Total cost = 630K$/year

Figure 4.8 shows the results for the *local* net. Both 3-connected and loop configurations were analyzed. Various capacity assignments, leading to different solutions, were considered. Average delay T in the local nets is much higher than in the national and regional nets, because of the extensive use of 9.6 Kbps and 19.2 Kbps channels, especially in the low cost, low throughput configurations. The delay can be reduced by reducing the traffic load, as shown in Figure 4.8. The local network does not require, in general, redundant processors; the message processor cost is given by:

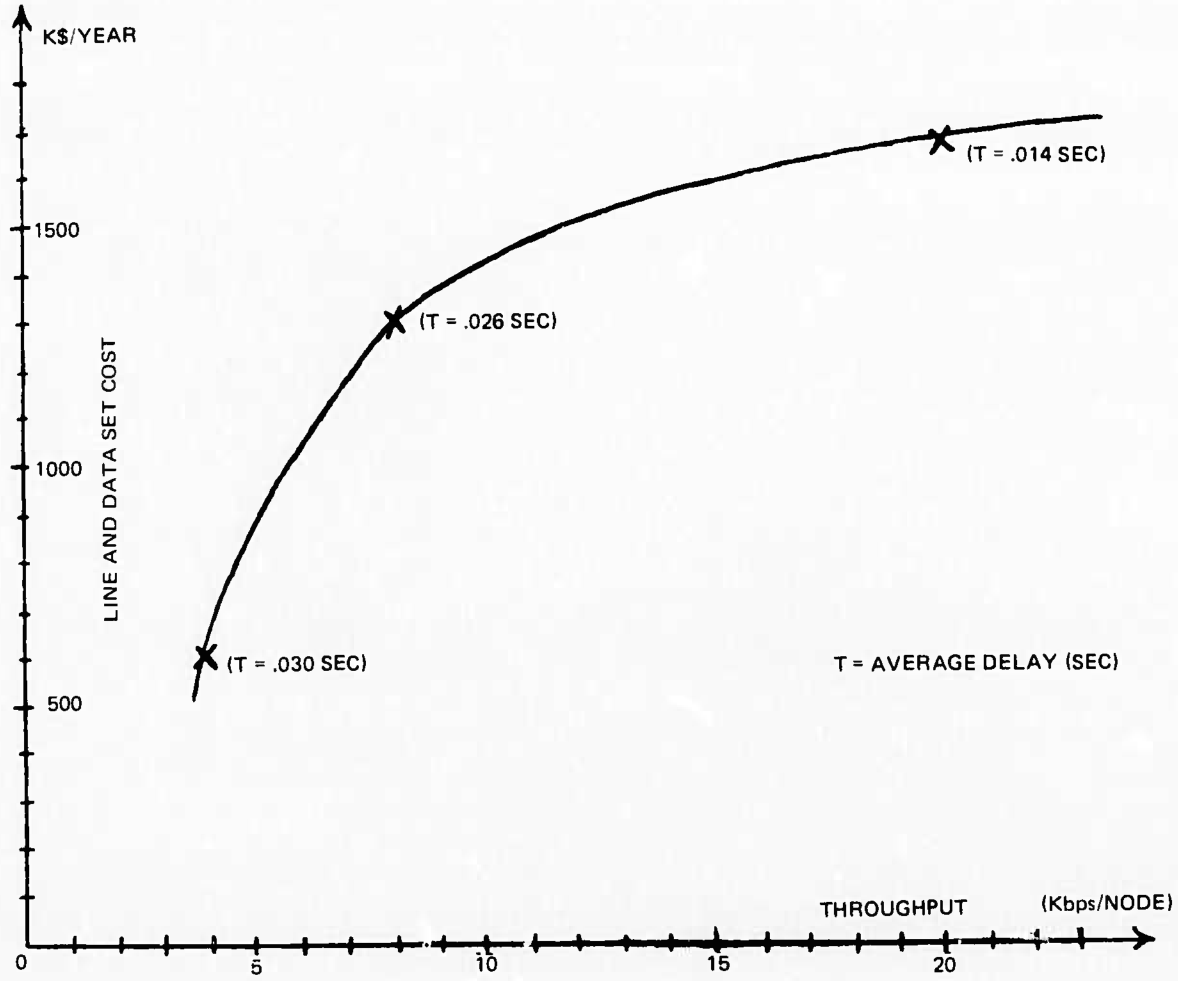


Figure 4.7: Regional Network (Texas)

Local network:
9 x DDP-316 IMPs, cost = 135K$/year

The results for the *global* net are obtained as follows:

a. For each throughput level, the lowest cost national, regional and local solutions that can accommodate such a throughput are selected.

b. The total cost D_t is given by:

$$D_t = D_n + 10D_r + 100 \times D_\ell$$

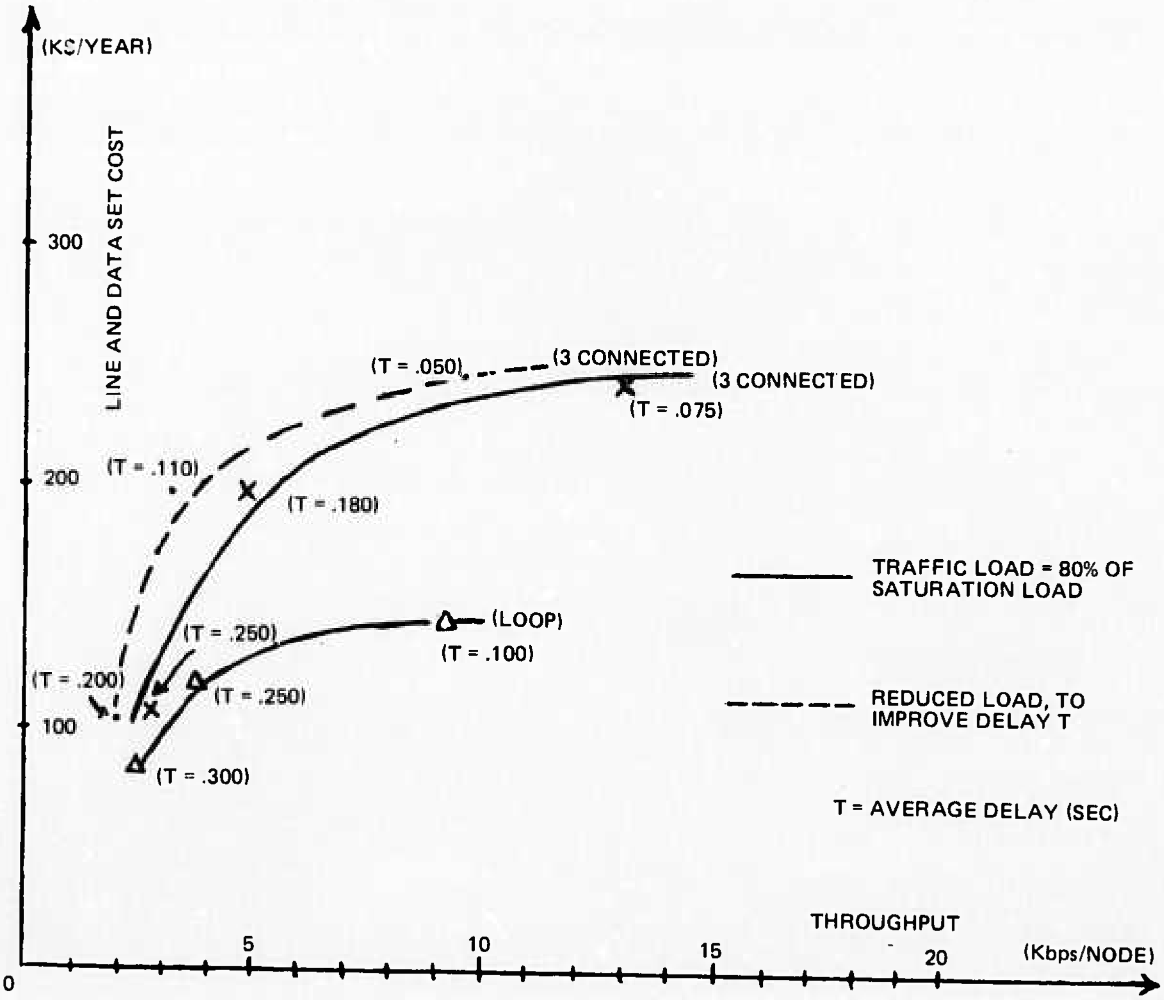


Figure 4.8: Local Network (Section of Texas)

where D_n = national net cost

D_r = regional net cost

D_ℓ = local net cost

c. The total average delay T_t suffered by a packet traveling from source to destination is typically given by:

$$T_t = T_n + 2T_r + 2T_\ell$$

where T_n = national network delay

T_r = regional network delay

T_ℓ = local network delay

Figure 4.9 shows channel cost and delay of the 1000 node net for both 3-connected and loop local net configurations while Figure 4.10 shows total communication cost.

