Bolt Beranek and Newman Inc.

Report No. 4038

ARPANET Routing Algorithm Improvements
Third Semiannual Technical Report

E.C. Rosen, J.G. Herman, I. Richer, and J.M. McQuillan

March 1979

Prepared for:
Defense Advanced Research Projects Agency
and
Defense Communications Agency

UNCLASSIED

SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered)

REPORT DOCUMENTATION PAGE

READ INSTRUCTIONS BEFORE COMPLETING FORM

1. REPORT NUMBER: 4088

2. GOVT ACCESSION NO.: AD-A086340

3. RECIPIENT'S CATALOG NUMBER:

4. TITLE (and Subtitle): (6) ARPANET Routing Algorithm Improvements ~~Third Semiannual Technical Report~~

5. TYPE OF REPORT & PERIOD COVERED: Semiannual Technical Report 10/1/78 - 4/1/79

6. PERFORMING ORG. REPORT NUMBER: 4088

7. AUTHOR(s): (10) E.C. Rosen, J.M. McQuillan, J.G. Herman, I. Richer

8. CONTRACT OR GRANT NUMBER(s): (15) MDA903-78-C-0129

9. PERFORMING ORGANIZATION NAME AND ADDRESS: Bolt Beranek and Newman Inc., 50 Moulton Street, Cambridge, MA 02138

10. PROGRAM ELEMENT, PROJECT, TASK AREA & WORK UNIT NUMBERS: JJ ARPA Order-3491, ARPA Order No. 3491

11. CONTROLLING OFFICE NAME AND ADDRESS: (11) Defense Advanced Research Projects Agency, 1400 Wilson Blvd., Arlington, VA 22290

12. REPORT DATE: Apr 1979

13. NUMBER OF PAGES: 168

14. MONITORING AGENCY NAME & ADDRESS (if different from Controlling Office): Defense Supply Service - Washington, Room 1D 245, The Pentagon, Washington, DC 20310

15. SECURITY CLASS. (of this report): UNCLASSIFIED

15a. DECLASSIFICATION/DOWNGRADING SCHEDULE:

16. DISTRIBUTION STATEMENT (of this Report): UNCLASSIED/UNLIMITED

(14) BBN-4088 (12) 171

17. DISTRIBUTION STATEMENT (of the abstract entered in Block 20, if different from Report)

(9) Semiannual technical rept. no. 3, 1 Oct 78-1 Apr 79

18. SUPPLEMENTARY NOTES

19. KEY WORDS (Continue on reverse side if necessary and identify by block number): computer networks, routing algorithms, ARPANET, line up/down procedures, distributed data base, buffer management, network measurement, network testing, updating

20. ABSTRACT (Continue on reverse side if necessary and identify by block number):

This report describes progress made during the third six months of a contract to make several improvements to ARPANET routing. During this period all aspects of the ARPANET's new routing algorithm were implemented and the new algorithm was run through an extensive series of tests. The results of these tests are presented, along with a discussion of our testing goals, techniques, and tools. A full description of the procedures needed to handle a dynamically changing topological data base is also presented.

(continued)

DD FORM 1 JAN 73 1473 EDITION OF 1 NOV 65 IS OBSOLETE

UNCLASSIFIED

SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered)

060100

20. continued
Measurements on the performance of the ARPANET's new line up/down protocol are presented. Lastly, the procedures used in the ARPANET for managing buffer space are described.

BBN Report No. 4088

ARPANET Routing Algorithm Improvements

Third Semiannual Technical Report

April 1979

SPONSORED BY
DEFENSE ADVANCED RESEARCH PROJECTS AGENCY AND
DEFENSE COMMUNICATIONS AGENCY (DOD)
MONITORED BY DSSW UNDER CONTRACT NO. MDA903-78-C-0129

ARPA Order No. 3491

Submitted to:

Director
Defense Advanced Research Projects Agency
1400 Wilson Boulevard
Arlington, VA 22209

Attention: Program Management

and to:

Defense Communications Engineering Center
1860 Wiehle Avenue
Reston, VA 22090

Attention Dr. R.E. Lyons

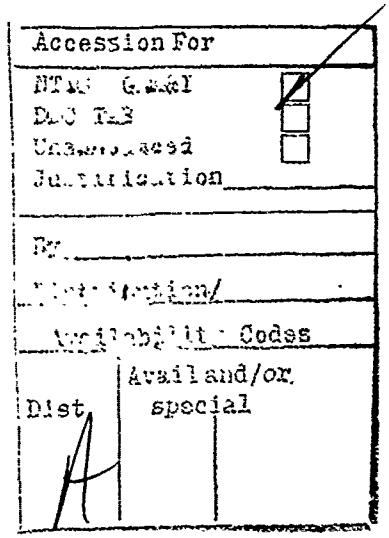


TABLE OF CONTENTS

INTRODUCTION

This report covers work performed during the period from October 1, 1978 to April 1, 1979 on the contract to study, design, and implement improvements to the ARPANET routing algorithm.

In September 1978, a new line up/down protocol was installed in the ARPANET. During the past five months we have been collecting statistics on the performance of the protocol. These statistics, summarized in Chapter 1, show that the protocol is working well.

The design and implementation of the new routing algorithm in the Honeywell IMPs is now completed. Chapter 2 discusses the last part of the algorithm to be designed, the data base management procedures. The new routing algorithm requires a data base which specifies the network's topology. Since the ARPANET topology changes frequently, as IMPs are reconnected or even reconfigured, it is important for the data base to be modifiable on a dynamic basis. That is, when a topological change occurs, the data base should automatically change to reflect the new topology, without requiring human intervention. Chapter 2 discusses the procedures we developed for maintaining the data base dynamically.

The new algorithm has been installed in most of the network in parallel with the old algorithm. The old algorithm is still used ordinarily for operating the network, but the network is capable of switching over to the new algorithm. During the past several months we have run an extensive series of tests in which the network was operated with the new algorithm, and these tests are discussed in Chapters 3-6. Testing a new routing algorithm is a complex task, which must be approached systematically, and with a specific set of goals in mind. Our approach to the testing is discussed in Chapter 3. In order to be able to test the new algorithm in the ARPANET while minimizing the possibility of disrupting network operations, we had to develop a complicated series of testing procedures. These procedures are discussed in Chapter 4. In Chapter 5 we discuss the software tools which we developed in order to test the new algorithm.

The results of our testing are presented in Chapter 6 and in the five appendices. Our main results are:

1) Utilization of resources (line and processor bandwidth) by the new routing algorithm is as expected, and compares quite favorably with the old algorithm.

2) The new algorithm responds quickly and correctly to topological changes.

3) The new algorithm is capable of detecting congestion, and will route packets around a congested area.

4) The new algorithm tends to route traffic on min-hop paths, unless there are special circumstances which make other paths more attractive.

5) The new algorithm does <u>not</u> show evidence of serious instability or oscillations due to feedback effects.

6) Routing loops occur only as transients, affecting only packets which are already in transit at the time when there is a routing change. The few packets that we have observed looping have not traversed any node more than twice. However, the loop can be many hops long.

7) Under heavy load, the new algorithm will seek out paths where there is excess bandwidth, in order to try to deliver as much traffic as possible to the destination.

Of course, the new routing algorithm does not generate optimal routing -- no single-path algorithm with statistical input data could do that. It has performed well, however, and we are ready to cut the network over to the new algoritm permanently.

As a prelude to developing improved congestion control techniques for the ARPANET, we have been investigating the buffer management procedures currently implemented in the ARPANET.

These are described in Chapter 7, and some possible improvements to the procedures are discussed.

1. LINE UP/DOWN MEASUREMENTS

During September 1978 the new line up/down protocol (described in our previous semiannual reports) was installed in the ARPANET. In addition to providing better performance than the old protocol, the new protocol provides greater flexibility since the parameter values can be adjusted over a significant range. As a result of measurements made both prior to the actual installation and during the initial operation of the new protocol, the following parameter values were selected for the various types of network links. The notation (k,n) signifies that if the higher numbered IMP -- the master -- misses k I-Heard-Yous during n successive intervals, then the line is brought down; NUP denotes the number of consecutive I-Heard-Yous needed to bring a line up:

speed	type	(k, n)	NUP	interval
50 kbps	terrestrial, satellite	(4,20)	60	640 ms. (slow tick)
230 kbps	terrestrial	(5,5)	60	128 ms.
9.6 kbps	terrestrial, satellite	(4,20)	60	1280 ms.

Measurements of the number of line downs were taken over approximately a five-month period (147 days) from October 1978 through March 1979. Because of the specific implementations of the protocol and the measurement package, the number of times the

master declares a line down is a more accurate indication of the number of line failures than the number of times the slave declares the line down. Therefore, except where otherwise specified below, the number of line downs refers to the number of times the master IMPs declared lines down. It is important to note, however, that this measurement overestimates the actual number of line failures because a given line failure may cause several line down reports, and because other network phenomena can result in a line being declared down. (A striking example occurred during a four-week period when IMP 13, GUNTER, had many power outages; the resulting line downs reported by neighbors of IMP 13 were excluded from the measurements given below.)

We now discuss the observed performance of the various types of links. For the 50 kbps terrestrial links, the most common links in the ARPANET, we observed an overall average interval between failures on an average line to be 1.8 days. That is, a "typical" line in the network failed slightly more often than once every second day. However, during the first two months of the five-month measurement period, the average interval was 1.4 days, whereas during the last three months, the average was 2.2 days, a 50% improvement over the first two-month period. Figure 1 shows the weekly measurement data. We do not have an explicit explanation for this improvement, although it could be the result of the various topology changes that have been made during the

past six months. During the most recent three-month period, three lines (between IMPs 62 and 13, IMPs 62 and 4, and IMPs 59 and 33) accounted for about 18% of the line down reports. On each of these lines, the average interval between failures was 0.72 days; and for the remaining lines the average interval then improves to 2.5 days. All the above values were obtained by dividing the number of lines by the average number of recorded line downs per day. Since the topology of the ARPANET changes frequently, the "number of lines" is really an average; also, it should be noted that Pluribus IMPs were not instrumented to provide measurement statistics, and reports from test IMPs in BBN's lab were ignored; thus there were typically about fifty-three 50 kbps lines on which measurements were taken.

The line between IMPs 15 and 36 is the only 50 kbps satellite link, and IMP 36, HAWAII, is a stub. Thus, if there is a line failure a report cannot be transmitted from the master. For this line we therefore examined the number of downs reported by the slave side: over the five-month measurement period the average interval between failures was 1.3 days, significantly worse than for the terrestrial links.

The average failure interval for the 9.6 kbps terrestrial line between the LONDON and NORSAR IMPs was 0.62 days (15 hours). As with the HAWAII line, LONDON is the master and is a stub, so the data is based on reports from the slave.

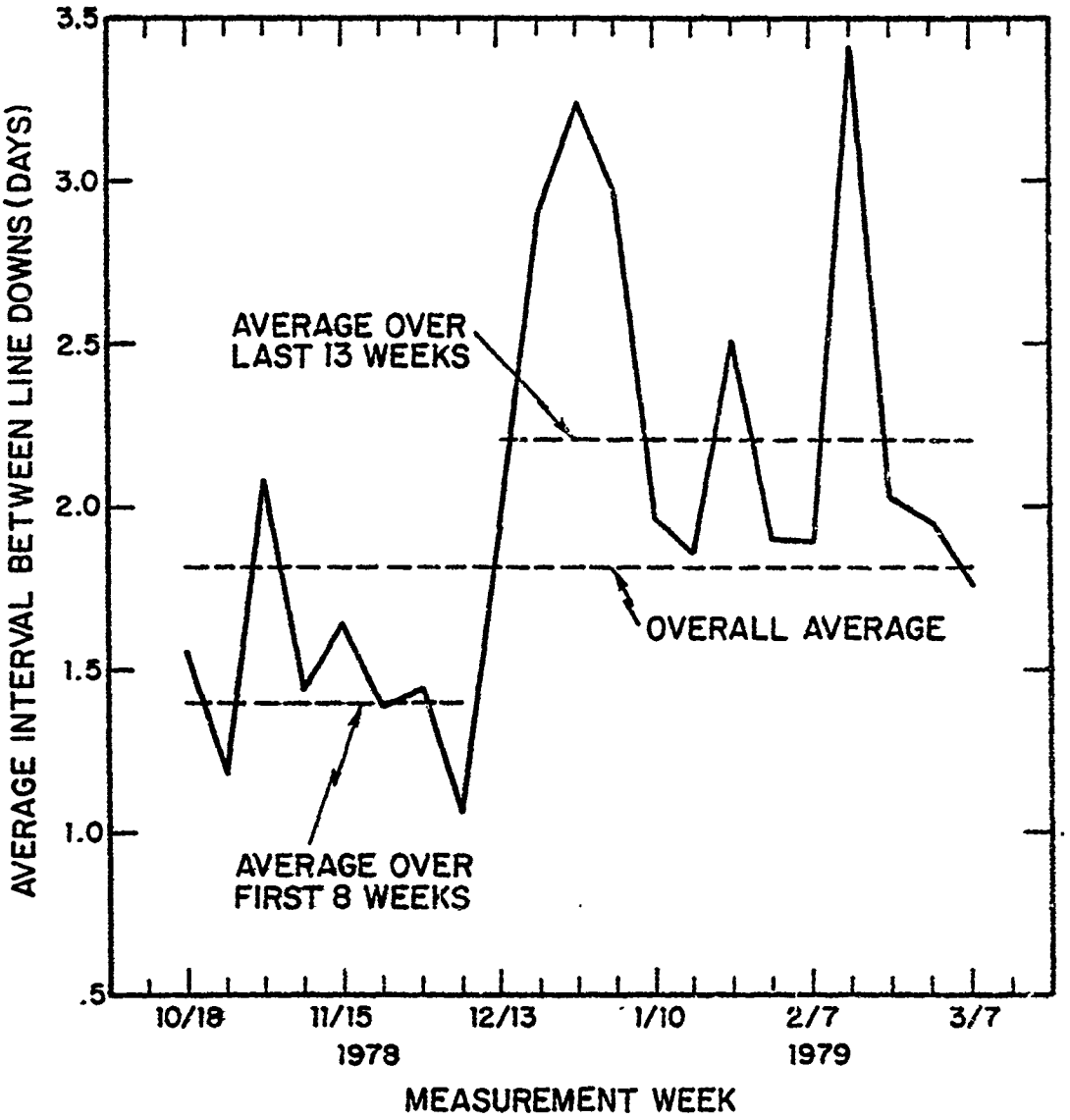


Figure 1-1 Line down data for 50 kbps terrestrial links

No data was obtained on the 9.6 kbps satellite channel linking NORSAR and SDAC because the master (NORSAR) is a stub, and SDAC, a Pluribus, does not report line downs.

Finally, for a 230 kbps line, which uses different parameter values from the lower speed lines, the average interval between failures was 7.7 days. (There are six high speed lines currently in the network.)

2. THE SPF TOPOLOGY DATA BASE

The original specification of the data structure for the SPF routing algorithm assumed a fixed topology data base. No consideration was given, at first, to the issues of initializing and maintaining this data base. To test the first few stand-alone versions of SPF, the network line table and the IMP connectivity table were pre-assembled into the program.

This is not sufficient for the final implementation. The ARPANET topology is not fixed. It changes constantly due to retrunking, as well as the addition, relocation and deletion of nodes. These activities regularly involve standard sites and they occur many times a day in the BBN test lab. It is clear that the data base must be capable of responding to these changes. To deal with the problems of maintaining dynamic topology tables, we constructed the data base management module, which detects and handles messages about lines that are not already part of the data base as well as performing needed consistency checking. It also implements a mechanism for the essential garbage collection function.

The following description of the data base management module will discuss the design choices that were made, as well as possible alternative approaches. The description starts by discussing briefly the structure of the data base itself and the

messages which are used to update it. The major design choices for the dynamic treatment of the data base are then examined, and lastly, we give a full description of the module as currently implemented.

2.1 Data Base Structure

The basic element in the data base is a line entry. A line in this context is the unidirectional trunk between two IMPs. The reverse direction of a trunk is considered a separate line entry. Thus each line has two end points, one of which can be uniquely designated as the source and the other as the destination. Associated with each line entry is the information needed by the SPF computation, such as the delay over the line.

These line entries are grouped into blocks according to their source IMP numbers, and the blocks are arranged in order of ascending source IMP number. This is the most logical grouping for the purpose of the kind of searching that is done during routing processing. It also allows for a more compact representation of the line entries, since the source IMP number does not have to be kept for each line entry. Instead, a table, indexed by source IMP number, is kept that contains indexes into the table of line entries. The index for each IMP points to the first entry in the block of line entries for which that IMP is the source. The individual line entries contain the destination IMP number for the line they represent as well as other information associated with the line. The table of line entries is called LTB and the table of indexes into it is called NTB.

One useful consequence of this structure is that the number of line entries for a site can be computed by subtracting its index from that of the IMP number one greater than it. In the original specification of the SPF data structure, this information was explicitly carried in a separate connectivity table. Another feature to notice is that nodes which are not on the network do not have to take up any room in the LTB table, since the NTB table contains the information that a node has no line entries. The indexes for the site not on the network and the IMP one greater than it will be the same, indicating that the site has zero lines. Note in particular that the tables may be initialized by the simple procedure of clearing call indexes to zero so that all nodes appear to have no lines.

When an IMP is started, the data base is built up gradually as routing updates are received. In the course of normal operation, updates can also cause alternations in the structure of the data base. These updates are generated at each IMP in response to changing network conditions. They are circulated throughout the network by the flooding transmission mechanisms described in Chapter 4 of the second Semiannual Technical Report. An update message contains the IMP number of the node which generated it and an entry for every line for which the originating IMP is the source. These line entries are of the same format as the ones in the LTB table. As we will see later,

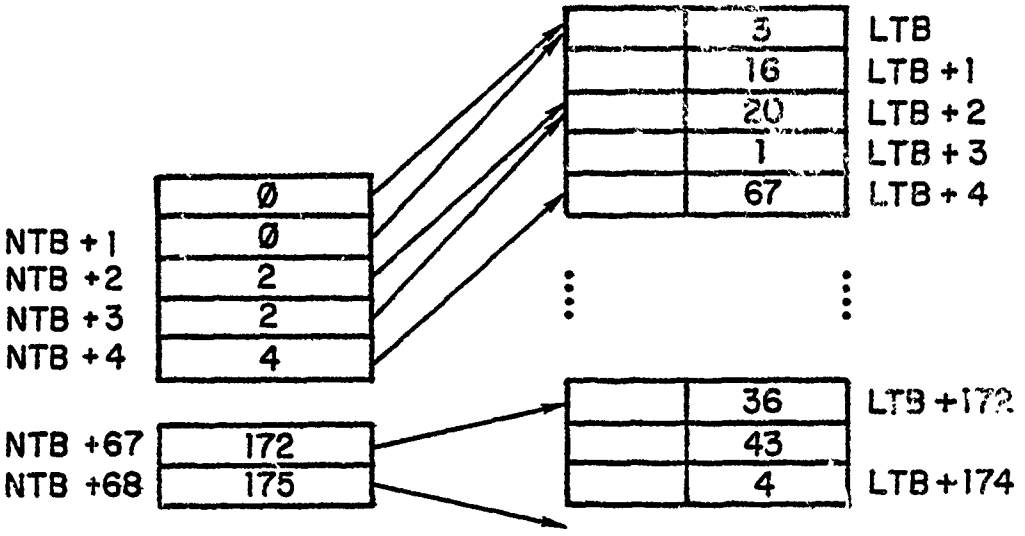
it is an important requirement of the data base management module that each update from a particular source IMP contain a line entry for each of that IMP's lines.

2.2 Requirements for the Data Base Management Module

The SPF algorithm makes some implicit assumptions about the structure and consistency of the topology data base. Insuring that the dynamically changing data base conforms to them at all times has been a major source of complexity in the management module. Some of these assumptions are discussed below.

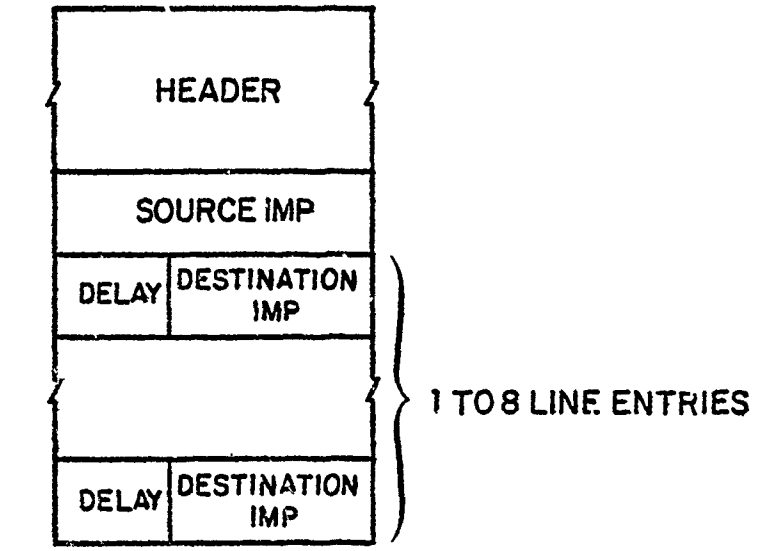
2.2.1 New line entries must be detected

The module that processes a line update assumes that an entry for that line already exists in the data base. The first step is to compare the current entry against the previously received one in order to determine if there is a change being reported. If a change is detected, the signed amount of change is computed and used to determine which routes should change. It is possible, however, to receive an update about an entirely new line that has no entry in the data base. The obvious instance of this is when a site is reconfigured, that is, when the number of modems at a site is decreased or increased. The simple reconnection of IMPs will also cause new lines to appear. During a reconnection, the number of modems on an IMP does not change, but the neighbors to which its lines are connected change. Since



DATA BASE TABLE STRUCTURE

NOTE: THERE IS NO IMP Ø OR IMP2 ON THE NETWORK



UPDATE MESSAGE FORMAT

Figure 2-1

a line is defined in our data base by its endpoints, without any reference to the modem numbers involved, connecting an IMP to another IMP, with which it previously had no connection, will generate a new line that must be added to the data base.

It is necessary, therefore, for new entries to be detected and dealt with in some manner. We have chosen to create an entry in the data base for the new line which represents the line as having been dead. This allows the processing module to then handle the new update in the same manner as any other update which reports that a previously dead line has come up.

2.2.2 All line entries must be paired

In order for the SPF processing module to function properly, every line entry in the data base must have a corresponding line entry describing the reverse direction. For example, if the data base has an entry in IMP 1's block for a line from IMP 1 to IMP 2, there must be one in IMP 2's block for the line from IMP 2 to IMP 1. Since any given update message can only report on one direction of the line, this requirement is not necessarily met. It is quite possible for an update with a new line entry to arrive and for there to be no entry in the data base for the reverse direction of this line. Obviously, one direction of the line must be processed before the other.

Rather than rewriting the processing module to tolerate an unpaired line entry, we decided to check for this condition before processing the line entry. That is, after checking to see that a particular line entry exists in the data base, we simply do the same check again with the end points of the line reversed. If the reverse entry does not exist, it will be treated in the same manner as a new entry and a dead line entry created for it, insuring that the original line now has its twin in the data base. (It will be seen later, when we discuss garbage collection, that this requirement that all lines be paired provides further complication of the data base management routines.)

2.2.3 Detection of implicitly dead lines

It is possible for lines to go dead in the network, and possibly even be removed entirely, without some IMPs receiving any updates that declare them dead. A network partition during which some IMP has its neighbors changed provides a situation in which this can happen. Imagine that a segment of the network containing IMPs 1, 2 and 3 is isolated from the rest of the network. During the isolation, IMP 1 has one of its lines disconnected from its usual neighbor, IMP 2, and reconnected to a different one, IMP 3. The IMPs in the isolated segment first receive an update declaring IMP 1's line to IMP 2 dead. They also should receive an update from IMP 2. Later they will

receive a new update from IMP 1 announcing its new line to IMP 3. Meanwhile, the rest of the network IMPs do not receive these updates because they are partitioned from IMP 1. They still believe IMP 1 has a live line to IMP 2. When the partition ends and IMP 1 next sends an update, all the IMPs that could not see it before will create a new entry for the line from IMP 1 to IMP 3. However, they will still believe there is also a line from IMP 1 to IMP 2, and they will receive no update that will tell them to declare it dead. Similar situations can arise when IMPs crash and change neighbors before coming up to the network.

It is possible, however, to detect these lines that should be declared dead. If every IMP follows the rule that it report on all its live lines in every update, then any entries in the data base which are not in the latest update message must be for lines that are now dead. This means that before processing an update message, the data base management must take each live entry in its block for the source of the update and search for it among the entries in the message. If it fails to find a match, it must invoke the SPF computation with an update entry declaring this line dead, thus simulating the effect of an actual update declaring that line to be down.

In the above example, when the partition is ended and IMP 1 sends its first update, all IMPs will check its contents against their data bases. Those IMPs which were isolated from IMP 1 will

detect that there is no entry in the update with a destination of IMP 2. They will call SPF with an entry showing the line from IMP 1 to IMP 2 dead and then proceed to process the rest of the update normally.

Note that with this procedure in effect, routing updates need never contain entries for lines that have gone down, since the absence of any entry for a dead line has the same effect as an entry which explicitly declares the line to be dead.

2.3 Dynamic Treatment of the Data Base

2.3.1 Expanding the data base

In order for the data base to adapt to changing topology, it must be possible for it to expand to accept new entries. Since our table structure requires that a new entry be inserted into the block for its source IMP, it is necessary to shuffle all the blocks above it up one slot and adjust all the appropriate indexes.

Although this is a time-consuming procedure, the only way to avoid ever having to expand the data base would be to allocate the maximum number of line entry slots to each IMP. Allocating less than this would not allow for the addition of extra lines at a site or the reassignment of an IMP number to a new site. Even allocating the maximum number of slots will avoid the need for dynamically expanding the data base only if dead line entries can be reused for new line entries. As will be seen later, the SPF routine does not guarantee this. More importantly, this scheme is extremely wasteful of table storage. Since the average ARPANET connectivity is about 2.5, and the maximum number of lines at a site is currently 5 on Pluribus IMPs, it would require twice as many line entry slots as are really needed.

2.3.2 Garbage Collection

If the data base is allowed to expand and to accommodate new entries, some mechanism must be implemented that garbage collects dead line entries. Otherwise, the steady stream of network reconfigurations will cause the data base to grow until it exceeds the space allocated to it and the IMP would have to restart. The data base will fill up with defunct entries unless some method of reusing that space is implemented.

Many choices are available in choosing a garbage collection strategy, but all strategies follow the same basic operation by reusing the space occupied by dead line entries. However, as discussed in section 2.2.2, a line must always have its reverse direction in the data base. This means that a dead line can be removed from the data base only if its reverse direction entry is also dead and also removed at the same time. It should be noted that lines are frequently found to be dead in only one direction in the data base. If an IMP crashes, the neighbors around it report their lines to it as dead. But no updates are generated by the dead IMP, so its lines in the data base remain live, though the SPF computation will not use them since the node is unreachable. Similar situations arise during network partitions.

It is necessary, therefore, to construct a routine that searches for pairs of dead lines and removes them in some fashion

from the data base. A choice arises at this point in how to dispose of the collected entries. The most complicated would be to compress the data base, completely removing the two entries which are being reclaimed. This would make space for two more entries available for use anywhere in the data base.

A second alternative would be to simply mark the entries as unused and allow new line entries to use them, rather than to cause expansion of the data base. This does not require the extra program code to contract the data base, but it also does not make the reclaimed space generally available. Instead, it only provides free slots in the same blocks in which the previously dead entries were situated. If new entries arrive from other source IMPs, and these source IMPs have no free entries in their blocks, it will be necessary to expand the data base to accommodate these other entries even though there are already some unused entries. Thus it is possible for the data base to completely fill up even though there are many unused slots in it. If no routine for contracting the tables is available, then the IMP will have to restart. This could happen in instances of major retrunking where the number of lines at various sites changes. If only the connections between sites change, with the number of lines at most sites staying constant, then this scheme would be sufficient to keep the data base from overflowing as long as the garbage collection is done between the

time that the old line goes dead and the time that the new one appears. This method does have the advantage of avoiding expanding or contracting the data base, both of which are lengthy processes that lock out the rest of the IMP while they run.

We now come to the question of when to invoke garbage collection procedures. Four possibilities present themselves. Dead line entries can be removed from the data base as soon as they are processed by SPF. Alternatively, the arrival of new entries can prompt the search for slots that they can reuse. It is also reasonable to construct some periodic process that garbage collects the data base when no routing is being processed. Lastly, we can wait until the table reaches its maximum size and then shrink the data base back to its minimum size. Each of these approaches will be explored and their advantages and disadvantages discussed.

It is certainly possible, upon completion of processing an update that declares that a line has died, to check the line's reverse direction, and if it is also dead, to remove both entries from the data base. This test would fail in more than half the cases, since one direction of a line must necessarily be processed before the other. In some instances, as shown in section 2.2.3, the reverse direction of the line cannot be declared dead. The major disadvantage of this approach, though, is that it would mostly result in unnecessary work. The vast

majority of cases where a line goes dead involve no retrunking. The dead line will reappear as soon as it comes back up, perhaps in no more than a minute. If the original entries for the line are garbage collected as soon as they are both marked dead, then the updates that later herald the reappearance of the line will have to be treated as new entries requiring special action. In other words, the IMP would be spending its time unnecessarily removing and reinstating the same entries.