The diagram in Figure 4.11 displays line and modem cost per node versus network size, for two different values of throughput. The shadowed area represents the cost of networks with local connectivity ranging from 2 to 3 for the 1000 node design. The cost for N=1000 seems to be slightly higher than the trend displayed for N up to 200. It should be remembered, however, that:

a. The cost estimate for N=1000 is not precise

b. The cost for N ⩽ 200 was minimized using link exchange procedures [18], while the cost for N=1000 is the cost of feasible but unoptimized network

Thus, we can expect that optimized network cost for N=1000 would be lower and could follow closely the trend established for N up to 200. The upper bound for the N=1000 curve represents 3-connected local topology as well as 3-connected regional and national

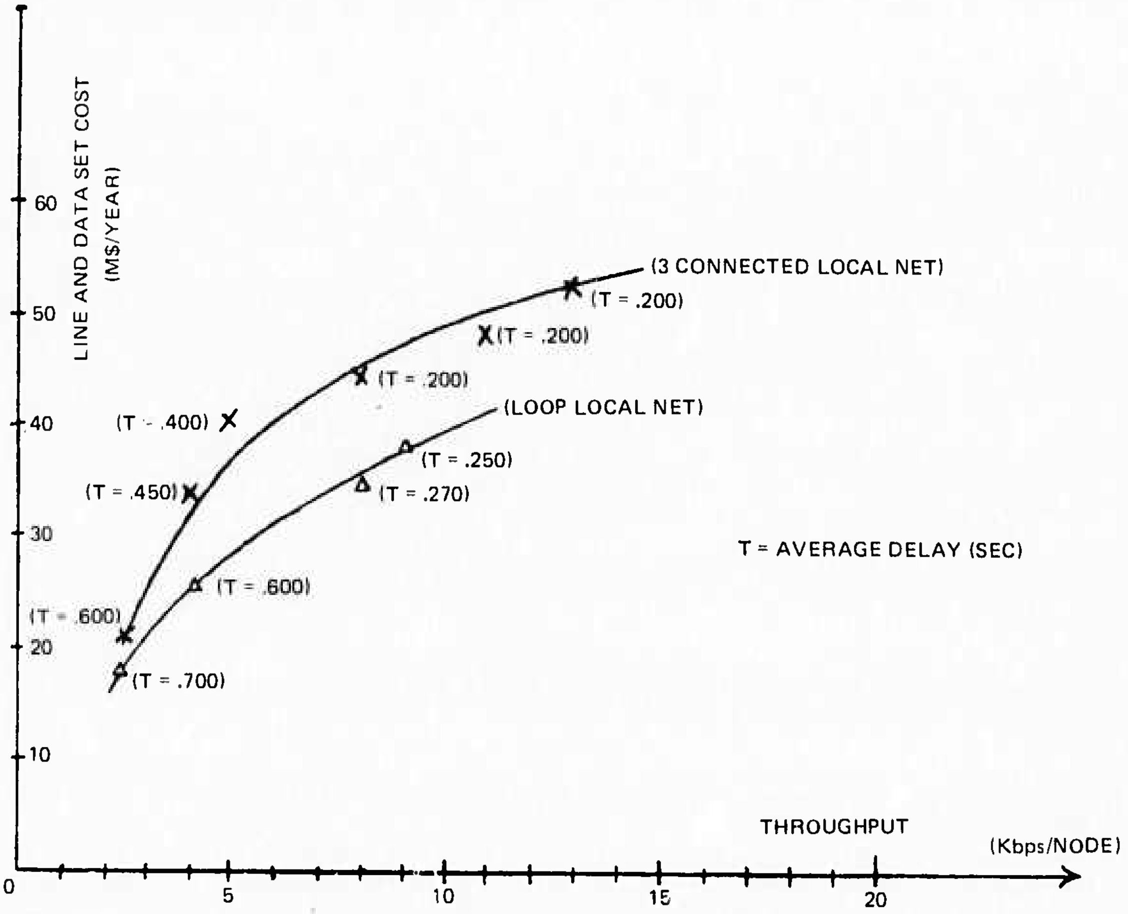


Figure 4.9: Global 1000 Node Network

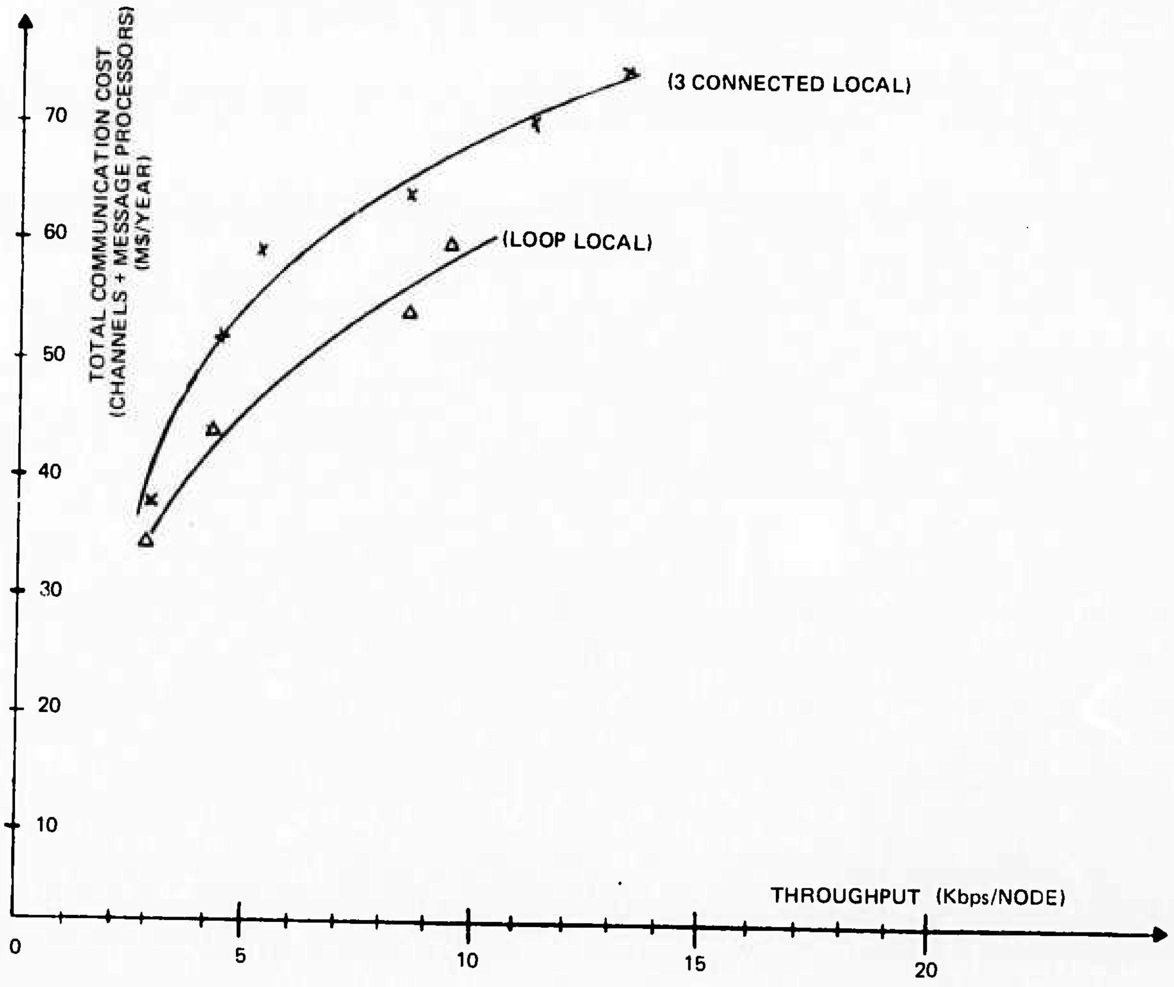


Figure 4.10: Global Node Network

topologies. Since this network can have very high reliability *without* local backup of nodes, we need never spend more than this bound for the 1000 node network. The region between the 2-connected and 3-connected points represents costs that might be achieved using optimization on each local network (with reliability as a constraint). Further economies would require the restructuring of the overall network hierarchy and partitioning achemes. We have noted that this optimization problem is extremely difficult.