A better scheme is to have the arrival of a new entry prompt a search for a reusable dead one. If such an entry is found in the block of lines for the source of the new entry, it can immediately be "usurped". The reverse direction of the reused entry can either be marked unused or its space compressed out of the data base. This approach is attractive for two reasons. First, since most new entries result from a reconnection of sites in the network and not actual reconfigurations, it is likely that a usurpable entry will be found for the current update. The previously existing line will have been declared dead already, in most cases in both directions. Secondly, this method allows garbage collected entries to be reused immediately, thus avoiding the costly compression of the data base at the same time as avoiding the need to expand the data base for the new entry. However, if this is the only garbage collection mechanism used, it is still

possible for the data base to overflow even though there is unclaimed reusable space in it. Since only the block of entries where the new entry must be inserted is searched, usurpable entries in other blocks will not be reclaimed. The success of this approach thus depends on a correlation between new entries and usurpable ones.

A periodic garbage collection mechanism is yet another possibility. It would run when there is no routing work to do and might search only part of the data base at a time, limiting the amount of processing done in any one period. The primary problem with this method focuses on what to do with the entries that can be removed. As we discussed above, merely marking them unused does not insure that the data base will not overflow. To be effective, this method would have to compress the data base as it removes entries. One pass could be made that identifies and marks all removable entries, and then a second pass could delete all marked slots. The problem with this is that every new entry that arrives will have to expand the data base so it can be inserted. This manipulation takes many more instructions than the job of finding a usurpable entry. The periodic approach has the further inefficiency of removing lines that are only momentarily dead.

The last approach we have considered does garbage collection only when the tables overflow. It would function in the same

basic manner as the periodic mechanism. There are two advantages to this method. First, no periodic scheduling mechanism needs to be constructed. Second, since it will be rarely invoked, most lines which momentarily die in the network will not be garbage collected. Its main disadvantage is that when it is invoked, it must lock out the rest of the IMP while the lengthy compression is performed. Such a transient may have global network effects. This scheme also carries the disadvantage of requiring all new entries to expand the data base.

We have chosen to adopt the method of garbage collecting when a new entry arrives. The full scheme will be presented in the next section, but a brief description is given here. When a new entry arrives we attempt to usurp a dead line for it. If a usurpable pair is found, the reverse direction line is marked unused, and the forward direction is used for the new entry. If a usurpable or unused entry is not available, then the data base is expanded. This scheme is admittedly imperfect, but we feel it is a reasonable compromise given the present extreme memory constraints in the Honeywell IMP. It is possible with this scheme to overflow the data base, at which time the IMP will have no choice but to restart. We feel it is unlikely that this will occur for two reasons. First, it is believed that most new lines represent reconnections and not reconfigurations. This means that usurping should work most of the time. Second, we have

provided some slack entries in the data base to provide for the relatively rare reconfigurations. Also remember that when an IMP restarts for any reason such as power failures or after maintenance, its SPF data base starts out completely compressed. This should provide for an eventual compression in the data bases around the network. The degree to which our garbage collection scheme keeps the data base from growing will have to be watched in the first few months that SPF runs in the network. Only real experience with the network will show if it is sufficient.

2.4 Specification of the Data Base Management Module

2.4.1 Detect lines that have died

The first step in processing an update message is to look for lines that are not reported. Take each live line in the data base block for the IMP that is the source of the update and try to find a corresponding entry in the update message. If a match is found, continue with the next data base entry. If no match can be made, create a dummy update entry for the missing line that shows it to be dead. Call the SPF processing module with this entry. Upon return, continue searching the current data base entries. When finished with this procedure, iterate through the update message entries, calling SPF for each one. When the entire packet has been completed, it can then be marked processed and another message started.

2.4.2 Insure that entries exist in the data base

Before the SPF module can begin to process an update entry, a check must be made to insure that entries exist in the data base for it and its reverse direction. SPF will make a call to the data base management routine, FNDENT, first for the reverse direction line and then for the line that it is about to process. FNDENT returns the index of the line entry currently in the data base. Now the SPF computation can proceed since it can be sure that a pair of entries exist in the data base.

2.4.3 FNDENT

This routine performs all the manipulations on the data base. It is called to locate an entry for a line, and if one cannot be found, to create one. As a byproduct, it also performs garbage collection. It requires two arguments, the source and the destination IMP numbers for the line it must find.

The first step is to search the source IMP's block for the appropriate line entry. If such an entry is found, the line already exists in the data base and FNDENT exits. This is the result in almost all cases.

If the search fails to find an entry, the line must be new. FNDENT must now create a new entry in the data base or usurp an old one. It searches the source's entries again, looking for dead lines. If it finds one, it checks the destination number. If it is zero, this is an unused entry which can be immediately usurped. It does not have a reverse direction. The new destination number is written into the entry and FNDENT exits. If the dead entry has a real IMP number in its destination field, it must be further checked to see if it can be usurped. A dead line can be usurped only if its reverse direction is also dead. If this check succeeds, the dead line and its twin are usurped. First mark the reverse entry unused by zeroing its destination number. Then change the destination number of the forward

direction entry to the new line's destination. FNDENT can then exit. If the reverse direction of the candidate for usurping is not dead, this entry cannot be touched. The search for dead entries in the source IMP's block must continue until a useable entry is found or the end of the block reached. If no usurpable entries are found, the data base must be expanded.

The data base is expanded by the following algorithm. Save the index of the first line entry for the IMP whose number is one greater than that of the source IMP. This will be the slot where the new entry will go. Check that there is room for one more entry in the data base. If not, restart the IMP immediately. If there is room, lock out the retransmission generation routine which also uses the data base. Increment by one the indexes for all IMPs greater than the source IMP. Starting with the last line entry, move each entry up one slot, making sure to copy all information associated with the entry, including the SPF tree flags. Stop when the slot whose index was saved has been copied. Initialize the new slot to be dead, not in the tree, and to have the destination IMP number of the new line. Unlock the data base tables and exit FNDENT.

3. TESTING THE NEW ROUTING SCHEME -- GOALS

In a distributed packet switching network, there are many aspects of the network's design which affect its performance. None is more important, however, than the routing scheme. It is the routing scheme which has the major responsibility for ensuring that packets get delivered to their destination in as timely a manner as possible. If the routing scheme performs badly, packets may never reach their destinations at all, even if there is a free and clear path to the destination. Packets may be sent into areas of congestion, even if there is a path around the congestion. Packets may be sent on a long-delay path, even if a short-delay path is available. Packets may be routed to dead lines or dead IMPs. Because of the importance of routing to the general performance of the network, it is not desirable to change the routing scheme of an operational network (such as the ARPANET) without first putting the new routing scheme through an extensive series of tests. This is no simple matter. Any routing scheme will have several different modules, each of which must be tested separately from the others and also jointly with the others. Measurement tools must be designed and implemented. Measurement testbeds must be developed. A series of milestones must be planned, so that the testing can proceed in an orderly and systematic manner. In this chapter we will discuss the thinking behind our testing procedures. In the next chapters we

will discuss our testing techniques and tools, and will present some of the results of our testing.

Any distributed, adaptive routing scheme can be thought of as having five separate components -- a measurement process, an updating protocol, a "shortest-path" computation (the quotes are used because the definition of "short" is relative to a metric which may bear no relation to any intuitive notion of shortness), a procedure for managing the data base used by the shortest-path computation, and a procedure for forwarding packets on the basis of the output of the shortest-path computation. It should be noted that the term "routing algorithm" has been avoided here, because it has been used ambiguously in the past, sometimes referring to the entire routing scheme, sometimes referring only to the shortest-path computation. To prevent confusion, the term "routing algorithm" will not be used here at all. In order for a routing scheme to perform well, each of its components must perform well, and in addition they must perform well jointly. In trying to develop a set of testing procedures for the routing scheme, each component offers a different set of problems; testing their joint operation offers more problems still. Some of these problems will be briefly discussed in the following paragraphs.

Both the new and the old ARPANET routing schemes are single-path schemes. That is, the shortest-path computation in each scheme defines one and only one path between any given pair of IMPs. The forwarding procedure for single-path schemes is quite simple. The shortest-path computation generates a table indexed by destination IMP whose values specify the next inter-IMP trunk to use for each destination. The actual decision as to which trunk to forward a packet on is made by a simple table look-up. Because the new routing scheme uses exactly the same forwarding procedure as the old, no explicit testing or evaluation of the forwarding procedure is necessary. However, it is worth pointing out that other sorts of routing schemes might require non-trivial forwarding procedures for which extensive testing would be necessary. Consider, for example, a multi-path routing scheme. In such a scheme, the shortest-path computation would define several paths to a given destination and would specify what fraction of the traffic to that destination is to flow over each path. In such a scheme, the decision as to which trunk to transmit a particular packet on could not be made by a simple table look-up. Rather, making this decision could require a computation which might have to be made for every packet. The effects of having to perform the computation would have to be carefully considered and tested for. In addition, the forwarding procedure of a multi-path routing scheme could well defeat the advantages of such a scheme. The shortest-path computation might

say to divide the traffic to a given destination over three paths in the ratio 0.17856, 0.25384, and 0.56760. However, it is unlikely that a forwarding procedure could effect such a division of the traffic. If not, then the routing scheme might not perform nearly as well as expected, and extensive testing would be necessary to determine whether its performance was satisfactory. However, since the forwarding procedure in the ARPANET is simple, and remains unchanged, it shall not be considered any further.

The measurement process used in the new routing scheme, however, is quite different from the measurement process of the old routing scheme. It is easy to see why any adaptive routing scheme must include some sort of measurement process. In any adaptive routing scheme, the output of the shortest-path computation is supposed to be sensitive to the state of the network. In other words, state information about the network must be provided as input to the computation. So there must be some sort of process which determines, at any given time, the actual values of the state information to be input to the computation at that time. This process may be dubbed a measurement process. The measurement process of the old routing scheme is quite simple. Every so often, an IMP simply notes how many packets are queued to each of its lines at that instant. These queue length measurements serve as the input to the old

shortest-path computation. The measurement process in the new routing scheme is much more complex. The average delay per packet on each network line is actually computed. (For a detailed discussion of the measurement process used on the new routing scheme, the reader is referred to our first two semiannual reports.) In order to test the performance of the new measurement process in isolation from the other components of the new routing scheme, we developed the following methodology. We actually implemented the measurement process in the IMP. However, the results of the measurement, rather then being input to a shortest-path computation, were sent to an PDP-10 TENEX system for inspection and analysis. Our second semiannual report discusses the results of an extensive series of tests made in this way. Suffice it to say that the measurement process had been quite throughly tested in isolation before being combined with the other components of the routing scheme.

It is important to realize, though, that a measurement process which performs well in isolation may not perform well when combined with the other components of the routing scheme. When the measurement process is run in combination with the other components, there are feedback effects which are not present when it is run in isolation. The inputs to the measurement process are the actual delays of packets flowing over a line. The output of the measurement process is the average delay per packet in

that line. These average delays are the input to the shortest-path computation, whose outputs are the inputs to the forwarding process. For a particular matrix of offered traffic, the forwarding process determines how much traffic flows over each line, and it is this fact that determines the packet delays on each line. (The word "determines" is being used here in the sense of "partially determines", rather than "wholly determines.") So the output of the measurement process at one time partially determines its inputs at a later time, i.e., there is a feedback effect. In our second semiannual report we present some mathematical analysis which shows that under certain idealized conditions, these feedback effects can cause an instability in the routing which could make the new routing scheme perform very badly. One of our major testing goals has been to determine whether any such undesirable feedback effects exist in the operational environment of the ARPANET.

The measurement process is the part of the routing scheme which is responsible for detecting congestion in the network. That is, when congestion exists on a particular network line, the output of the measurement process should be such as to cause packets to be routed around that line. It is known that the measurement process in the old routing scheme could not detect congestion, and that the old routing would often send packets into a congested area, thereby making the congestion worse. The

new routing's measurement process has been specifically designed so as to be able to detect congestion. However, to tell whether it actually fulfills this design goal, it is necessary to run all the components of the routing scheme together. One of our testing goals has been to determine whether the new routing scheme really is better at avoiding congestion than the old.

Another aspect of the measurement process (for both the new and the old routing schemes) is the procedure for determining whether a line is up or down. The up/down status of each line in the network is a very important input to the shortest-path computation. As discussed in our previous two semiannual reports, the line up/down protocol has recently been changed in order to make it provide a more meaningful indication of the usefulness of the line. The way in which this part of the routing scheme has been tested in isolation from the other components of the routing scheme has been fully described in our previous reports. It is worth noting that the line up/down protocol, unlike the other parts of the measurement process, is not subject to feedback effects. The decision as to whether to regard a line as up or down is made on the basis of the error rate of special protocol packets which are always sent on the line periodically. This determination is independent of the rate of flow of ordinary traffic on the line, so it is independent of the results of the shortest-path computation. Since the line

up/down protocol is not subject to feedback effects due to the other components of the routing scheme, it can be fully tested in isolation. That is, there is nothing additional to be learned by testing it in combination with the other components.

Exactly the reverse is true of the updating protocol. It can only be given a meaningful test in combination with the other components of the routing scheme. The updating protocol of the new routing scheme is not only totally different from the updating protocol of the old routing scheme; it is different from any other data transmission scheme found in the ARPANET. The details of the updating protocol and the rationale for its existence can be found in our previous semiannual reports. However, in order to formulate our testing goals with respect to the updating protocol, it is worthwhile to discuss briefly the role that the updating protocol plays within the routing scheme. Recall that the input to the shortest-path computation of any distributed adaptive routing scheme consists of state information about the lines in the network, which may be called the data base of the routing scheme. This state information is gathered by a measurement process, as discussed above. The state of a particular line can, of course, be directly measured only by the IMP which transmits over that line. This gives rise to the following problem: How can each node gain access to the entire distributed data base? This problem can be solved in two

different ways. One way is to distribute the shortest-path computation itself so that each piece of the computation has direct access to the part of the data base that it needs. This is the approach taken by the old routing scheme. The shortest-path computation of the old routing scheme has two sets of inputs. One input is the locally measured line state information. The other input is the output of the shortest-path computation at neighboring nodes. This approach requires each node to send its immediate neighbors the results of its own shortest-path computation. Point-to-point communication between a pair of neighboring nodes does not offer much of a protocol problem, and the updating protocol of the old routing scheme is very simple. Despite its apparent simplicity, however, there are serious problems in any attempt to distribute the shortest-path computation. In the old routing scheme, there is no functional relation between the routing data base at one time and the output of the shortest-path computation at that time. That is, the output of the computation depends not only on the state of the lines around the network, but also on the history of the computation, and the order in which certain events occur around the network. It is this fact which gives rise to many of the problems of the old routing scheme (such as looping and slowness to react to changes) which we have discussed in detail in previous reports.

In order to ensure that the output of the shortest-path computation at a given time is a function only of the state of the routing data base at that time, we decided that the new routing scheme should not have a distributed computation. This means we had to take an alternative approach to solving the problem of the distributed data base. The alternative was to develop a quick and reliable updating protocol for transmitting changes in the data base to all nodes in the network. This makes the entire distributed data base (i.e. the output from each of the local measurement processes) locally available to each IMP, enabling each IMP to maintain a complete copy of the entire data base. This permits a purely local shortest-path computation, so that there is a deterministic relation between the data base and the result of the computation, thereby avoiding many of the problems inherent in the old routing scheme. It must be pointed out, however, that the ability of the new routing scheme to avoid such problems is dependent on the updating protocol's really being quick and reliable. (A full discussion of such issues is presented in our second semiannual report.) Our updating protocol was specifically designed to ensure quick and reliable updating under all conceivable network conditions; one of our major testing goals has been to determine whether the updating protocol really meets its design goals. The only real means of determining this is to run the entire routing scheme under operational conditions, while monitoring the updating protocol to

see whether it does or does not get the routing updates around the network in a sufficiently timely and reliable manner. There is no way to get significant results by testing the updating protocol in isolation from other components of the routing scheme.

Another component of the routing scheme which is closely related to the updating protocol is the data base management procedure (discussed in detail in Chapter 2). This is the procedure that receives the routing updates from the updating protocol and uses them to build tables which are suitable as input to the shortest-path computation. The sorts of problems which the data base management procedure gives rise to are not very subtle. If the tables are not built correctly, the shortest-path computation will probably either halt, or else go into an infinite loop, giving immediate feedback as to the nature of the problems. Problems with other components of the routing scheme, however, are more likely to result in poor or incorrect routing, a condition which is much more difficult to test for than an infinite loop or a halt. The main problem of any procedure which builds tables from updates is that any arbitrary combination of updates may arrive in any arbitrary order, which means that the procedure must be completely free of order dependencies. It is almost impossible for a programmer to debug such a procedure without testing it out in a fully operational

environment (i.e. in combination with the other components of the routing scheme), since only in such an environment can one expect to see enough different combinations of events. One of our major testing goals was to run the new routing scheme under a wide enough variety of conditions to be able to gain confidence in the data base management procedures.

The final component of the routing scheme is the shortest-path computation. It may seem odd to discuss this component last, since most discussions of routing tend to concentrate primarily (if not exclusively) on the shortest-path computation. Nevertheless, it is the component which is most amenable to isolated testing in the absence of the other components of the routing scheme. The SPF computation has been extensively tested as a stand-alone program on a TENEX system, as well as a Honeywell 316 and a Pluribus. These tests, made with the use of a test data generator, were carried out even before any design work had been done on the other components of the new routing schemes. Since then we have run the SPF computations many times in combination with the other components of the new routing scheme. In all that testing, no problem with the SPF computation has ever been discovered. Apparently, all problems were discovered in the isolated testing.

4. TESTING THE NEW ROUTING SCHEME -- TECHNIQUES

In the previous section, we emphasized the need for testing the new routing scheme in the operational environment of the ARPANET. However, testing new software in a distributed network is a complicated procedure, involving different problems than, say, testing a new operating system for a single computer. To test an operating system in a computer which supports a user community, it is necessary to schedule some down-time, during which users are prevented from accessing the computer. The new operating system is loaded and put through a series of tests. If it halts, or goes into an infinite loop, the operator can regain control of the computer from the console. At the end of the testing period it is a simple matter to reload the old system. Testing new software in the ARPANET, however, is nowhere near so simple. For one thing, we are not allowed to schedule "down-time" in the ARPANET. We were able to schedule software testing periods during the early morning hours. During these periods we were permitted to disrupt the network, in the sense of letting the network run in a less reliable manner than usual. But we were requested to keep the network accessible to users as much as possible. For another thing, most of the ARPANET's IMPs run unattended. Should an IMP halt, or go into an infinite loop, there is no operator present to regain control. To be sure, the Network Control Center (NCC) at BBN has an extensive set of

facilities for controlling unattended IMPs at remote sites. However, these facilities all make use of the network itself. An IMP which has halted, or which is running in a tight loop, will not respond to commands from the NCC; such IMPs are out of the NCC's control. Furthermore, any software problem which causes the network as a whole to fail or run in a degraded condition can cause the network to be non-responsive to the NCC's commands. Because the ARPANET has been designed for operational robustness, there are very few problems which can cause the network to fail as a whole. Unfortunately, failure of the routing scheme is among these few problems. If the routing scheme fails there may be no way for commands to get from the NCC to the IMP. Lastly, loading the entire network with the new routing scheme is a much more complicated operation than merely loading a single computer with a new operating system. One cannot simply load the new routing into the IMPs one by one, until all the IMPs have it. That would mean that during some period of time, some IMPs were using the new routing scheme while others were using the old; in general, the network will not run properly unless all IMPs are using the same routing scheme. Thus even the process of loading the new routing scheme into the net can cause problems if it is not done carefully.

A partial solution to these problems is to do a lot of testing in the lab before doing any testing in the field (the

field being the actual ARPANET). In fact we did do a great deal of testing in the lab, and we never tried anything in the ARPANET without trying it in the lab first. Our lab resources, however, are not well-suited for the testing of a new routing scheme. The lab contains only two Honeywell 316s for use in testing IMP software. This may have been sufficient for testing most of the changes which have been made over the years to the IMP software, but it is not sufficient for testing a routing scheme. It should be obvious that the ability of a routing scheme to run properly in a two-node network has little bearing on its ability to run in a large distributed network. Such a trivial network just does not give rise to the sort of problems which apply stress to a routing scheme, since the routing problem in a two-node network is completely trivial. Fortunately, we were able for a few months to collect a total of four Honeywell 316s in our lab. From the perspective of routing, a four-node network is significantly more complex than a two-node network, and we were able to do quite a bit of testing in the lab network. Nevertheless, a four-node network is much simpler than the ARPANET, and we would not expect, a priori, that the result of lab testing would be the same as the result of similar tests done in the field. So we still had to develop techniques for doing field testing that would minimize the possibility of major disruptions of the ARPANET due to failure of the new routing scheme.

One should not get the impression that the NCC is completely helpless if some major problem does arise during testing of the routing scheme. A small number of operational IMPs are located on BBN's premises, and these can be controlled from their consoles. There is a procedure known as "demand reload" by which one IMP can forcibly reload its neighbors, even if the neighbors are not communicating with the NCC. Thus if any major problems arise in the testing of the routing scheme, a good release can be easily loaded in BBN's local IMPs, and these IMPs can forcibly reload their neighbors, one at a time, until all the IMPs in the network have the good release. In fact, we did have to use this procedure on two occasions to restore the network to operating condition after problems developed during our testing of the new routing scheme. However, though the procedure is an effective way of recovering from problems, it is really something that should be used only as a last resort, not something that should become a part of our everyday testing procedures. It was therefore incumbent upon us to develop testing procedures which, as much as possible, minimized the chances of a major problem occurring during our tests of the rew routing scheme.

We decided to use a testing procedure similar to the one we developed for testing the new line up/down protocol. Recall that in order to test the line up/down protocol, we created an IMP release that ran both protocols at once, in parallel. However,

at any given time, in any given IMP, one of the protocols was running in "controlling mode", and the other was running in "phantom mode." That is, both protocols were always running, exchanging their own special protocol packets over the line, and coming independently to a decision as to whether the line should be declared up or down. The actual operational up/down status of the line, however, was affected only by the decisions of the protocol which was running in controlling mode. The decisions of the protocol running in phantom mode were reported to the NCC for later analysis, but they had no operational effect on the line's up/down status. In addition, the NCC had the capability of switching the protocols from one mode to another in order to test the new protocol under a variety of conditions.

We adopted a similar, though more complex, approach to the testing of the new routing scheme. That is, we prepared an IMP software release which contained all the code for both routing schemes. This release has five different routing states, each corresponding to a particular degree of parallelism. The five states are the following:

I) In state I, the old routing scheme is in controlling mode, and most of the new routing scheme is deactivated. The only active component of the new routing scheme is the measurement process. However, even though the packet delays are always being measured, routing updates are never generated, so

none of the other components of the routing scheme have any work to do. This is the state in which the IMPs run when we are not doing any testing.

II) State II is similar to state I, except that a little bit more of the new routing scheme is activated in the phantom mode. The measurement process causes routing updates to be generated; and enough of the updating protocol is activated to ensure that all the updates are sent to all the IMP's in the network. However, the reliable transmission aspects of the updating protocol are not activated. In this state, the data base management procedure and the shortest path computation do not run at all; routing updates created by the new routing scheme are simply discarded, instead of being processed.

III) In state III, the old routing scheme is still run in the controlling mode, but the new routing scheme is fully activated in the phantom mode. That is, all aspects of the new routing scheme are operating just as if the old routing scheme were not there. However, all ordinary user packets are routed as specified by the old routing scheme, and the results of the new scheme do not have any effect on the operation of the network. It is possible in this state to flag certain test traffic so that it (but not other traffic) is routed according to the new scheme.

IV) In state IV, the new routing scheme runs in the controlling mode, and the old routing scheme runs in the phantom mode. All packets are routed according to the new scheme.

V) In state V, the new routing scheme runs in the controlling mode, and the old routing scheme is completely deactivated.

Each of these states has a greater impact on the network's operation than the states preceding it. This makes the states progressively more dangerous, in that problems arising when the network is running in the later states are likely to have a worse effect, and to be harder to recover from, than are problems arising when the network is running in the earlier states. So we attempted to do as much testing as possible in the earlier states before proceeding to the later ones. Each of the states is useful for some kind of testing, but not for others. (State I, of course, is not used for testing at all, but only for normal network operation.) In state III, it is possible to test the updating protocol, the data base management procedure, and the shortest path computation to see how they perform together. However, while the new routing scheme is running in the phantom mode, there is no feedback between the measurement process and the shortest-path computations, since packets are not routed according to the new routing scheme. We can generate some imperfect feedback by sending large amounts of special test

traffic, and flagging that traffic so it will be routed according to the new scheme. If the amount of test traffic is much greater than the amount of ordinary traffic, state III becomes indistinguishable from state IV. To do a full test of the routing scheme, though, with all feedback mechanisms engaged, it is necessary to go to state IV.

State II has no utility in and of itself, but is very useful when combined with state III. That is, one IMP can be put in state III, while all the rest are in state II. The effect of this is to use the network as a test data generator while performing a state III test in a single IMP. This does not yield as thorough a test as would be obtained by placing the entire network in state III; it is particularly useful, though, if one is afraid that a bug in the new routing code will cause the IMP to halt or loop under actual network conditions.

There are two major reasons why it is important to test the new routing scheme in state V. One reason has to do with the distorting effect the old routing scheme, by its mere presence, may have on the packet delays in the network. In our first semiannual report we discussed the spikiness and high variability of the packet delays under what would appear to be steady-state conditions. We argued that some of this high variability may be due to the presence of the old routing scheme. The old routing scheme periodically causes long routing update packets to be

transmitted over the lines, thereby causing spikes in the queuing delay seen by packets. Also, the old shortest-path computation runs periodically and takes a large number of processor cycles, causing spikes in the processing delay seen by packets. These effects are no less present and no less significant when the old routing scheme is in phantom mode than when it is in controlling mode. It is possible that when the old routing scheme is fully deactivated, the characteristics of the packet delays will be very different. Since the new routing scheme actually measures the packet delays, any significant change in the characteristics of the packet delays could have a significant effect on the performance of the new routing scheme. That is, the new routing scheme may perform differently when the old routing scheme is fully deactivated (state V) then it does when the old routing scheme is running in the phantom mode (state IV). This makes it important to test the new routing scheme in state V.

The other reason for testing the new routing scheme in state V has to do with the integrity of the IMP program itself. Removing one routing scheme from the IMP and replacing it with another is not a simple job. The IMP program is a ten-year-old highly optimized program which has been under continuous development. It is not implemented according to "structured programming"; little pieces of the old routing scheme are scattered around the IMP program. The task is further

complicated by the fact that other IMP functions have been piggybacked on various functions of the old routing scheme. For example, at one time the old routing updates doubled as the Hello packet of a line up/down protocols. For another example, at one time channel acknowledgements were sent periodically in null packets which were always transmitted immediately after the old routing updates. Before releasing the new routing scheme, therefore, it is important to demonstrate that the network can function with the old routing scheme totally deactivated. This can be demonstrated by testing in state V.

Operating a network which contains two routing schemes gives rise to a number of problems which we had to solve before we would do our testing. The most straightforward problem had to do with the limited amount of memory in the IMP. When the amount of code in the IMP program increases, it takes up space that would otherwise be available for packet buffers. In order to run our tests, we had to add code for the new routing scheme, while leaving in all the code for the old scheme. We also had to add a significant amount of code for instrumenting and measuring the performance of the new scheme. In addition, there is a significant amount of code required to implement the capability of switching among the five states described above. After adding all this code, the IMP had only 27 buffers left. We have in the past run the IMP with as few as 29 buffers without encountering

any problems; yet when we tried running it with 27 buffers, the network ran in a degraded fashion, producing long delays and low throughput which was quite noticeable at the user level. We were able to solve this problem by putting five buffers' worth of new routing code into a "package", which could be removed from the network whenever we did not need it for testing purposes. We still do not totally understand, though, why the difference between 29 buffers and 27 buffers makes such a big difference to the network performance.

Other operational problems arise when an attempt is made to switch between states. Neither routing scheme can be expected to perform properly unless it is started in a well-defined initial state. The only way to ensure the proper initialization is to restart the IMP whenever it is desired to activate a previously inactive routing scheme. Furthermore, when a routing scheme is deactivated it may have control of various scarce resources. If the IMP is to operate properly after a test period ends, the IMP must force the release of all scarce resources which were in use by the scheme which has been lately deactivated. Additional problems arise when we attempt to switch a routing scheme running in phantom mode to controlling mode. In order for the network to operate properly, all the network nodes must be under the control of the same routing scheme. This can be easily shown, as follows. Let nodes 0 and N be neighbors, and suppose that in

node O, the old routing scheme is in controlling mode, while in node N, the new routing scheme is in controlling mode. It is possible that, according to the old routing scheme, the best route from O to a destination node D is via N, while according to the new routing scheme, the best route from N to D is via O. Then packets for D may loop between O and N without ever being delivered to their destination. Since the two routing schemes operate independently, there is no way to detect and break this loop. Thus the network cannot be expected to operate properly unless the same routing scheme is in control of all the nodes.