4.6 Implications for Future Research

The results summarized here establish the feasibility, in terms of design techniques, cost, delay and reliability, of very large packet switched networks. Future steps in the research will be:

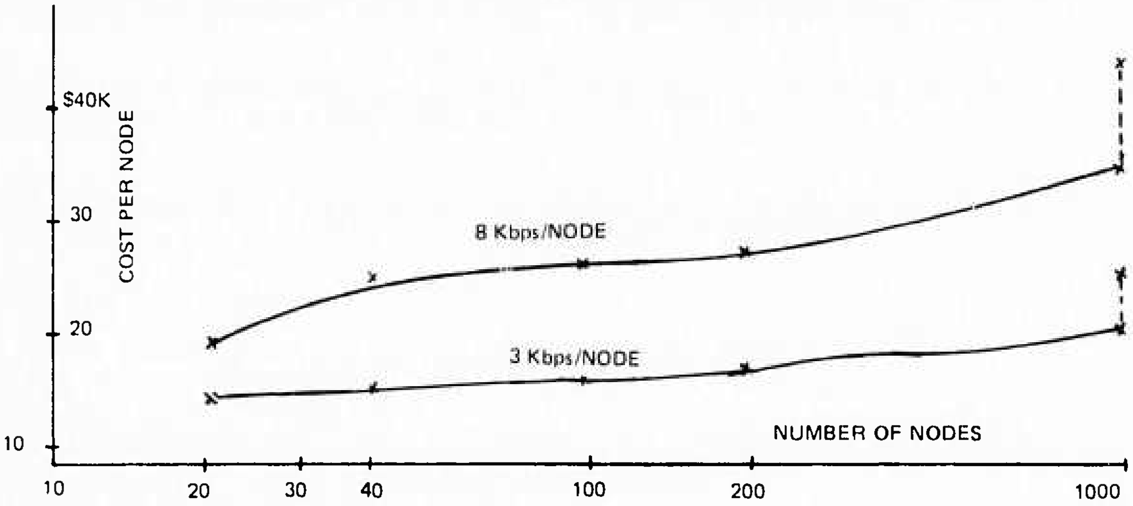


Figure 4.11: Network Size vs. Costs For Two Traffic Requirements

- Optimization of network design
- Performance evaluation
- Routing and flow control
- Use of different communication techniques at different hierarchical levels.

Some of the open areas are elaborated in the following.

4.6.1 Optimal Design

The design of hierarchical network requires: selection of number of hierarchical levels and of number of "nodes" for each level; determination of node partitions (on the basis of geographical distance, node requirements, etc.); separate minimum cost design for each partition and hierarchical level; combination of the partial designs into the global design. Low cost designs can be obtained with an iterative procedure, in which an initial configuration is successively improved, by properly modifying node partitions, local topologies, interconnections between different hierarchical levels, etc., until no more improvement is possible. One of the bottlenecks of the procedures is the local minimum cost network design, which must satisfy both traffic and reliability constraints. Present optimization techniques are inadequate and faster, and more efficient methods must be developed.

4.6.2 Performance Evaluation

The exact evaluation of throughput, delay and reliability for a 1000 node network requires a prohibitive computation time and memory space if performed with the present methods.

This is not so critical for the network design since approximate expressions of throughput, delay and reliability are probably sufficient. For the final configuration, however, a more precise performance evaluation is desirable, and therefore, new and efficient methods of large network analysis must be developed.

4.6.3 Routing and Flow Control

The traffic within each subnetwork can be routed and controlled with the present ARPANET techniques. However, proper modifications must be introduced to direct the traffic to external destinations. In addition, a multilevel flow control procedure could be implemented to obtain more efficient control of the traffic load in each hierarchical level.

4.6.4 Hybrid Communication Implementations

The hierarchical structure allows within certain limits, the use of different system implementations at different hierarchical levels. This feature can be exploited to obtain a more economical and efficient system. Possible configurations might include: broadcast "ALOHA type" radio techniques at the local level; packet switching techniques at the regional level; satellite broadcast techniques at the national level. It is of interest to investigate feasibility and economics of such hybrid implementations.

Chapter 5

RANDOM ACCESS PACKET TRANSMISSION

The most expensive part of a hierarchical communication system is often the lowest level since it contains the most elements. Local distribution and collection of data is usually characterized by very low utilization of facilities. Random access packet techniques provide promising schemes for satellite cable, and local radio communication. Hence, a basic understanding of these mechanisms is useful to the following chapters.

The exposition is aided here by one of those happy situations where the precise derivation is its own most intuitive explanation. We therefore present the derivation of the capacity of a random accessed channel, as originally devised for the ALOHA radio system [1]. Random access packet transmission is similar to time division multiple access with the following difference:

- Messages are broken into fixed size packets.
- Each transmitter can initiate a message at any time without reserving slots or requesting channel access.

Let τ be the duration of a packet and assume there are k active users. The overlap of two packets from different stations is illustrated in Figure 5.1. Assume that when an overlap occurs neither packet is received without error and both packets are therefore retransmitted. We also assume only full packets are transmitted.

Let those packets transmitting a given message from a station for the first time be message packets and let those packets transmitted as repetitions of a message be called repetitions. Let λ be the average rate of occurrence of message packets from a single active user and assume this rate is identical from user to user. Then the random point process consisting of the starting times of message packets from all the active users has an average rate of occurrence of:

$$r = k\lambda$$

where r is the average number of message packets per unit time from the k active users. Then, if we were able to pack the messages into the available channel space perfectly with absolutely no space between messages, we have:

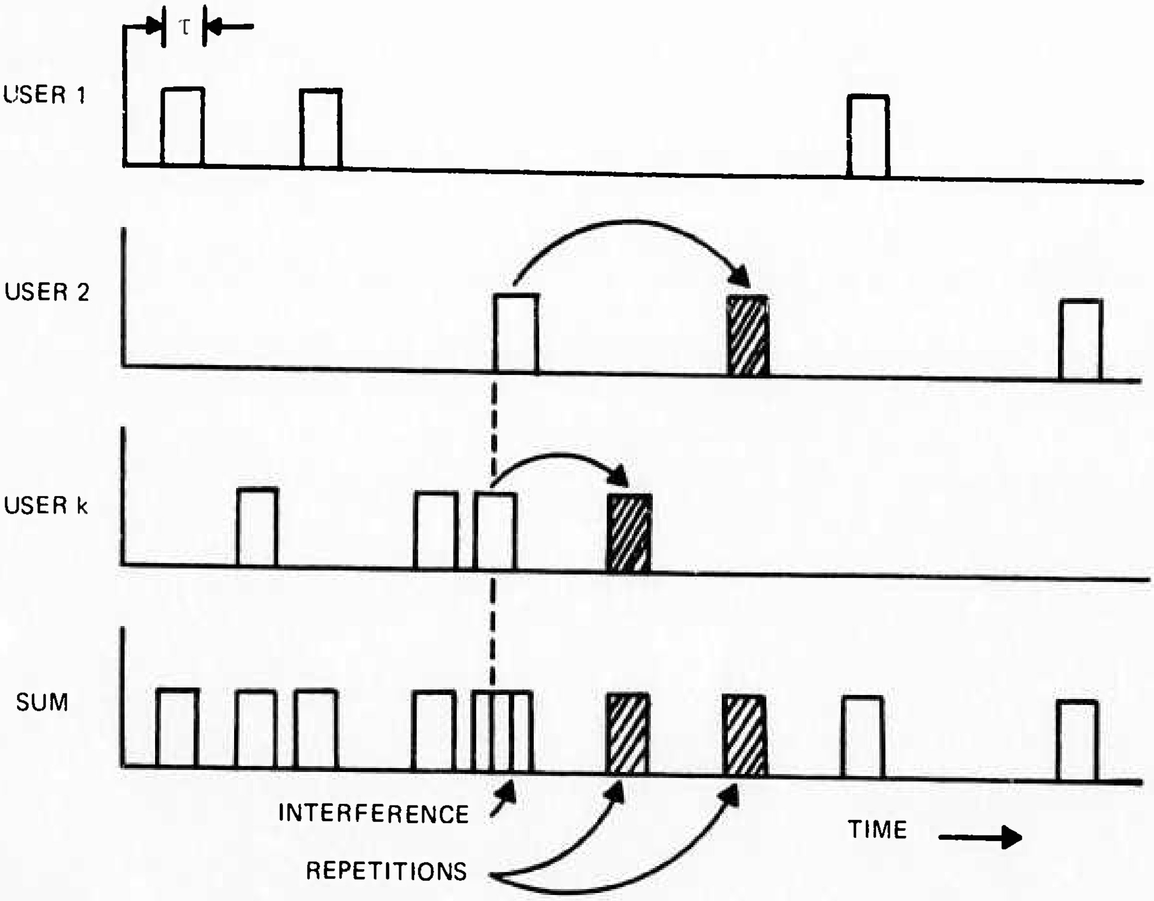


Figure 5.1: Random Access Packet Multiplexing

$$r\lambda = 1$$

Accordingly, we refer to rλ as the channel utilization. We will determine the maximum value of the channel utilization, and thus the maximum value of k, which this random access data communication channel can support.