The problem just described can occur even if the non-controlling routing scheme is running in the phantom mode. An even more serious problem can arise if the non-controlling scheme is fully deactivated. To see this, suppose the network is divided into two areas. In one area (the "old area"), the old routing scheme controls, and the new routing scheme is deactivated. In the other area (the "new area"), the new routing scheme controls, and the old is deactivated. Now if an IMP goes down or comes up in one area, the IMPs in the other area have no way of detecting that fact, since there is no flow of routing update information between the two areas. (Note that this would not be the case if the non-controlling routing scheme were running in the phantom mode.) Then the following situation can arise. Let O be an IMP in the old area, and let M be an IMP in

the new area. Suppose that M was initially down, so that O believes M to be unreachable. When M comes up, O will not be aware of the fact, and will still believe M to be unreachable. However, as long as O has a neighbor N which is in the new area, all the IMPs in the new area will believe that O is up. We now have a situation where M thinks O is up, but O thinks M is down. Suppose that M tries to establish a connection with O. It will send out control packets to O until it receives a reply from O. But since O thinks M is down, it will never reply to any of M's control packets. As a result, M will continue sending out control packets forever, without ever receiving any reply. This continuous and uncontrolled transmission of control packets can lead to network congestion. If routing loops have formed, as discussed in the previous paragraph, excessive re-transmissions of control packets can severely aggravate the problem; making the congestion much more severe than it would otherwise be.

We see then that when we want to put a non-controlling routing scheme into controlling mode, it is necessary to restart all the IMPs simultaneously. Otherwise it is impossible to ensure that the same routing scheme is in control of all the nodes at any given time. When we began our testing, the NCC did not have the capability to restart all the IMPs at once (though it did, of course, have the capability to restart the IMPs one at a time.) We had to develop such a capability especially for our

testing. This is just one of several improvements to the NCC capabilities that we developed as part of the routing contract.

The operational problems we have been discussing are neither particularly profound nor especially difficult to resolve. We have discussed them at such length because of the significant impact they made on our testing schedule. The technique of running the two routing schemes in parallel, and of having five different states (constituting varying degrees of parallelism), was developed in order to minimize the possibility that our testing would cause a widespread or long-lasting network disruption. We believe that we took a sound approach; but we wish to emphasize that safety comes at a price. Not only was the network software made more complex, but we had to develop new capabilities for the NCC and we also had to carefully refine our operational procedures. Furthermore, the vast majority of problems we encountered during our testing were operational in nature. That is, when a major problem did develop during our testing, it was much more likely to be due to a flaw in our procedures than to a problem in the new routing scheme.

5. TESTING THE NEW ROUTING SCHEME -- TOOLS

In oder to determine how well the new routing scheme was performing, we had to develop various instrumentation and measurement tools. In general, we opted to keep our measurement tools very simple. In our experience, when tools of great complexity and sophistication are used for measuring network performance, there is great difficulty in interpreting the data generated by their use. It is difficult to be sure that such tools are doing exactly what they are supposed to be doing, and it is extremely difficult to get such tools debugged. When one is attempting to measure a network's performance, one must be able to have confidence in the measurement tools and procedures, so that one can be sure that the measurement results really do represent particular states of the network, and are not simply artifacts of the tools. This argues for keeping the tools as simple and easy to understand as possible.

The most obvious means of testing the performance of a routing scheme is to create various offered traffic loads and then see how much of the traffic was routed over the various alternative paths. To do this we developed a tool known as the "tagged packet." A tagged packet is just an ordinary packet with a particular bit set in the host-IMP leader. However, the data field of a tagged packet is used to carry a trace of the packet's path through the network; instead of ordinary user data. When a

tagged packet is first submitted to the network, its data field does not contain any meaningful information. At each intermediate node that the packet passes through on the way to its destination, the packet is "tagged" with that node's IMP number, and with the delay that the packet experienced in traveling through that IMP. Thus when the packet arrives at its destination, it contains a precise record of the path it traversed, as well as its delay. The value of delay which appears in the packet tag for a particular node is the same value which is input to the measurement process at that node. (Actually, the delay which appears in the packet tags is not quite so precise as the delay which is really input to the measurement process. The latter is measured in units of 0.8 milliseconds, but the former is truncated to units of 6.4 ms., so that we can fit it into an eight-bit field.)

Figure 5-1 shows the contents of a typical tagged packet. This particular packet traveled a very long path, 15 hops, from IMP 43 to IMP 9. (Note that since the tags are created as the packet is about to be sent on an inter-IMP line, there is no tag entry for the destination IMP.) The packet did not encounter much queueing delay on this path -- it was a long packet, and 19.2 milliseconds is its transmission delay on a 50 kbps line. Only on the lines between 43 and 56, 12 and 47, and 47 and 6 did it experience a delay larger than its transmission delay. Note

that the line between 6 and 44 has a speed of 230.4 kbps, so the delay through IMP 6 is less than 6.4 ms.

```
MESSAGE   838
          IMP:  43          DELAY:  25.6  MS.
          IMP:  56          DELAY:  19.2  MS.
          IMP:  11          DELAY:  19.2  MS.
          IMP:  15          DELAY:  19.2  MS.
          IMP:  45          DELAY:  19.2  MS.
          IMP:  34          DELAY:  19.2  MS.
          IMP:   4          DELAY:  19.2  MS.
          IMP:  25          DELAY:  19.2  MS.
          IMP:  24          DELAY:  19.2  MS.
          IMP:  12          DELAY:  25.6  MS.
          IMP:  47          DELAY:  44.8  MS.
          IMP:   6          DELAY:   0.0  MS.
          IMP:  44          DELAY:  19.2  MS.
          IMP:  10          DELAY:  19.2  MS.
          IMP:  37          DELAY:  19.2  MS.
```

Figure 5-1

When many thousands of tagged packets are generated, it is desirable to have a program which reduces the data to some suitable form. We developed such a program, and figure 5-2 shows some sample output from it. In this example, we have collected tagged packets from three source IMPs -- 14, 47, and 50. All had a single destination -- IMP 9. The output shows exactly how many packets from each source were collected (2672 from IMP 14, 2667 from IMP 47, and 2675 from IMP 50). It also shows the average delay per packet for the packets from each source (34.94 ms. for

```
SOURCE:  14      COUNT:    2672      DELAY:  34.94  MS.

         PATH: 14-18-10-37- 9
                          COUNT: 2672      DELAY:  34.94       100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 4.00   HOPS

SOURCE:  47      COUNT:    2667      DELAY:  32.40  MS.

         PATH: 47-55-59- 9
                          COUNT: 2667      DELAY:  32.40       100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 3.00   HOPS

SOURCE:  50      COUNT:    2675      DELAY:  43.56  MS.

         PATH: 50-14-18-10-37- 9
                          COUNT: 1417      DELAY:  45.62        53%

         PATH: 50-29-46-60-58- 9
                          COUNT: 1258      DELAY:  41.24        47%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 5.00   HOPS


TOTAL MESSAGES FROM ALL SOURCES = 8014
PERCENT LOOPING PACKETS = 0.00%
```

Figure 5-2

IMP 14, 32.44 ms. for IMP 47, and 43.56 ms. for IMP 50). All the packets from IMP 14 followed the same path, as did all the packets from IMP 47. On the ther hand, two different paths were used to route traffic from IMP 54. The output shows exactly how many of these packets (in absolute numbers and percentages) travelled each of the paths, as well as the delay on each path. The reduction program also computes the average path length for each source-destination pair, and counts the number of looping packets (packets which traverse some IMP more than once) from each source. The output of this program enables us to see exactly how packets get routed under a variety of network conditions.

In order to process the tagged packets, it is necessary first to collect them on our TENEX system, which is a host on IMP 5. However, it is sometimes convenient to experiment with traffic flows that do not have IMP 5 as their destination. (Note that the packets shown in Figures 5-1 and 5-2 were directed to IMP 9, not IMP 5.) We devised the following procedure to achieve this. Whenever a raw tagged packet is destined for the discard fake host at any IMP, it is not discarded; rather it is forwarded to TENEX without any further tagging. (A "raw" packet is a datagram, a special kind of data packet which is not subject to any end-end flow control or sequencing. The "discard fake host" in each IMP is a special destination which ordinarily

serves as an infinite data sink, discarding all packets it receives.)

It is worth pointing out that although the tagged packet mechanism is quite simple in concept, it turned out to be surprisingly difficult to implement correctly. It is one of those things that is easy to design into a network initially, but difficult to graft into an existing network. One source of difficulties is that the IMP really does not expect the contents of a packet's data field to change, and a good deal of special care must be given if this is not the case (as with the tagged packets). Another source of difficulties is that there are many special causes to consider. For example, The number of nodes traversed by the packet may be greater than the number of data words in the packet (in which case the packet is too small to contain all the tags), or a packet may first be queued for a line, then re-routed to another because the first line goes down. Nevertheless, once all these difficulties were discovered and eliminated, the tagged packet provided a very simple and straightforward means of evaluating the new routing scheme's performance under a variety of conditions.

In order to use the tagged packet, we had to have a way of generating variable amounts of tagged traffic. To do this, we used the IMPs' message generator. Message generator is a fake host in each IMP. It is capable of transmitting one packet every

n "fast ticks", where n is a power of two and a fast tick is a period of 25.6 ms. The size of the generated packets, the value of n, the destination of the packets, and the settings of other bits in the host-IMP leader field are parameters which have to be set before the message generator is activated. (Among these other leader bits are the bits which specify whether a packet is raw or not, as well as whether it is tagged or not.) We did not find it necessary to make any changes to the message generators themselves, but we did make some improvements to the NCC's capability of controlling the message generator. One improvement was to extend the functionality of the message generator command so that it would set all the message parameters at an IMP at once, and another was to set the same parameters at several IMPs at once. This sort of improvement may not seem very important, but it must be remembered that our testing sessions were quite limited in length. Thus it was very important to be able to start up our experiments quickly, with as small a probability of making an error as possible. The improved message generator command contributed greatly to this.

Another simple and straightforward method of measuring the performance of the new routing scheme is simply to install a set of counters in the IMP, each of which counts the number of occurrences of some important event. We established the following ten counters:

1) Updates generated - Each IMP keeps a counter of the number of routing updates it generated. The number of updates generated by an IMP depends on the changes in delay on the lines leading from that IMP -- the more frequently there is a significant change, the more often there are updates generated. The delay does not usually change frequently by a significant amount unless there is a great deal of traffic, so the value of the counter should be roughly proportional to the amount of traffic through the IMP. Since it is the measurement process which decides when to send an update, this counter enables us to make a simple check on its performance.

2) Update packets processed - Each IMP keeps a count of the number of routing updates it has processed, so we can determine if the frequency of routing update processing at some IMP is similar to what we expect. Also, since all routing updates are processed at all IMPs, at any given time this counter should have the same value in all IMPs. If that were not the case, it would prove that the updating protocol was failing to deliver all the updates to all the IMPs.

3) Line updates processed - Every update packet from a particular IMP contains information on all the lines emanating from that IMP. That is, every update packet contains several "line updates", a line update being the update for a particular line. The shortest path computation processes the line updates

one at a time, so it is the number of line updates rather than the number of update packets which determines how often the computation runs. In order to see how often the computation is run, the number of line updates is counted. This number should also be the same in all the IMPs.

4) Average length of routing update queue - When routing update packets arrive at an IMP they are queued for processing. The average size of this queue is an indication of how much computational load is placed on the IMP due to the processing of routing updates. We measure the average queue length with the following techniques. Whenever a routing update packet arrives, the number of routing update packets which are already on the queue (including any packet which is currently being processed) is counted, and this count is added to a cumulative counter. When this cumulative counter is divided by the total number of update packets processed, the result is the average number of update packets behind which a newly arriving update must wait, i.e., the average queue length.

5) Maximum length of routing update queue - In order to get some idea of the variance in the length of the routing update queue, we also keep track of its maximum length.

6) Number of line updates which report changes - The new routing scheme sometimes causes updates to be generated even if

there is no change in the line-state information. It is interesting to know what proportion of the line updates actually report changes. This number should also be a constant from IMP to IMP.

7) Number of updates which may cause changes in shortest-path tree - The shortest-path computation that runs in each node produces a shortest-path tree of the network with that node as the root. Certain line updates can never cause changes in a node's shortest-path tree. If the line update reports no change, or if it reports on a line which is not in the shortest-path tree and that line has not improved, then no change in the tree can result. Each IMP keeps a count of the number of line updates it processes which may cause changes in its tree (which is not the same as the number which actually do cause changes in the tree). Since each IMP has a different shortest-path tree, this number is not a constant from IMP to IMP. Rather, it gives an indication of which IMPs have to perform the most computational work to react to changes in network delay.

8) Old updates - The updating protocol has been carefully designed to ensure that updates which are received out of sequence are discarded, rather than being processed out of order. Each IMP maintains a count of the number of update packets arriving out of sequence.

9) Duplicate current updates - Because of the way the updating is done, it is possible for an IMP to receive duplicate copies of some update. The number of such duplicates is counted.

10) Retransmissions of routing updates - The updating protocol employs a positive acknowledgement retransmission scheme. An update will be retransmitted on a line if the acknowledgement for the update is not received within a certain amount of time. The number of retransmissions made on each line is counted.

11) Received spurious retransmissions - A "spurious" retransmission is one which was not really necessary. For example, if an acknowledgement does not get through, the update being acknowledged will be retransmitted, although it has already been correctly received. The same thing will happen if the retransmission time-out period is too short. All update packets carry a special bit indicating whether or not they are retransmissions. This enables the IMPs to count the number of spurious retransmissions they receive. By comparing this number with the number of retransmissions sent to the IMP, one can determine what fraction of the retransmissions are spurious.

It is possible to zero out all these counters in all IMPs from the NCC at the beginning of each test session. At the end of each test session, the counters are collected into a TENEX

file, where they are converted to a human-readable form. Unfortunately it is not possible to take a network-wide snapshot of the state of the counters at some instant. It takes about 10-15 minutes to collect the counters, and during this interval the counters keep on counting. This means that in comparing counters from different IMPs we must remember that the counters have not all been collected simultaneously. However, this has not proven to be a real problem in practice. Appendix 1 shows the values of these counters after one test session.

It is often useful, when testing the new routing scheme, to be able to look at the shortest-path tree that has been computed by an IMP. To facilitate this, we developed a program which looks into a running IMP (or, alternatively, an IMP core dump), figures out what the tree is, and prints it out in an easily readable format. Figure 5-3 shows some sample output from this program. The "father-son" relation is indicated by vertical spacing, and the "sibling" relation is indicated by a horizontal line. This program for displaying trees is a very valuable software tool. Before it was developed, the only way to look at the tree in some IMP was to stop the IMP, dump it, and then spend 30 minutes crawling through the dump, trying to reconstruct the tree. (Any tool which saves a person 30 minutes at a shot is very valuable indeed!)

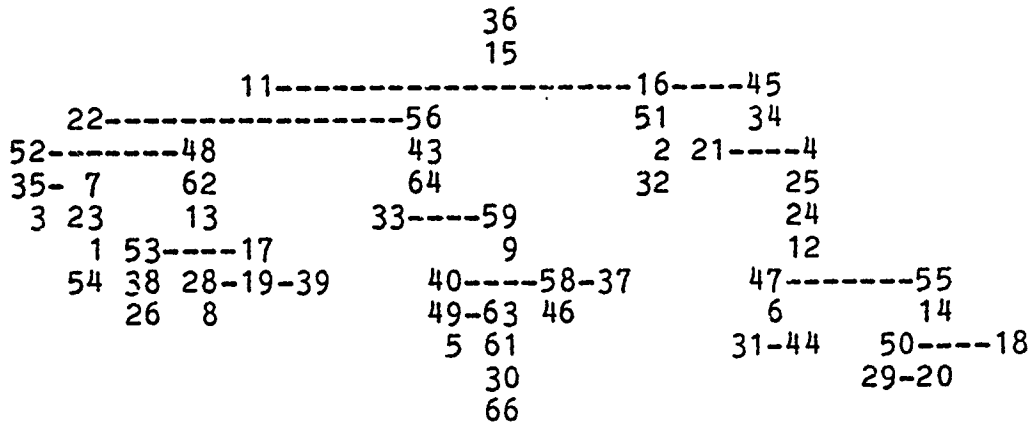


Figure 5-3

We also developed a set of programs enabling all routing updates, or only those routing updates from a particular IMP, to be collected on TENEX, where they can be processed. This enables us to directly inspect the outputs created by the delay measurement process while the new routing scheme is running. We have also used the IMP's standard trace package to collect data on the delays of individual packets.

Before proceeding to the next chapter, where the results of our testing will be summarized, it is worthwhile discussing briefly some of the possible software tools which we considered but chose not to develop. For instance, we did not develop any

instrumentation code for timing the new routing scheme. As a result, we cannot claim to have measured the amount of CPU bandwidth devoted to routing. Since the shortest-path computation is interruptable by many other processes, attempting to time it would involve a large amount of IMP code which turns the timers off whenever the code is interrupted, and then back on when the code resumes. Additional complications arise from the fact that pieces of the routing code occur at different places in the IMP, and at different priority levels. Although it would have been very desirable to have exact timing information about the new routing scheme, we were dissuaded by the complexity of the task. Not only would it have required an excessive amount of programmer time, but it would have caused us to violate the rule that all measurement tools should be simple.

In our first semiannual report, we described the Snapshot Measurement Package, a software package which can be loaded into the IMP and used to evaluate the performance of a routing scheme under extremely heavy loads. Unfortunately, we have not been able to use this package recently for the simple reason that it will not fit into the IMP along with the two routing schemes.

As discussed previously, each IMP counts the number of line updates which <u>may</u> result in changes to its shortest-path tree. It would have been more desirable to count the number of updates which <u>do</u> cause changes, and to measure to magnitude of these

changes. Unfortunately, there is no single point within the shortest-path computation at which one can tell whether a change has been made. During the computation, the shortest-path tree may be transformed several times, but there is no simple way of knowing whether any of these transformations will survive to the end of the computation, or whether they will be transformed again, possibly back to what they were initially. The only way to tell whether a routing change has been made in the course of a particular instance of running the shortest-path computation is to save a copy of the initial tree, so it can be directly compared with the tree which exists after the computation is done. The memory and processing expenses to do this (especially the former) are too great to make it worthwhile.

As described previously, we have the ability to display the shortest-path tree in any IMP. This enables us to do some spot-checking but it does not give us a means of systematically examining the way the trees change in response to particular events. We considered the possibility of implementing an event-driven routine which would write a representation of the shortest-path tree into a packet, which could then be transmitted to our TENEX system. In order to do this, the IMP would have to run with interrupts inhibited for as long as it takes to copy the tree. This is clearly undesirable, so we decided not to implement the feature.

6. TESTING THE NEW ROUTING SCHEME --- RESULTS

In this chapter we present selected results from our testing of the new routing scheme. It is not our intention here to discuss all our tests nor to present all our data. Rather, we present results which we believe to typify the performance of the new routing scheme. For presentation purposes, we divide our tests into five categories: topological stress tests, resource utilization tests, updating protocol tests, traffic flow tests, and packet delay tests. This way of categorizing our tests is purely a matter of presentation, and does not correspond to any chronological or operational categorization.

1. Topological stress tests. A routing scheme must be able to react quickly and correctly to sudden changes in the topology of the network (a line or a node coming up or going down.) The routing scheme must not only be able to respond to single topological changes, but also to multiple simultaneous changes, including those sets of changes which lead to network partitions. One way to stress the routing scheme is to induce such topological changes, both singly and in combination. This puts stress on all the significant components of the routing scheme. When a line goes up or down, the measurement process must detect that immediately, and cause an update to be sent. Determining whether this actually works is more complicated than it may seem, since there are many situations that may bring a line down, and

the measurement process must detect all of them. When lines go up and down in rapid succession, a great deal of stress is placed on the data base management procedures. Applying topological stress is the best way to detect any incorrect order-dependencies which exist. Inducing topological stress also tests the reachability algorithm of the shortest path computation. When the topology changes, the shortest-path computation must be able to determine which IMPs are unreachable. Topological changes also stress the updating protocol. In our second semiannual report, we discussed the way in which partitions of the network can cause the IMPs to get out of sync with each other. The updating protocol was carefully designed to avoid any such problems, but only by actually partitioning the network could we determine whether our design really worked. Also, the presence of topological changes puts greater than normal stress on the part of the updating protocol which attempts to ensure that no updates are processed out of order. Ordinarily, updates from a given IMP must be separated by at least 10 seconds (the measurement period). However, if a line at some IMP goes down or comes up, two updates from that IMP may be sent with an arbitrarily small interval between them. This makes it much more likely that the updates will arrive at some other IMP out of order, and thus it exercises the part of the updating protocol that detects out-of-order updates.

We caused lines to go up and down using five different methods:

a) Altering a particular memory location in the IMP. This causes the line to go down, and then to come back up as soon as possible (i.e. one minute later).

b) Looping and unlooping lines by command from the NCC. A line looped in this way stays down until it is unlooped by another command.

c) Looping and unlooping lines by pressing a button on the modem simulator box. (This could only be done at IMPs located on BBN's premises, which used modem simulators rather than real modems and phone lines.)

d) Physically pulling out and inserting cables into a modem simulator box.

e) Physically pulling out cables and re-connecting them in a different configuration. Thus not only does a line go down, but when it comes back up, the IMPs it is connected to have different neighbors over that line than they did before.

To induce IMPs to go up and down, we sometimes restarted them by command from the NCC (which causes the IMP to go down and then come back up within several minutes), and we sometimes halted

them manually by means of the console switches. We have induced partitions by various combinations of these procedures; and have experimented with partitions of various sizes and durations. In addition, during many of our field tests, lines and/or IMPs went up or down due to "natural causes", providing further unplanned topological stress tests.

These tests turned up both program bugs (mostly in the data base management procedure) and design bugs. It is interesting to note that we were not able to detect all these bugs by testing in the lab. We found additional bugs when we tested in the ARPANET, with all IMPs but one in state II, and a single IMP in state III. We found more bugs yet when we tested the whole network in state III. After these corrections, it appears that the new routing scheme does respond correctly and quickly to topological changes. Such changes do cause immediate transmission of routing updates. These updates are processed correctly, and the correct changes are made in the shortest-path tree. Nodes are considered unreachable by the new routing scheme when, and only when, they really are unreachable. The network recovers correctly from partitions, without loss of update synchronization. When a lot of traffic is being routed over a line which goes down, the traffic is re-routed without creating a network disturbance. When a line comes up, immediate use is made of it, whenever possible. Since we have put the new routing scheme through a

large variety of topological stress tests, we are confident of its ability to withstand the topological stresses that are placed on it under operational conditions.

2. Resource utilization. Several of the counters described in the previous chapter have enabled us to draw conclusions about the utilization of resources by the new routing scheme. These are reported on in this section. All measurements reported here were taken while the network was running in state V.

a) Length of routing queue. When measured over a period of about an hour, most IMPs show a maximum routing queue length of 2. That is, at most IMPs, there was at least one routing update which arrived during the hour that had to wait on the queue while two other routing updates were processed first. Many IMPs show a maximum queue length of 3. Once in a rare while, a maximum queue length of 5 is detected, and maximum queue lengths of 1 are not uncommon. The average queue length, on the other hand, has never been observed to be above 0.05 in any IMP, regardless of its maximum queue length. Typical values of the average queue length are 0.02 and 0.03. This means that almost all routing updates are processed as soon as they are received; it is very rare for a routing update to have to be queued.

b) Total number of updates generated. The number of updates generated at a particular IMP varies greatly. Lightly loaded IMPs generate updates at close to the minimum frequency (once every 50 seconds), and heavily loaded IMPs at close to the maximum frequency (once every 10 seconds). Several one-hour measurements taken during our testing periods have shown the total number of update packets processed by each IMP to be about 5000, or about 1.4 update packets per second. If we assume that every update packet contains 2.5 line updates (actually the average update packet seems to contain about 2.2 - 2.4 line updates) then the total line bandwidth due to routing updates (assuming that no updates are retransmitted, and that all updates flow on all lines) is 246 bits per second. This is 0.5% of a 50 kbps line, and 2.6% of a 9600 bps line. This compares favorably with the 3% - 15% used by the old routing scheme.

c) Fraction of the updates which report changes. Since routing updates are always sent at least at some minimum frequency, even if there is no change in delay, not all line updates report changes. The fraction of line updates which do report changes has been observed to vary from as little as 0.26 to as much as 0.56, with

0.45 a more typical value. The fraction of line updates which may cause changes in some IMP's shortest-path tree varies greatly from IMP to IMP, but is typically about one-third. (That is, if 45% of the line updates report changes, about one third of these, or 15%, may cause changes in the shortest-path tree of an average IMP.)

These figures indicate that the amounts of processor and line bandwidth taken by the new routing scheme are quite modest, well within our expectations, and present no problem. It must be pointed out, however, that our testing periods are generally during early morning hours, when the network is quite lightly loaded.

3. Updating Protocol. The average number of retransmissions of routing updates per line varies considerably, depending on network conditions. During a fairly typical one-hour measurement, we found the average number of retransmissions per line to be 66 (with a standard deviation of 118). Since there were about 5000 update packets sent on each line during that period, the average increase in line bandwidth due to retransmissions is about 1.3%. However, the peak increase due to retransmissions approaches 15%.

Almost all of these retransmissions have been unnecessary. That is, in most cases, the number of retransmissions made on

lines leading to a given IMP is exactly the same as the number of spurious retransmissions received by that IMP. We have, however, observed small numbers of IMPs (about 5) receiving small numbers (fewer than five) of non-spurious retransmissions. It must be pointed out, however, that our measurements have been made during early morning test periods when the network was quiet. Significantly different results may be obtained when measurements are done during network busy hours.

Only a tiny number of routing updates have ever been observed to arrive out-of-sequence. In one one-hour measurement, a single IMP received 17 out-of-sequence updates. In another, each of five IMPs received a single out-of-sequence update.

Each IMP also counts the number of updates it receives which, although they are in proper sequence, have already been seen (i.e. are duplicates). The updating protocol sends all updates on all lines, so each IMP necessarily sees each update n times, if it has n lines. So if there are a total of m update packets, each IMP must see at least m*(n-1) duplicate updates. Any received spurious retransmissions are also counted as duplicates. When the expected duplicates and the spurious retransmissions are subtracted from the count of duplicates, we found during a one-hour measurement that each IMP received an average of 193 duplicate updates (with standard deviation of 161). When retransmissions and duplicates are taken into

account, the average line utilization due to routing updates is 0.53% of a 50 kbps line and 2.7% of a 9600 bps line.

4. Traffic tests. We performed a number of tests to see how the new routing scheme reacts to particular offered traffic loads. One of our goals was to determine how well the new routing scheme does at routing traffic around congested areas. Figure 6-1a shows the shortest-path tree at node 30 before one of

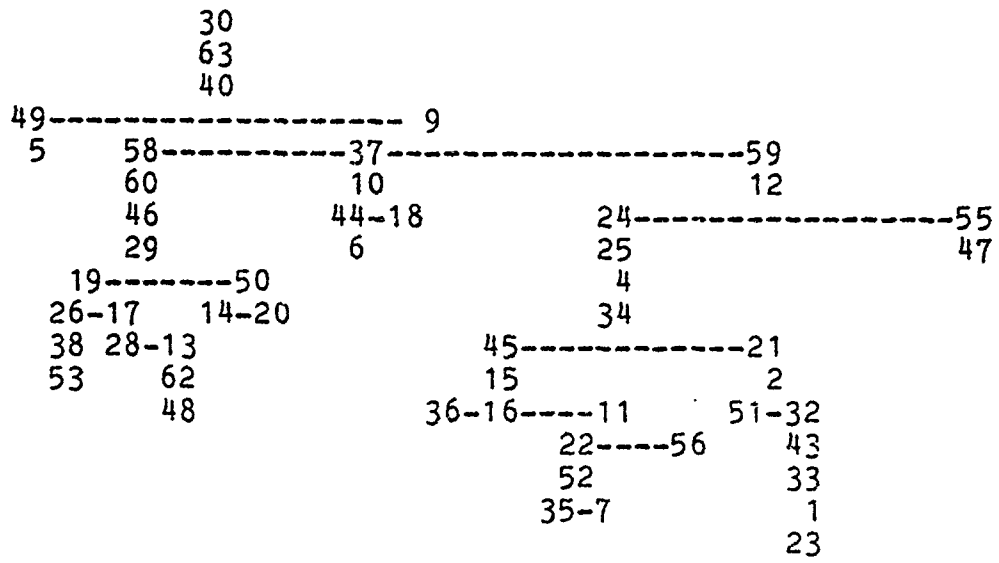


Figure 6-1a

our experiments. (The network was running in state V at this time.) Note that node 24 has a rather large subtree, consisting of 21 nodes. After displaying the tree, we turned on a message

generator to send traffic from node 24 to node 25. The generator was set to its maximum rate. Figure 6-1b shows node 30's tree

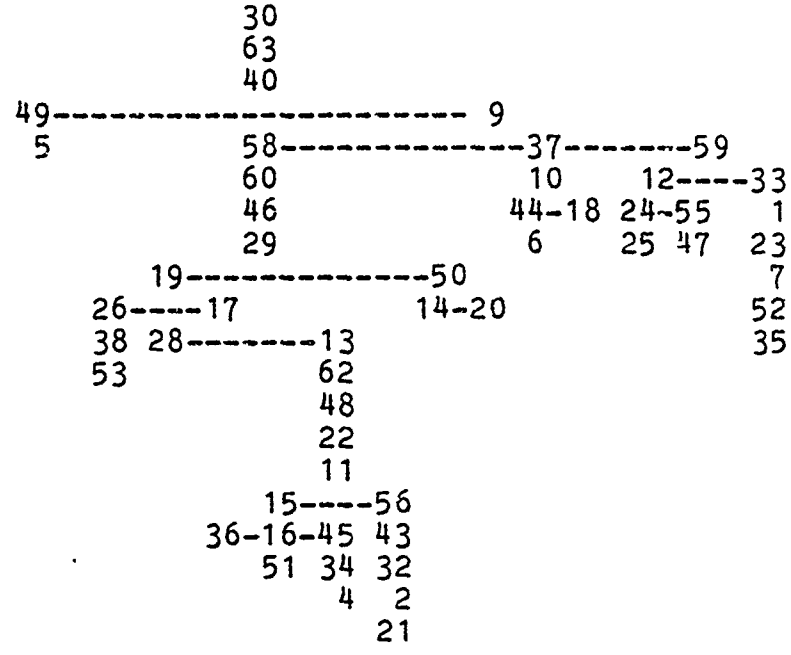


Figure 6-1b

after the generator was turned on. Note that the size of node 24's subtree has been reduced to 1. That is, before the generator was turned on, node 30 was willing to use the line between 24 and 25 to send traffic to any of 21 IMPs; after the generator was turned on, node 30 was willing to use that line only for traffic destined for node 25. Figure 6-1c shows node 30's tree after the generator was turned off. The subtree of node 24 has now been enlarged to 10 nodes. This shows that the new routing scheme was able to detect the load placed on the line between 24 and 25, and react to it properly.