Define R as the average number of message packets plus retransmissions per unit time from the k active users. Define Rτ as the *channel traffic* since this quantity represents the average number of message packets plus retransmissions per unit time multiplied by the duration of each packet or retransmission. We now calculate Rτ as a function of the channel utilization, rτ.

Assume the interarrival times of the point process defined by the start times of all the message packets plus retransmissions are independent and exponential. If the retransmission delay is large compared to τ , and the number of retransmissions is not too large, this assumption will be reasonably close to the true distribution. Under the exponential assumption, the probability that there will be no events (starts of message packets or retransmissions) in a time interval T is exp(-RT).

Two packets overlap if there exists at least one other start point τ or less seconds before or τ or less seconds after the start of a given packet. Hence, the probability that a given message packet or retransmission will be repeated because of interference with another message packet is:

$$[1 - \exp(-2R\tau)]$$

Thus, the average number of retransmissions per unit time is:

$$R[1 - \exp(-2R\tau)]$$

Therefore,

$$R = r + R[1 - \exp(-2R\tau)]$$

$$r = R\tau e^{-2R\tau}$$

The plot of $R\tau$ versus $r\tau$ is given in Figure 5.2.

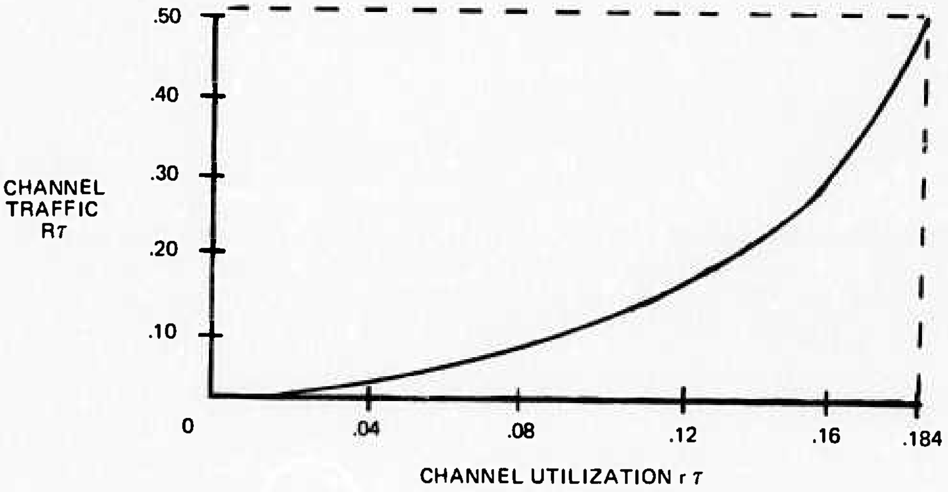


Figure 5.2: Channel Utilization vs. Channel Traffic

The channel utilization reaches a maximum value of $\frac{1}{2e} = 0.184$. For this value of $r\tau$ the channel traffic is equal to 0.5. The traffic on the channel becomes unstable at $r\tau = \frac{1}{2e}$ and the average number of retransmissions becomes unbounded. Thus, we may speak of this value of the channel utilization as the *capacity* of this random access data channel. Because of the random access feature, the channel capacity is reduced to roughly one sixth of its value if we were able to fill the channel with a continuous stream of uninterrupted data. To obtain the maximum number of interactive users the system can support, we get:

$$r\tau - k\lambda\tau = \frac{1}{2e}$$

Solving for the maximum number of active users, we have:

$$k_{max} = (2e\lambda\tau)^{-1}$$

A modification of the system called a slotted system allows message origination only at fixed intervals. It introduces some synchronization problems, but raises k_{max} to $(e\lambda\tau)^{-1}$. It is most important to realize that k_{max} is the number of users who can use the communications channel simultaneously. In contrast to the usual frequency or time multiplexing methods, while a user is not active, he consumes no channel capacity so that the total number of users of the system can be considerably greater than k_{max}.

Chapter 6
UPGRADING A TERRESTRIAL NETWORK USING SATELLITE LINKS

To illustrate the complexity of the decisions involved in upgrading communications, we now give one simplified example developed in Semiannual Report No. 2 of the introduction of a satellite into an ARPANET. The costs of the satellite facilities are roughly comparable to those proposed by a number of satellite companies with proposals already approved by the FCC.

The satellite facilities include: satellite channel; ground stations; Satellite Interface Message Processor (SIMP); and line connections from SIMP to station, or from IMP to station.

The following costs are assumed:

a. Satellite Segment:

Bandwidth (Kbps)	Costs ($/Mo.)
50 full duplex	2,500
230 full duplex	5,500
1,500	8,000

b. Local Loop (Station to SIMP, or Station to Central Office):

Bandwidth (Kbps)	Cost ($/Mo.)
50 full duplex	1,000
230 full duplex	1,300

c. SIMP. Two types of SIMP's are assumed. The regular SIMP has bandwidth greater than 1,500 Kbps and cost of 5,500 $/mo. This SIMP corresponds to the high speed version of IMP presently under development of BBN. It can support a combination of land traffic rates L and satellite traffic rate S such that:

$$L + 3S \leqslant 1{,}500$$

A small SIMP, with a bandwidth of 600 Kbps, and cost of 1,400 $/mo., is structurally similar to the H-316 IMP, and is presently being developed by BBN. The throughput constraint is :

$$L + 3S \leqslant 600$$

The network without satellite links used as a basis of comparison is a recent 43 node ARPANET configuration shown in Figure 6.1. We then consider one upgrade (see Figure 6.2) using only terrestrial links as well as two upgrades (see Figures 6.3 and 6.4) of the network using satellite links as well. In both satellite designs, five ground stations in San Francisco, Los Angeles, Washington, D.C., New York and Chicago are available for satellite access. We include the possibility of capacity reductions of terrestrial links from 50 Kbps to 19.2 and 9.6 Kbps. In the first satellite design, we use point-to-point access which divides the satellite channel bandwidth into subchannels, each corresponding to a full duplex point-to-point connection between given ground stations. In the second satellite design, we use the slotted random access packet mode described in the last Section. Computed for each network configuration is: total cost; terrestrial costs of all terrestrial links; satellite cost; cost of SIMP (if applicable); connection from SIMP to station or from IMP to station, and satellite bandwidth; total throughput; traffic on satellite and satellite channel delay. The results are in Table 6.1.

Several observations can be made from this simple example. Because of the high cost of network to ground station connections, satellite links become attractive only for throughput levels which are about 50% higher than the ARPANET configuration. Furthermore, even to achieve these efficiencies, changes in network topology and reduction of some link

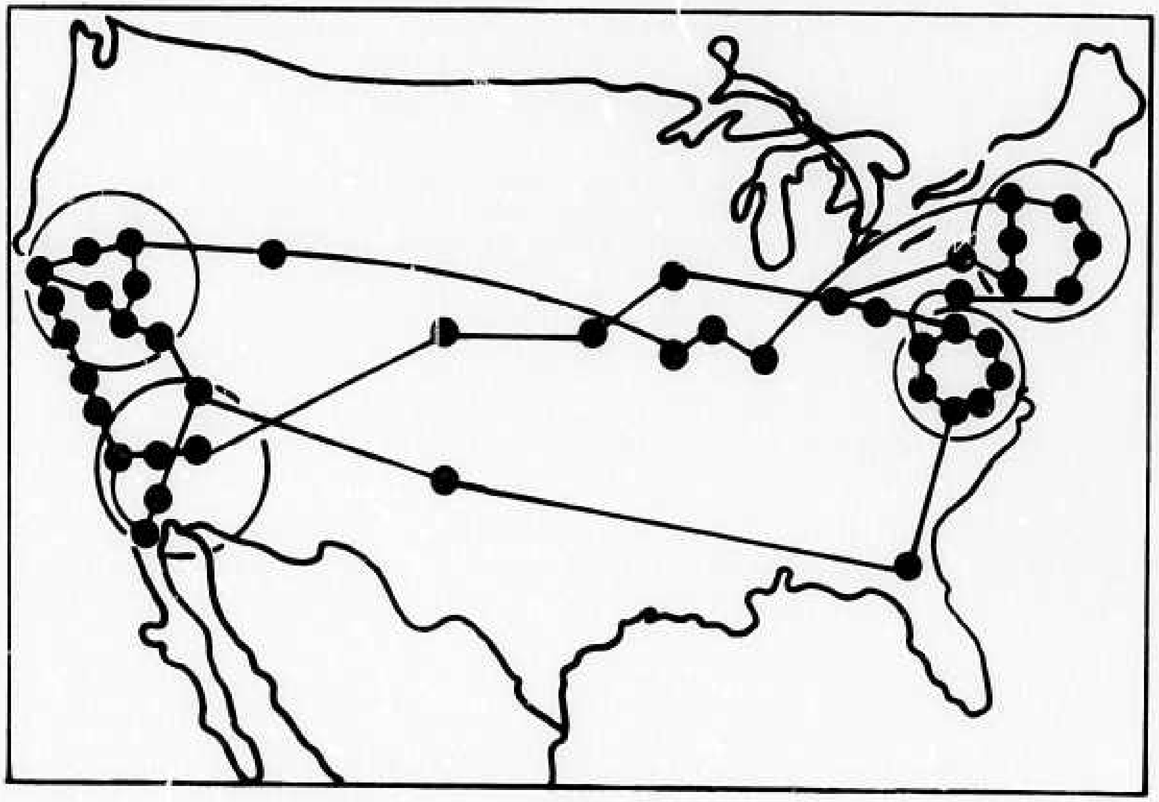


Figure 6.1: Present ARPANET Configuration (October, 1973)

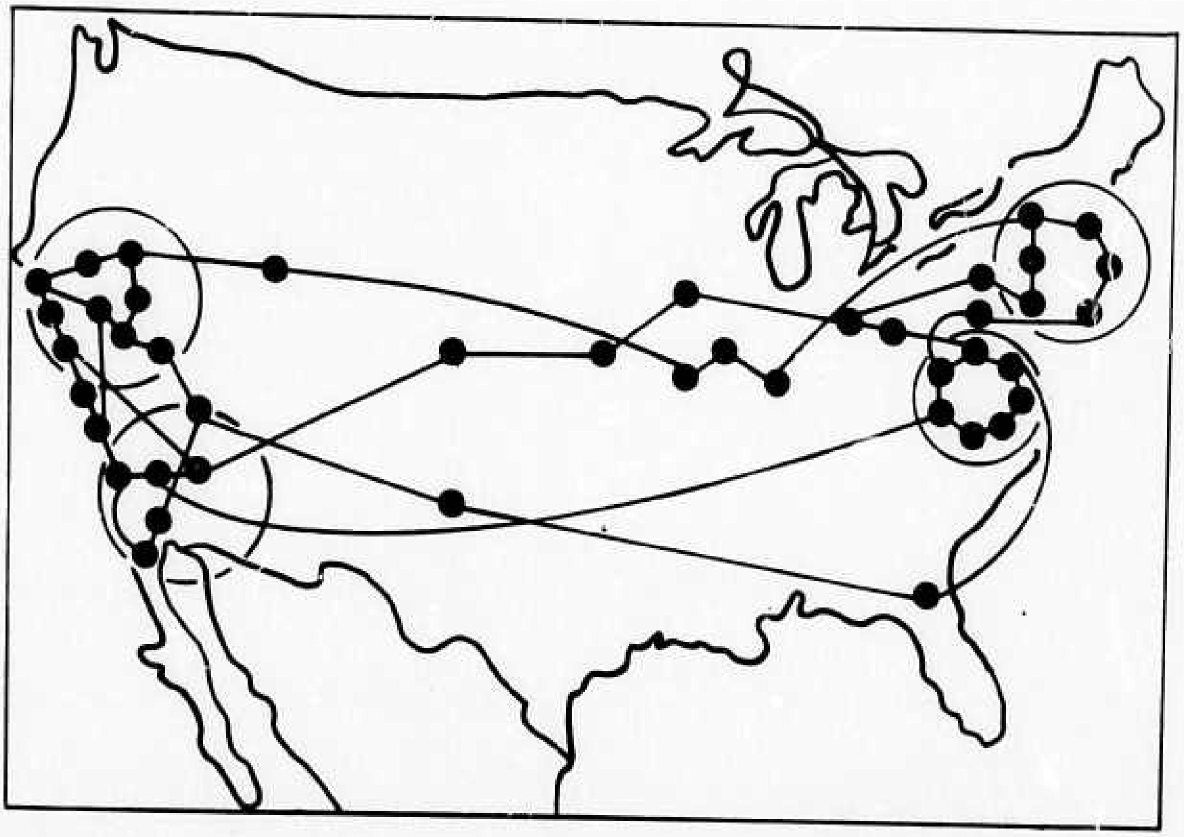


Figure 6.2: Upgraded 43-Node Configuration

capacities to 19.2 Kbps and 9.6 Kbps are required. Examination of the designs also indicates that the proper location of ground stations is important. For example, the location of a ground station in Chicago allows the reduction of channel capacity on cross country connections.

The comparison of cost-throughput trends between implementations with and without satellite, when network throughput is increased, shows that satellite implementations can provide higher throughput at a lower cost, especially if the terrestrial network is reoptimized; but the savings are by no means guaranteed by only routine introduction of satellite links. Many tradeoffs are involved and careful optimization is required.

Furthermore, it is clear that many other factors disregarded in this simplified analysis must be taken into consideration before general cost-performance trends are evident. In particular, the evaluation of point-to-point satellite link cost assumed that the standard IMP software can support satellite rates up to 230 Kbps. There are indications, however, that such a high rate will require modifications of the IMP hardware and software, and therefore will raise the cost of point-to-point links to the same levels as those of random access.