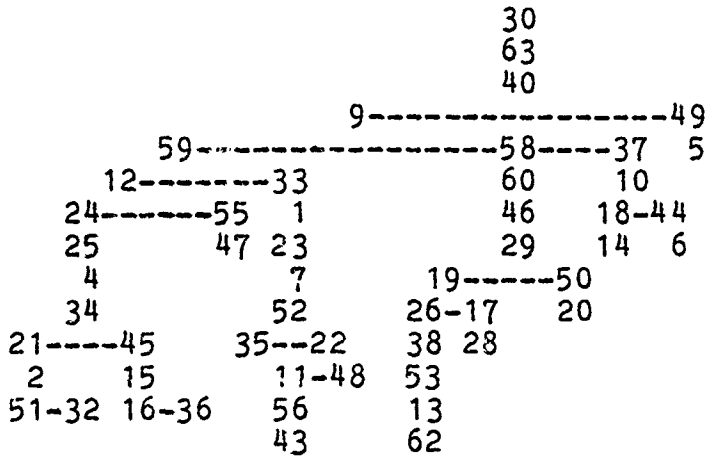


Figure 6-1c

We also performed experiments in which we deliberately induced severe congestion in order to see how the new routing scheme would react. We prepared a special software patch which, when placed in an IMP, would cause that IMP to refuse to acknowledge a specifiable percentage of the packets which arrive over a particular line. These packets then have to get retransmitted by the neighbor, causing the delay on that line to get very large. We placed this patch in IMP 59, causing it to refuse to acknowledge packets from its neighbor IMP 64. Packets from other neighbors of 59 were not affected by this patch. We

sent 20 packets per second (each packet being 1192 bits long) from 64 to 59, in order to induce a high delay on the line between 64 and 59. Then we sent tagged packets from IMP 43 (a neighbor of 64) to IMP 9 (a neighbor of 59). The min-hop path for such packets is 43-64-59-9. However, because of the congestion on line 64-69, we would expect at least some of the packets to travel alternate routes of greater hop length. We performed six experiments. The results of these experiments are shown in Appendix 2, along with a map of the ARPANET as it was when these experiments were done. The reader should refer to this appendix while reading the discussion below.

a) For our first experiment, we set node 59 to reject (i.e. to fail to acknowledge) 80% of the traffic arriving on its line from 64. We sent tagged packets from 43 to 9 at the rate of 10 packets per second, with each packet containing 1192 bits (including all overhead and framing). The results shown in Appendix 2a indicate that all the traffic from 43 to 9 avoided the min-hop path, travelling over five longer alternate paths. Close inspection of these five paths shows that there are really only two disjoint paths used, the other three being minor variants of the first two. The traffic from 43 divides almost equally over their two paths, with slightly more traffic using the path which is slightly

shorter. This is the best result that can be expected from a single-path routing scheme.

It is worth noting that this sort of performance would not be possible with the old routing scheme. Note that some packets traveled a 12-hop path, even though a (badly congested) 3-hop path was available. The old routing scheme assigns each network line a "delay" which is between 4 and 12. Thus the highest possible delay on a 3-hop path would be 36, and the lowest possible delay on a 12-hop path would be 48. Since 36<48, the old routing scheme would always use the 3-hop path, no matter how badly congested it was. The new routing scheme shows much greater adaptability.

b) In our second experiment, whose results can be found in Appendix 2b, we set node 59 to reject only 67% of the packets arriving on the line from 64. This causes line 64-59 to have a lower delay than in the first experiment. We see now that 28% of the traffic from 43 to 9 did use the min-hop path, experiencing a rather large delay. The other 77% of the traffic traveled over alternate paths. As can be seen from the results, the amount of traffic on an alternate path is inversely related to the delay on that path.

In comparing experiments a and b, we see that the new routing scheme has a tendency to gravitate to min-hop routing unless there is a very strong reason to avoid it.

It is also worth noting that of 2010 packets collected, only one looped.

c) Our third experiment was just a repeat of our first, except that the amount of traffic from 43 to 9 was doubled to 20 packets per second. Here 2% of the traffic from 43 traveled the high-delay min-hop path, and the other 98% split evenly between variants of the two disjoint alternate paths. What is interesting to note here is that the number of variants of these paths has increased from 5 to 7. This illustrates an interesting property of the new routing scheme. As the offered traffic load increases, there is a tendency to look for paths where there is excess bandwidth, and to try to make use of the excess.

d) Our fourth experiment is a repeat of the second, with the traffic rate from 43 increased to 20 packets per second. (Or it may be considered as a repeat of the third experiment with the percentage of packets from 64 rejected by 59 decreased from 80% to 67%). The results

are similar to those of the second experiment. A significant proportion of the traffic traveled the min-hop path, and the remainder is divided over variants of the two disjoint alternate paths. The number of non-looping variants is only 4, but there are 10 paths which contain loops. The percentage of looping packets is less than 1%; these packets were already on their way across the network when a routing change was made causing them to backtrack and change paths. In none of these loops is any node traversed more than twice. This indicates that the loops are a purely transient phenomenon occurring during a period of adaptation, rather than a long-term phenomenon due to some problem with the routing scheme. However, it must also be noted that some of these loops contain a large number of hops, and packets which loop do have a significantly larger end-end delay than do other packets.

e) Our fifth experiment was a duplicate of the first, except that we also sent tagged packets from node 56 (a neighbor of 43) to node 9. Both 43 and 56 sent at a rate of 10 packets per second. Virtually no traffic traveled over the min-hop path. Node 43 sent about 60% of its traffic along the "northerly" alternate path, and 40% along the "southerly" one. Node 56 split its traffic in the opposite proportion.

This last routing pattern is interesting, and deserves some discussion. One might have expected, a priori, that node 56 would have sent all of its traffic on the southerly route, while 43 would have sent all of its traffic on the northerly route. These are the respective minimum delay routes, and use of these routes would prevent traffic flows from the two source nodes from interfering with each other (until they actually reach their destination). However, this sort of reasoning must be used with extreme caution. It is true that the traffic from 43 which went north experienced a smaller delay than the traffic that went south. But is simply does not follow that it would have been better had all the traffic from 43 gone north. This is especially true since we have no way of knowing what other traffic flows existed in the rest of the network at the time we did our experiment. It is interesting, though, to compare the delay from 43 to 9 which we observed in the first experiment with that which we observed in this experiment. Other things being equal (which might or might not have been the case), this will enable us to see how the introduction of the traffic flow from 56 to 9 impacted the delay of the traffic flow from 43 to 9.

In the first experiment, the average delay of packets from 43 to 9 along the northerly path (43-32-2-21-34-4-25-24-12-59-9) was 312.3 ms., or 31.23 ms. per hop. The corresponding delay in the fifth experiment was 333.87 ms. or 33.39 ms. per hop. This

is an increase of only 2.2 ms. per hop. In the first experiment, the average delay from 43 to 9 over all paths was 317.51 ms., or 28.66 ms. per hop. The corresponding delay in the fifth experiment is 33.12 ms. per hop. This is an increase of less than 4.5 ms. per hop. The new routing scheme treats the per-hop delays in quantized units of 6.4 ms.; increases of less than this value would not be expected to cause a large change in the routing patterns.

Another interesting fact about the fifth experiment is that the average delay from 56 to 9 along the southerly path is almost the same as the average delay from 43 to 9 along the northerly path. That is, there are a pair of neighbors who see approximately equal delays along a disjoint pair of paths to a common destination. Under the old routing scheme, this sort of situation tends to cause the formation of long-lasting ping-pong loops between the pair of neighbors. Under the new routing scheme, no such loops are formed, and the only penalty is a small increase on the average per-hop delay (though it must be admitted that there is a larger increase in the variance of the per-hop delay.)

In many of these experiments, paths were used which are slight variants of the main paths. These variants tend to be very similar to the main paths, but have a few more hops. Examining these variants shows an important property of the new

routing scheme, namely that it shows a tendency to seek out and use paths on which there is excess bandwidth, as long as these paths do not diverge too greatly from the paths of least delay. Under conditions of overload, the paths with the fewest number of hops fill up quickly, and the new routing scheme has been observed to try to use all possible paths in order to deliver the packets to their destination. That is, under overload conditions, the new routing scheme can attempt to fill the whole net with traffic. This is appropriate from the perspective of routing, but it illustrates the need for improved flow control and congestion control techniques to prevent the network from overloading.

In our second semiannual report we presented some mathematical analysis which purported to show that the new routing scheme would enter an unstable state under certain conditions. When in this state, the traffic in the network would oscillate wildly from one bad path to another, never settling down to a good path. This sort of oscillation was predicted to be especially bad when the network consists of a loop topology. We engaged in an extensive series of tests to determine whether instability could be a real problem for the new routing scheme as implemented in the ARPANET. Appendix 3 contains the results of some of the experiments which we did in our lab. We set up a four-node loop network at the lab, and gave it a stub connection

to the ARPANET (so that we could use the ARPANET to collect data from our lab net) - this network is pictured in Appendix 3. The experiments are discussed and described below.

a) In our first experiment, we had each of nodes 60, 61, and 66 send 10 packets per second to node 30, with each packet being 1192 bits long. Nodes 61 and 66, the immediate neighbors of 30, each sent 99% of their traffic to 30 over the single hop path. Node 60 split its traffic over the two possible paths. If routing oscillations were present, we would expect that 61 would send half its traffic via 66, and 66 would send half its traffic via 61, but this has not occurred.

It is interesting to note the behavior of the traffic from node 60. Rather than splitting 50-50 over the two possible paths, it splits 60-40. Furthermore, the delay on the path via 66 is only half the delay on the path via 61. Our explanation for this is as follows. In node 30's internal numbering scheme, the line to node 66 is line 2, and the line to node 61 is line 3. If packets arrive simultaneously on both lines, the packet from the line with the smallest number is processed first. Therefore, if both lines are heavily utilized, a greater delay is seen on the line from 61 to 30 than on the line from 66 to 30. This sort of "unfairness" was observed very frequently in our lab tests.

b) Our second experiment was a repeat of the first, except that we doubled the rate at which packets were transmitted. We see that although the delays are much higher than in the first experiment, there is little change in the traffic patterns. The only difference is a slightly greater tendency for packets to enter 30 via 66. The routing does not show oscillation or instability.

The mathematical work presented in our second semiannual report suggested that, if oscillations did occur, they could be dampened by the use of a bias. A bias is a value which is added to the actual delay on a line before the shortest-path computation is done. Our experiments have all been done without the use of any explicit bias. However, it must also be pointed out that the new routing scheme will never report a delay of 0 on any line. (If it did, long-term routing loops might form.) The smallest reportable delay is 1 unit, or 6.4 ms. It is worth noting that when we repeated our experiments with a unit of 0.8 ms., we obtained the same results.

c) In our third experiment, we sent 10 packets per second from 60 and 66 to 30, but no traffic from 61. The packets were 1192 bits long. Again, no instability is

noted. Node 66 sent all its traffic on the one-hop path, while node 60 split its traffic almost evenly, with slightly more traffic traveling on the path with slightly less delay.

d) Our fourth experiment duplicated the third, but with double the amount of traffic. The delays are much longer, but there is no significant change in the traffic pattern.

e) For our fifth experiment, we added another line, between 61 and 66, to our lab network in order to introduce a more complex topology. Each of nodes 60, 61, and 66 sent 10 packets per second to node 30. Comparing the results of the experiment with those of our first experiment, we see that although the delay from node 60 is slightly better, the delay from nodes 60 and 61 is significantly worse. Node 66 sent 15% of its traffic over the cross-link, while node 61 sent 13% of its traffic over the cross-link, even though the delay of the traffic which used the cross-link was much worse than the delay of the traffic that did not. This seems to be due to the attempt of node 61 to take advantage of the fact that the delay on the line 66-30 is less than the delay on the line 61-30. That is, it makes sense for 61 to send some proportion of its traffic on the

two-hop path (61-66-30), and that is the sort of performance that would be expected of a multi-path routing scheme. However, in a single-path routing scheme, there is no way to control the exact proportion of traffic which uses the two-hop path. As a result, "too much" traffic is sent over the two-hop path, causing the delay on that path to get too high; at the same time, the delay on the one-hop path gets too low. When this happens, 61 switches back to using the one-hop path; which is correct. However, the routing scheme seems to overcompensate by causing some of the traffic from 66 to go on the two-hop path via 61. It is interesting that while the delay on the path 66-61-30 is twice the delay on the path 61-66-30, the latter path causes twice the traffic of the former path. This indicates that the more sub-optimal a path is, the sooner it is removed.