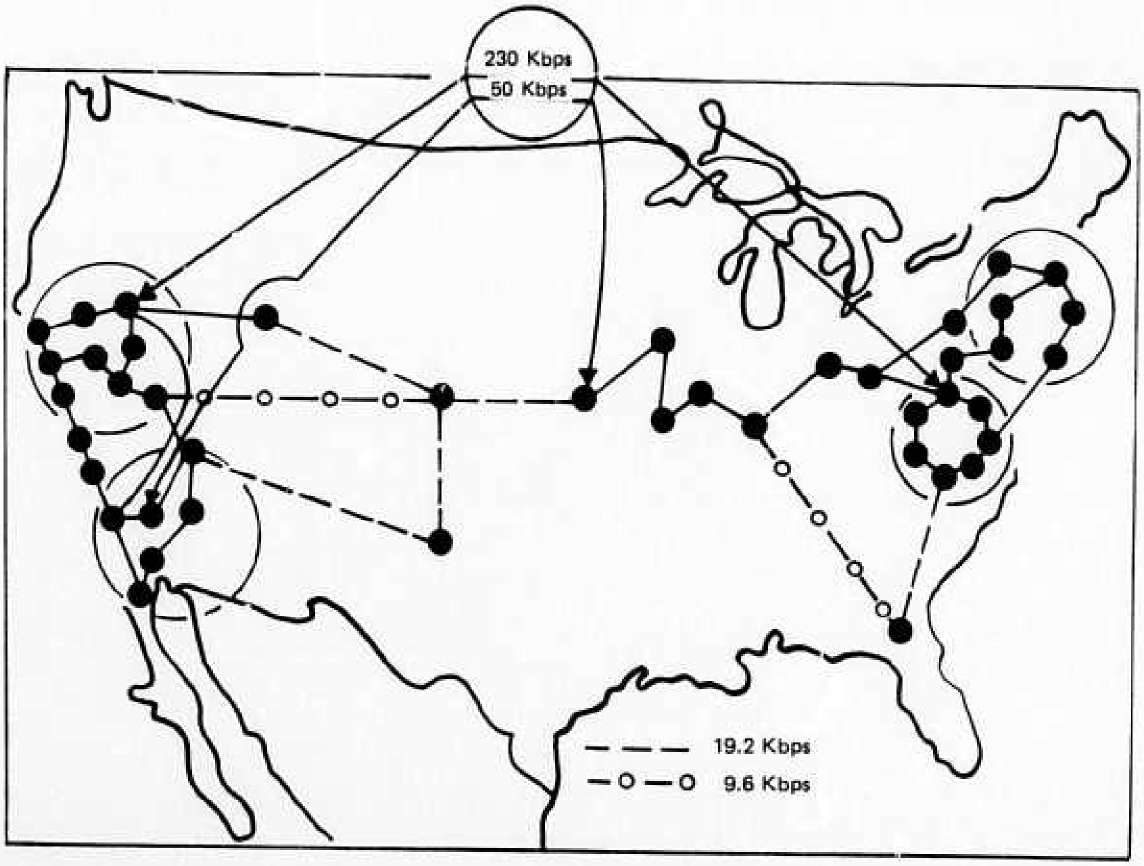


Figure 6.3: 2 Point-To-Point Links

As for satellite bandwidth efficiency, it must be mentioned that, with additional software cost, reservation techniques for multiple access can be implemented on the SIMP; and such techniques can theoretically increase effective satellite bandwidth up to full utilization. Furthermore, multiple access allocates satellite bandwidth dynamically, according to traffic pattern changes, and if needed, allows any two stations to use the full channel; while point-to-point access corresponds to a rigid bandwidth allocation between pairs of stations.

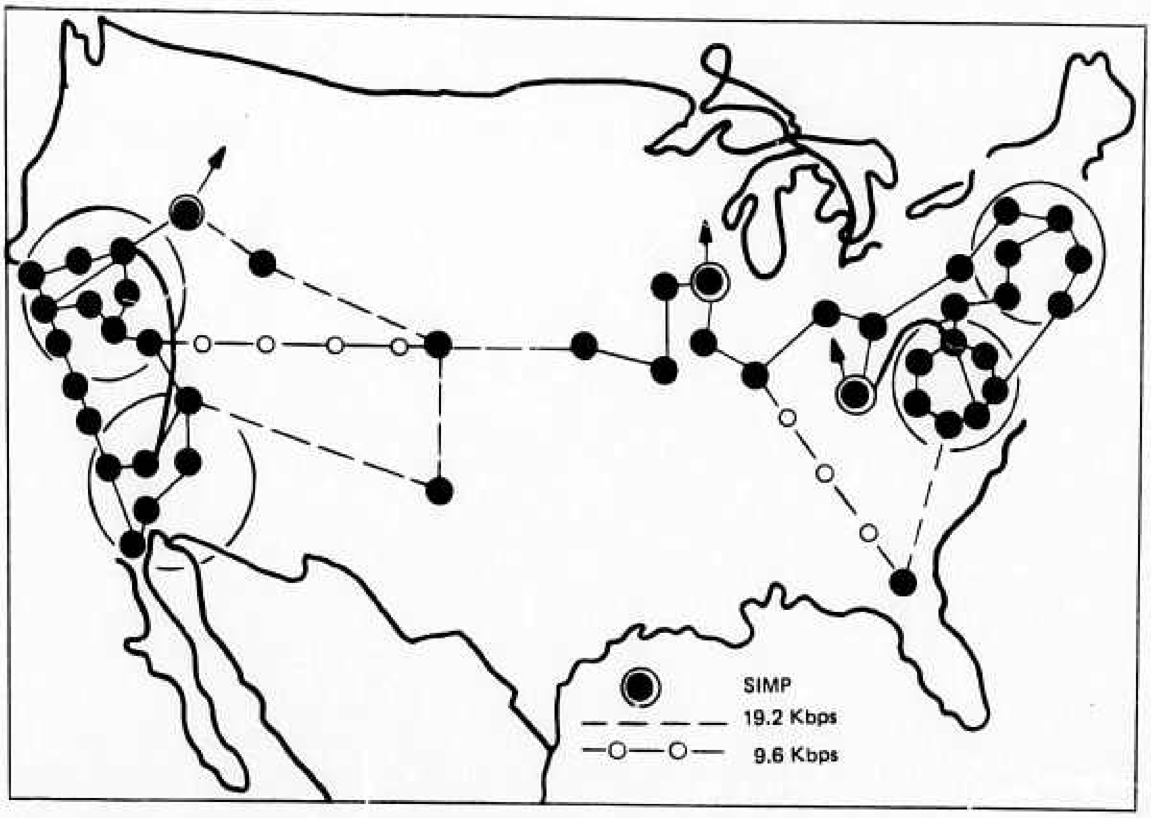


Figure 6.4: 3 Regular SIMP's

Table 6.1: Upgrading of Packet Switched Terrestrial Network Using Satellite Links

Network Configuration	Satellite Delay (sec)	Satellite Traffic (Kbps)	Cost K$/mo.			Throughput Kbps	Total Cost Throughput $/bit
			Terrestrial	Satellite	Total		
Present 43 node configuration (Figure 6.1)	.....	......	93	....	93	447	.208
Upgraded 43 node configuration (Figure 6.2)	.....	......	112.9	....	112.9	635	.178
Two point to point links (Figure 6.3)	.27	330	75	16.9	91.9	654	.141
3 regular SIMPS at San Francisco, Chicago and Washington, D.C. (Figure 6.4)	.5	393	79	29.3	108.3	686	.158

Chapter 7
MULTIDROPPED LINE NETWORKS FOR LOCAL ACCESS

7.1 Introduction

The interconnection of different time-sharing computers through a sophisticated communications subnet in the ARPANET gives terminal users access to a variety of time-sharing resources. Initially, ARPANET development was directed toward computer-computer communications and user protocols. Originally, only terminals connected directly to a computer in the network had access to the network. The successful completion of this initial phase led to a desire to complement resource development with increased user access. Many TIP's have already been installed and are currently in use, connecting users with a terminal, but with no local Host computer, to the network.

The use of the ARPANET approach within the Defense Department would involve hundreds of Hosts accessed by tens of thousands of low speed terminals. Effective, economical terminal access to the ARPANET, and to similar networks, will depend on continued development of such facilities as TIP's as well as on complementary development of techniques for cost-effective utilization of these facilities.

There are several ways to provide terminal access into the network. In particular, multidrop lines for connecting terminals to access ports, ring networks, CATV Systems and packet radio techniques all provide potential low cost network access methods. It is necessary to investigate all of these schemes to evaluate the merits of each and to determine the conditions under which each may be preferable. It is not unreasonable to anticipate that many approaches may be applicable within the same network. In this section, we consider multidrop lines. Other local access methods are considered in the following sections.

In NAC's semiannual reports, algorithms are described for the multidrop line-layout and TIP location problems. These algorithms consider not only the line layout, but also the number, location, and characteristics of the ports into the network. Models were developed to estimate the cost of connecting terminals to the network through the use of TIP's and multidrop lines. The estimates cover a wide range of terminal numbers and traffic conditions and serve as a basis for comparison with other access approaches and as a measure of the effectiveness of new design tools.

7.2 Network Modeling for Terminal Access

The process of investigating and developing approaches to the design of networks for terminal access requires, as a first step, the construction of an appropriate model. In this study, a primary goal is to determine the tradeoffs and parametric dependencies present in various approaches to terminal access. To compare these approaches, it is necessary to have a common data base to which each approach can be applied. Such a base has been constructed in the form of a model for the number and geographic distribution of terminals, for the terminal and terminal user, and for the multidrop communication lines. It is also necessary to model those components of a design peculiar to the particular approach considered. In this report we present a model for the TIP system.

7.2.1 Population

The cost of terminal access will depend on a variety of factors, including the number of terminals to be connected and their geographic distribution. To determine the parametric dependence of cost on the number of terminals, populations of from 100 to 2000 terminals were considered. The figure of 100 reflects the anticipated near-future requests for terminals. The figure of 2,000 reflects an order of magnitude estimate of the number of terminals that can be expected to be served by a fully developed network. (The above consider only terminals without a local Host.)

To have a meaningful data base, it is necessary to geographically distribute the terminals in a sensible manner. Terminals were located on the basis of population density because of the success of this approach in previous NAC investigations. A rectangular region was determined for each city, or collection of cities, to reflect the feasibility of the region to support a population segment with access to urban facilities. Thus, consideration was given to natural geographical boundaries, such as mountains, lakes, and coast lines, to major roads in the area, to the number of nearby smaller communities, and to the natural pattern of urbanization between relatively close major population centers. Using this approach, 123 regions were defined, with varying sizes of approximately 70 square miles.

Once a number of terminals has been allocated to a region in proportion to population, the geographic positions of the terminals within the region are uniformly randomly distributed. With a large number of terminals, it is reasonable to anticipate that some may be located at points with no discernible geographic significance; therefore, a fraction a of the terminals were located at random in a large geographic segment: east of Denver, west of Pittsburgh, north of Austin, and south of Milwaukee. The fraction a was selected on a sliding scale as shown in Table 7.1.

7.2.2 Terminal-Terminal User

Even though network resources in the ARPANET have been extended far beyond traditional time-sharing, the interactive user retains a significant role in network usage, and the

Table 7.1: Terminal Population

Number of Terminals	In Regions	Random
100	95%	5%
200	95%	5%
500	90%	10%
1,000	90%	10%
2,000	85%	15%

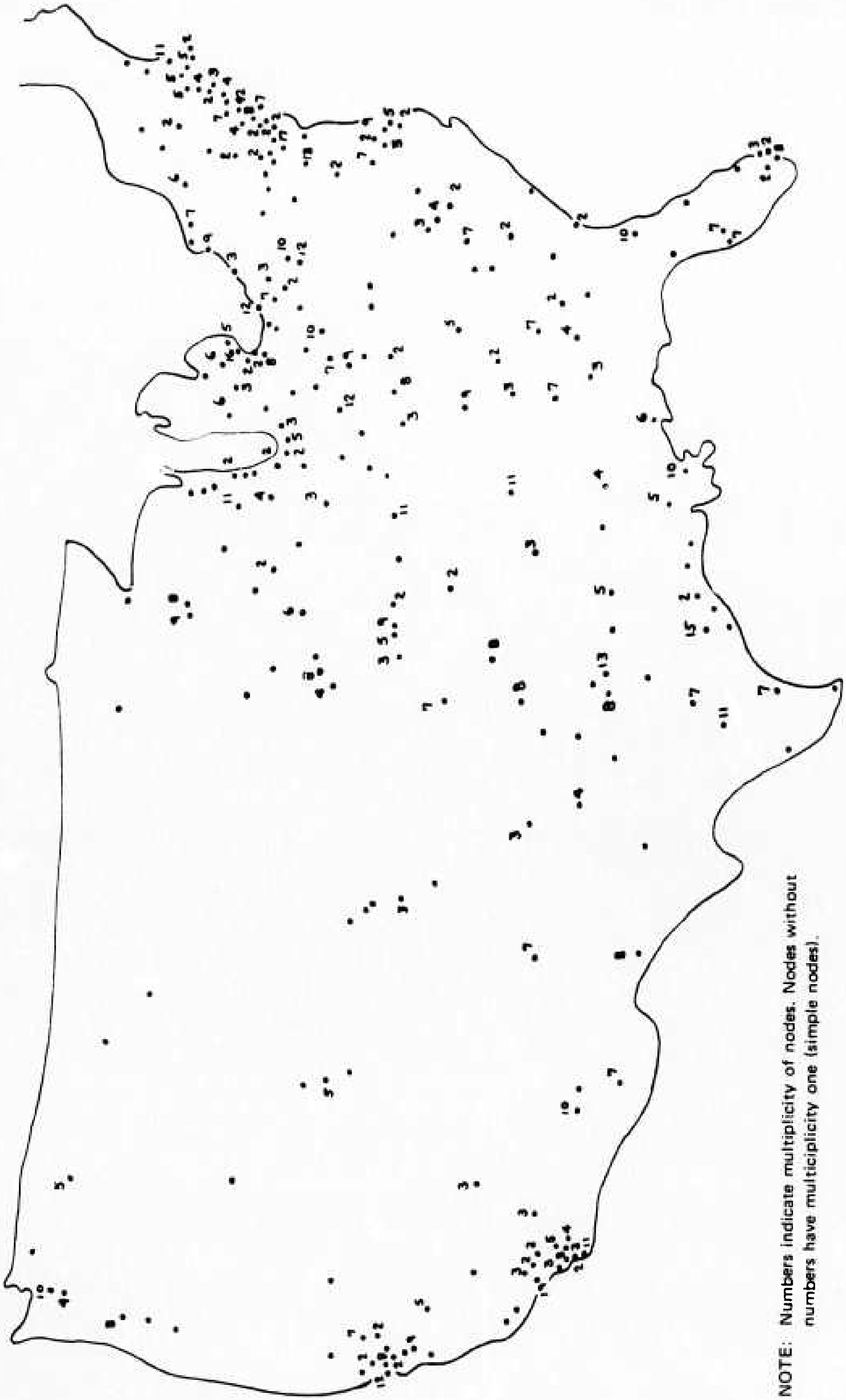
extension of accessibility to a terminal basis will give even greater significance to terminal-computer traffic. To effectively design and evaluate terminal access networks, it is necessary to model the terminal traffic. Two of the few definitive papers on time-sharing modeling from a communications perspective have been written by Jackson and Stubbs [33], and Fuchs and Jackson [22]. The following traffic characteristics of a terminal during a period of use are based on their results for time-sharing systems used in scientific applications, and extended in consideration of advances in terminal technology, higher speed lines being used, and more sophisticated time-sharing users and programs.

	Average Message Length	Minimum Average Traffic	Maximum Average Traffic
User Input	12.1 characters	.1 characters/second	1 characters/second
Computer Response	52.8 characters	1.0 characters/second	10 characters/second

In this study, traffic level was varied, with a range of variation from the minimum to the maximum values indicated above. The minimum average traffic level reflects the results of the noted study for scientific applications using low speed facilities and ordinary time-sharing programs. The maximum average traffic level reflects an extension of these results in consideration of "smart-fast" terminals, higher speed communication facilities, more advanded time-sharing programs, and more sophisticated users. For comparison purposes, all network designs were based on busy hour conditions of all terminals being active.

7.2.3 Communication Facilities

The current ports for access by terminals to the ARPANET (TIP's) may connect terminals directly, or remotely through modems and phone lines. In this study, a large number of terminals serving interactive users are considered, and to economically connect all the terminals, multidrop lines were used. The multidrop communications facility will be assumed to be a standard voice-grade line as described below.



NOTE: Numbers indicate multiplicity of nodes. Nodes without numbers have multiciplicity one (simple nodes).

Figure 7.1: Locations of the 1000 Nodes

7.2.3.1 Multidrop Line

Capacity (full duplex)	1200 bps
Cost	\$.50/mile + \$40/drop

The monthly cost is based on the Government rate of \$.42/mile plus 20% for non-direct routing. It should be noted that in this model the number of drops on a line is restricted only by the traffic constraint. In reality, the number is often additionally restricted by telephone company practices. The effect of a more severe restriction is easily seen by simply assuming a correspondingly higher traffic level.

7.2.3.2 TIP

The approach considered will be a TIP serving as the root of a centralized network of terminals. In this section we note the significant features of the TIP. The TIP, as described by Ornstein et. al [44], is characterized in Figure 7.2. Its characteristics indicate:

a. The TIP has 63 terminal I/O slots.

b. Each slot can handle direct terminal connections or connections via modems.

c. Asynchronous data rates handled by the TIP include 1200, 1800, and 2400 bps.

d. The TIP has a terminal program throughput of:
  - 100 Kbps one way traffic if messages are long ("many characters"), and
  - 5-10 Kbps if each terminal message is a single character.

e. The TIP uses 5% of its processing capacity to act as an IMP.

f. The TIP uses 10% of its processing capacity to field MLC interrupts.

g. The bandwidth capability of the TIP is summarized approximately by the formula:

$$P + H + 11T \leqslant 850$$

where P = total phone line traffic (Kbps)
H = total Host traffic (Kbps)
T = total terminal traffic (Kbps)

and full duplex units count twice baud rate, i.e., standard full duplex 50 Kbps phone line counts 100, and full duplex ASR-33 counts as 0.22. As noted, the TIP does not presently handle multidrop lines. Consequently, adjustments to its characteristics are required for the network model.

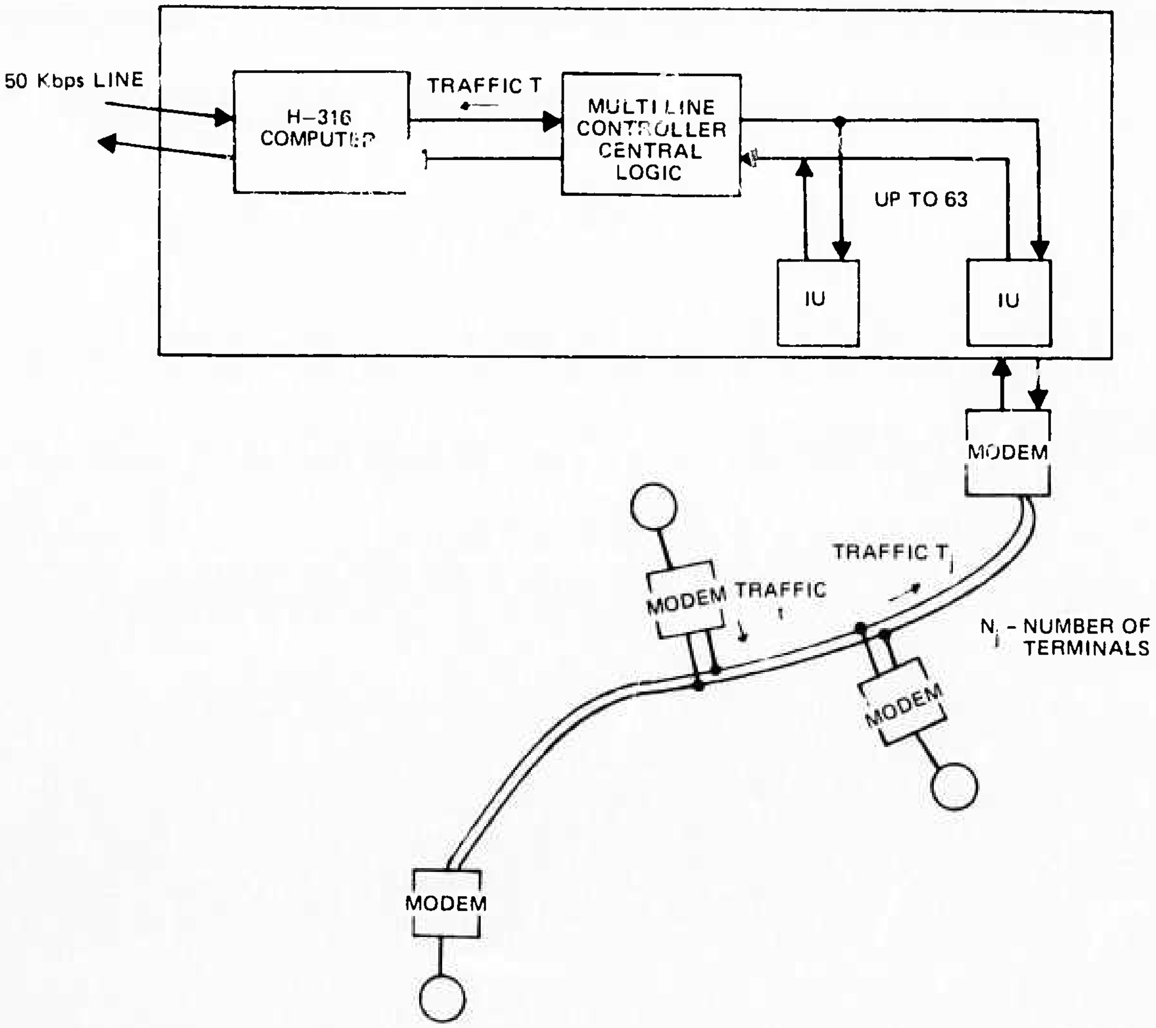


Figure 7.2: TIP Interfaces

The above noted characteristics reflect TIP capacity as a message concentrator for terminals. Additional consideration must be given to TIP cost, which includes estimated rental rate, the cost of its interconnect to the ARPANET, and the cost of the modems necessary to connect terminals. Therefore, the total cost is a function of the TIP's geographical relationship to the rest of the ARPANET, the topology of its interconnection, and the number of modems required for terminal connections. These costs will be determined as follows:

50 Kbps line (ARPANET interconnect) $5/mile + $425/end
(based on current ARPANET experience)

1200 bps line (terminal connection) $17/modem
(current standard cost)

TIP rental (assumed TIP cost of $100,000 to be amortized over 5 years at 10% interest compounded quarterly)	$2500/month

7.3 Design Results

As noted, the currently designed TIP has no provision for the support of multidrop lines. Both hardware and software modifications may be necessary for the acceptable addition of this capability. Significant requirements are line protocol for the multidrop lines and more extensive file manipulation resulting from the larger number of terminals. The line protocol must permit line utilization of approximately 50%, a conservative figure based on the use of ordinary polling techniques for multidrop lines. With the previously described traffic range of 10 bps to 100 bps, this gives a possible range of 6 to 60 terminals on a line. The sixty-three possible connected lines allow a maximum demand of 37.8 Kbps to be placed on a TIP by the terminals. Using the maximum demand figure in the TIP bandwidth formula indicates that such a TIP would have sufficient additional bandwidth to support a Host and also be connected to the ARPANET in a manner consistent with current practices. However, the number of terminals a TIP handles in the maximum demand case (378 to 3780) is far beyond the current maximum configuration (63). This increase in number should be anticipated as causing considerable additional overhead for file manipulation. Furthermore, additional overhead may be anticipated due to the burden of a multidrop line protocol. Under these conditions, the maximum number of terminals that a TIP can handle is assumed to be 630, one order of magnitude greater than its current direct connection capacity. This gives a network model as below:

TIP	1) up to 63 line connections 2) up to 630 terminals
Lines	up to $\frac{600}{t}$ terminals/line where t is the traffic/terminal in bps

Cost is estimated as a function of the number of terminals and their traffic level, subject to fixed TIP locations. In Table 7.2 below, costs are given for 100 terminal system at a traffic level of 100 bps each for different numbers of TIP's at different locations.

These results show that a higher number of TIP's yields lower line costs, but not necessarily a lower total cost. Consequently, the number of TIP's is varied until a local minimum is reached. Table 7.3 gives preliminary estimates of the cost of terminal connection as a function of the number of terminals and the level of traffic. Results are shown as points connected by straight line segments in Figure 7.3. The curves suggest that for low numbers of terminals, and thus, low numbers of TIP's, the line constraints and TIP locations have significant impact on cost. For large numbers of terminals, and thus, larger numbers of TIP's, costs are less sensitive to TIP placement and line constraints. Simplified illustrations of several of the network designs are given in Figures 7.4 through 7.7. Note

Table 7.2: Network Cost and TIP Location Relationship

# TIP's	Locations	100 Terminals (100 bps each) Monthly Line Costs	Monthly Line Costs And TIP Rental
1	Chicago	$14,007	$16,507
	Memphis	14,501	17,001
	New York	17,190	19,690
2	New York-Los Angeles	13,091	18,091
3	New York-Los Angeles-Chicago	11,375	18,875
	New York-Los Angeles-Chicago	11,302	18,802

Table 7.3: Preliminary Terminal–TIP Experiment Results

Number of Terminals	Traffic (bps)			
	10	20	50	100
100	$ 13,095	$ 13,231	$ 15,146	$ 17,607
200	18,906	19,875	23,373	31,818
500	36,050	39,138	49,208	56,099
1,000	66,775	72,893	83,886	94,189
2,000	119,570	125,085	144,759	165,817

that for low traffic (10 bps) it is cost effective to use as few TIP's as possible, while for high traffic (100 bps), savings are achieved by using more than the minimum number of TIP's. (With low traffic, many terminals can be chained together on one line to economically connect distant terminals to a TIP. With high traffic, only a few terminals can be placed on a line, and distant terminals result in several long, uneconomical lines.)

Since TIP's are relatively expensive when compared to conventional multiplexers, these simpler devices to achieve economy of scale will also be investigated as an alternative architecture.