Appendix 4 shows the results of an experiment done in the real network to test the stability of the routing in a topological loop under conditions of overload. We sent traffic from nodes 13, 53, 38, 26, and 17 to node 19. We removed the line between nodes 13 and 62 so that this traffic could not get to 19 by heading west, out of the loop. Nodes 26, 38, and 17 sent 10 packets per second, while nodes 13 and 53 sent 20 packets

per second. All packets were 1192 bits long. This amounts to 83.4 Kilobits per second of traffic. Since this is too much traffic to be sent to TENEX, we were not able to collect tagged packets from all five sources at the same time; rather, we collected tagged packets from two sources at a time. Therefore, we should not attempt to compare the absolute numbers of packets sent from each source but only the percentages.

It is easy to see that there is no way this 83.4 kbps can be delivered to node 19 without overloading the neighboring IMPs. Node 19 can receive traffic over two 50 kbps lines. However, if 53, 38, and 26 all sent their traffic into 19 over the same line, there is 48 Kilobits per second on a single line, not counting any user traffic, or retransmissions on that line. This is more traffic than can be handled. (This is especially true since the line from 26 to 19 is 19's line number 3, i.e., it is the least favored of 19's three modems.) But any other routing pattern results in more than 50 kbps of traffic on a 50 kbps line.

The results are quite interesting. Node 17 sent almost 100% of its traffic to 19 on the one-hop path (99.88%, to be exact). Node 26, the other neighbor of 19, sent only 94% of its traffic on the one-hop path, and 6% going around the loop the long way. The next neighbors, 38 and 13, each split their traffic in about an 80/20 ratio, with the majority of the traffic taking the min-hop route. Node 53 split its traffic in a 68/32 ratio, with

the majority taking the more lightly loaded path, which was also the path with less delay. The most interesting result of this experiment is the fact that the routing was such as to equalize the average amount of our test traffic on the lines leading towards 19. The line from 26 to 19 carried 41.4 kbps of traffic, while the line from 17 to 19 carried 42.4 kbps. The line from 38 to 26 carried 33.1 kbps of traffic, while the line from 13 to 17 carried 33.6 kbps of traffic. Thus it appears that under heavy load, the new routing scheme tends to equalize the average line loading over the long term.

Appendix 5 contains the results of two experiments designed to show how the routing behaves under more moderate loadings. These are described below.

a) In this experiment, nodes 38, 26, 13, and 17 were each set to send 19.5 kbps of traffic to node 19. This results in a flow of 78 kbps to node 19, which is quite a high load for the ARPANET. Nevertheless, all traffic traveled in the min-hop routes.

b) In the second experiment, each of nodes 45, 34, 21, 16, 51, and 2 sent about 2500 bits per second to node 15. All traffic was min-hop, except for that from 21, which split about evenly between the 3-hop path and the 4-hop path. It is interesting that the 4-hop path from 21

contains a 230.4 kbps line, the line from 16 to 15. This experiment shows that the new routing scheme is able to take advantage of lines of differing speeds. It also shows that routing tends to be min-hop under light loads.

The results of these traffic tests can be summarized as follows:

i) The new routing scheme is capable of detecting congestion, and will route traffic around congested areas.

ii) Routing loops only occur as transients, and packets never travel any node more than twice. However, the actual size of the loop can be many hops, resulting in a long delay for packets which do loop.

iii) Traffic tends to be routed min-hop in the absence of any special circumstances.

iv) Under heavy load, the new routing scheme does not give optimal routing (which would be impossible for any single-path algorithm). However, it does not oscillate wildly between bad routing patterns.

5. Characteristics of individual packet delays. In our first semiannual report, we presented data showing that the

delays of individual packets traversing a line are much more variable than would be expected. In particular, even when enough traffic is placed over the line to saturate it, many packets still show very low delays. We could not explain why the delays were so variable, but we speculated that much of the variability might be due to various side-effects of the old routing scheme. Now that we have the ability to turn the old routing off, we have gathered some more data on packet delays to see if turning off the old routing causes any major change in the characteristics of the packet delays. It does not. Figures 6-2 and 6-3 plot packet delay vs. time on the line between IMPs 12 and 24, and between 24 and 25, respectively. These plots were obtained by sampling every tenth packet through the IMP. In each plot, we first sampled only the ordinary user traffic for several minutes, then turned on a message generator to saturate the line, then turned it off again after about 10 minutes. (Full details of the experimental technique can be found in the first semiannual report.) Even though the old routing was turned off during these experiments, the extreme variability in delay remains. We still do not know whether this variability can be explained by queueing theory, or whether it is due to some sub-optimality in the IMP protocols or software. We are currently investigating this phenomenon by simulation, and will continue to look for the correct explanation.

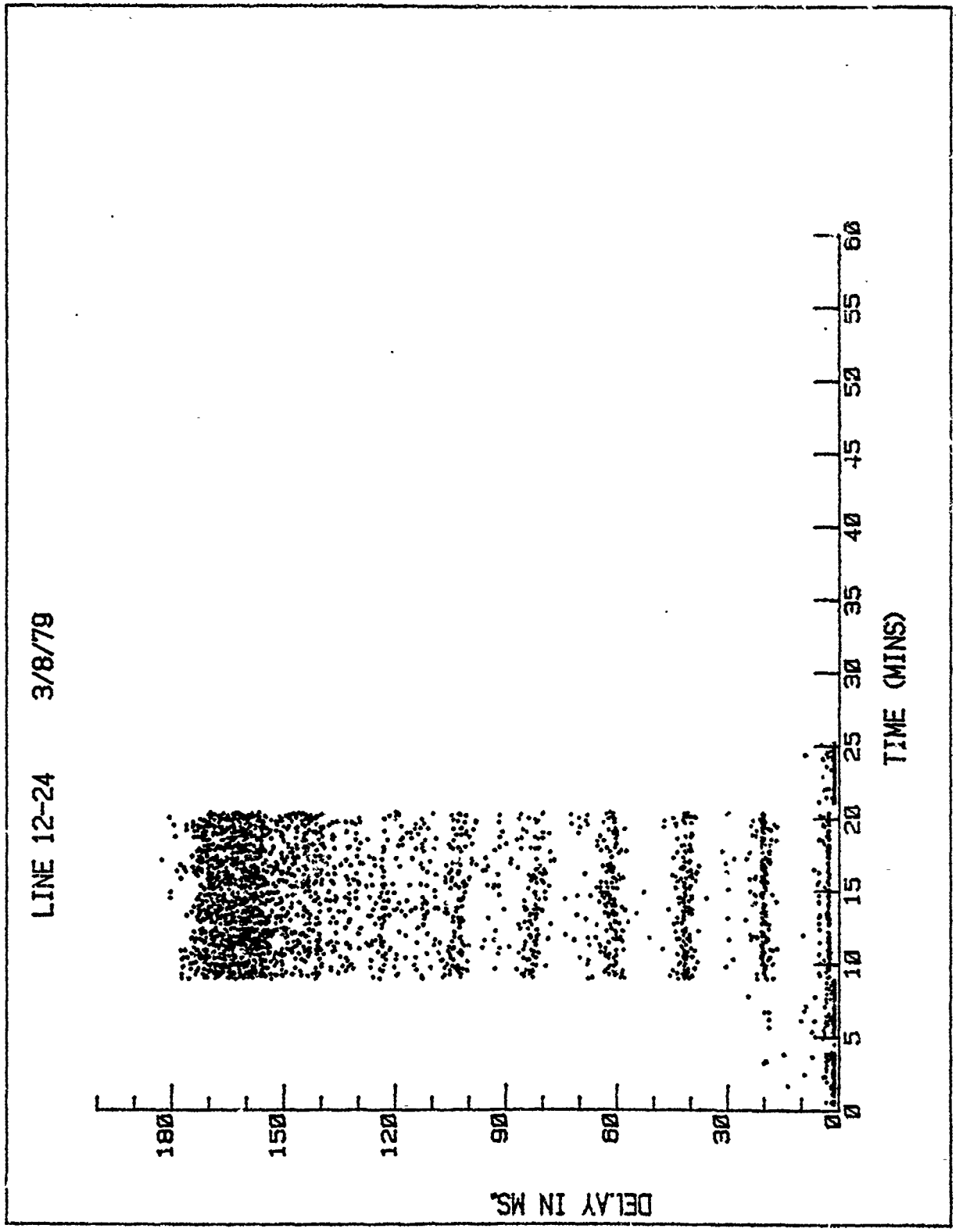

LINE 12-24 3/8/79
DELAY IN MS.
180
150
120
90
60
30
0
TIME (MINS)
0
5
10
15
20
25
30
35
40
45
50
55
60

Figure 6-2

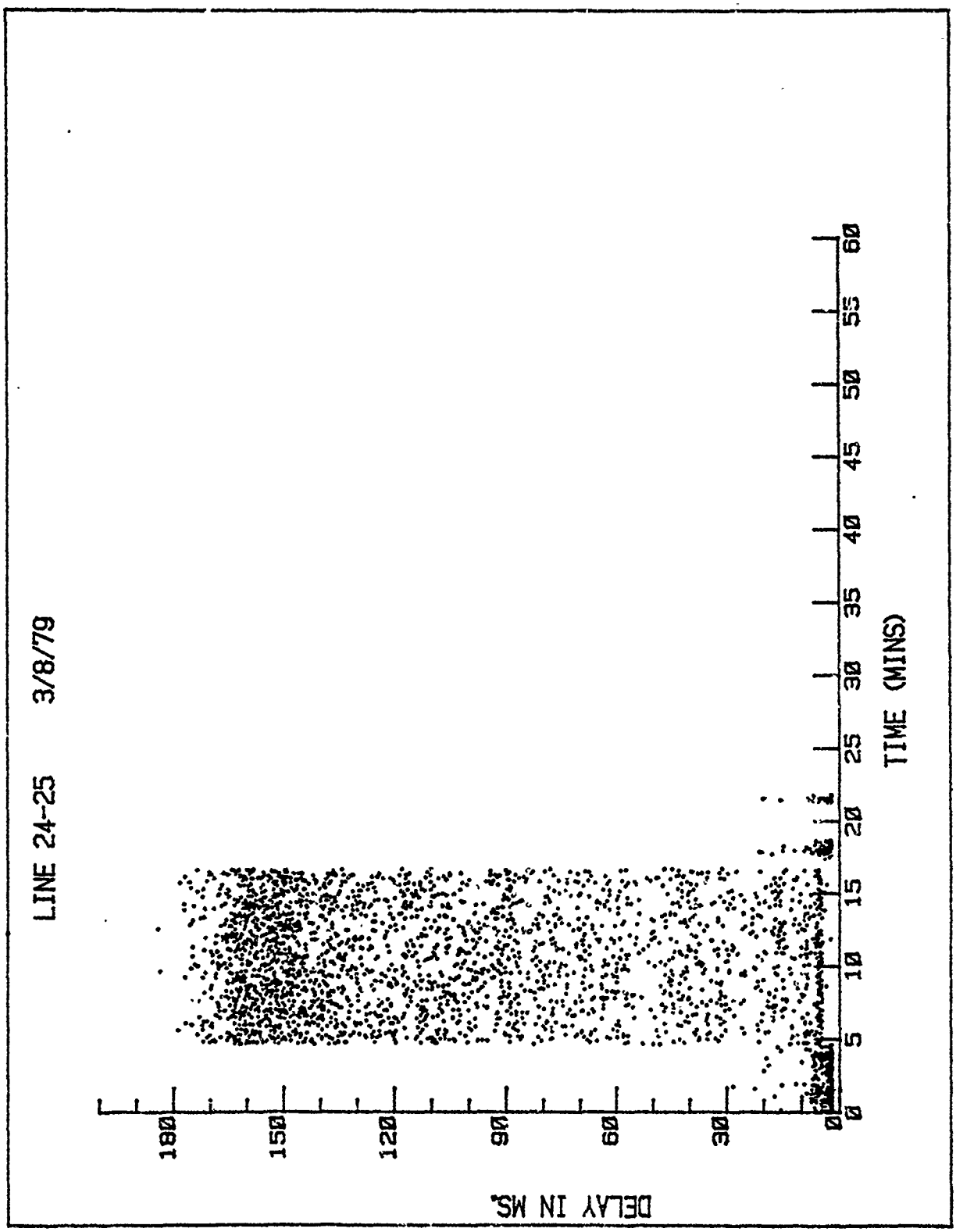

LINE 24-25 3/8/79
TIME (MINS)
0
5
10
15
20
25
30
35
40
45
50
55
60
DELAY IN MS.
0
30
60
90
120
150
180

Figure 6-3

To summarize, the new routing scheme seems to be working about as expected. While it does not result in optimal routing, it does perform well, and is successful in eliminating many of the problems associated with the old routing scheme. We have not encountered any unforeseen problems with the new routing -- its overhead is low, it results in good routing patterns, and it does not suffer from undesirable feedback effects. We are now ready to operate the network with the new routing scheme, removing the old routing for good.

7. BUFFER MANAGEMENT IN THE HONEYWELL 316/516 IMP

7.1 Introduction

Many congestion control schemes work by dividing the buffer pool into several smaller pools, and associating each of these pools with a particular function. By setting a lower limit on the size of a pool, one can ensure that there are always a certain minimum number of buffers available to serve the associated function. By setting an upper limit on the size of a pool, one can ensure that no more than a certain maximum number of buffers are devoted to the associated function. The way in which these maximum and minimum values are chosen determines the relative priority of the various functions. That is, if several functions are competing for a limited number of buffers, the competition is arbitrated by the maximum and minimum sizes of the various pools. As a prelude to considering congestion control schemes for the ARPANET, we have investigated the current buffer management scheme. The purpose of this chapter is to describe that scheme.

The IMP maintains four buffer counters, each of which specifies the number of buffers currently dedicated to specific functions. The counters are:

1) Free - the free count is just the number of free (unused) buffers.

2) Store-and-forward - this counts the number of buffers in use by the modem-out process, which includes:

   a) Buffers which are queued for transmission on an inter-IMP trunk.

   b) Buffers awaiting acknowledgement over an inter-IMP trunk.

   c) Buffers currently in transmission on an inter-IMP trunk.

   d) Buffers which were formerly queued for transmission on an inter-IMP trunk which went down.

   It is, however, possible for a buffer to be in one of the above categories without being counted as store-and-forward. This shall be discussed later.

3) Allocated -- This counts the number of buffers which have been pre-allocated by the source-destination protocol. An allocated buffer also counts as free until the packet for which it was allocated arrives.

4) Reassembly -- This counts the number of buffers correctly in use by all other functions.

It is also possible for buffers to be uncounted. There is a period of time after a buffer is no longer free, but before the IMP has decided what to do with it.

This is illustrated in Figure 7-1. When a packet arrives from a neighboring IMP, the "modem in" process gets a free buffer

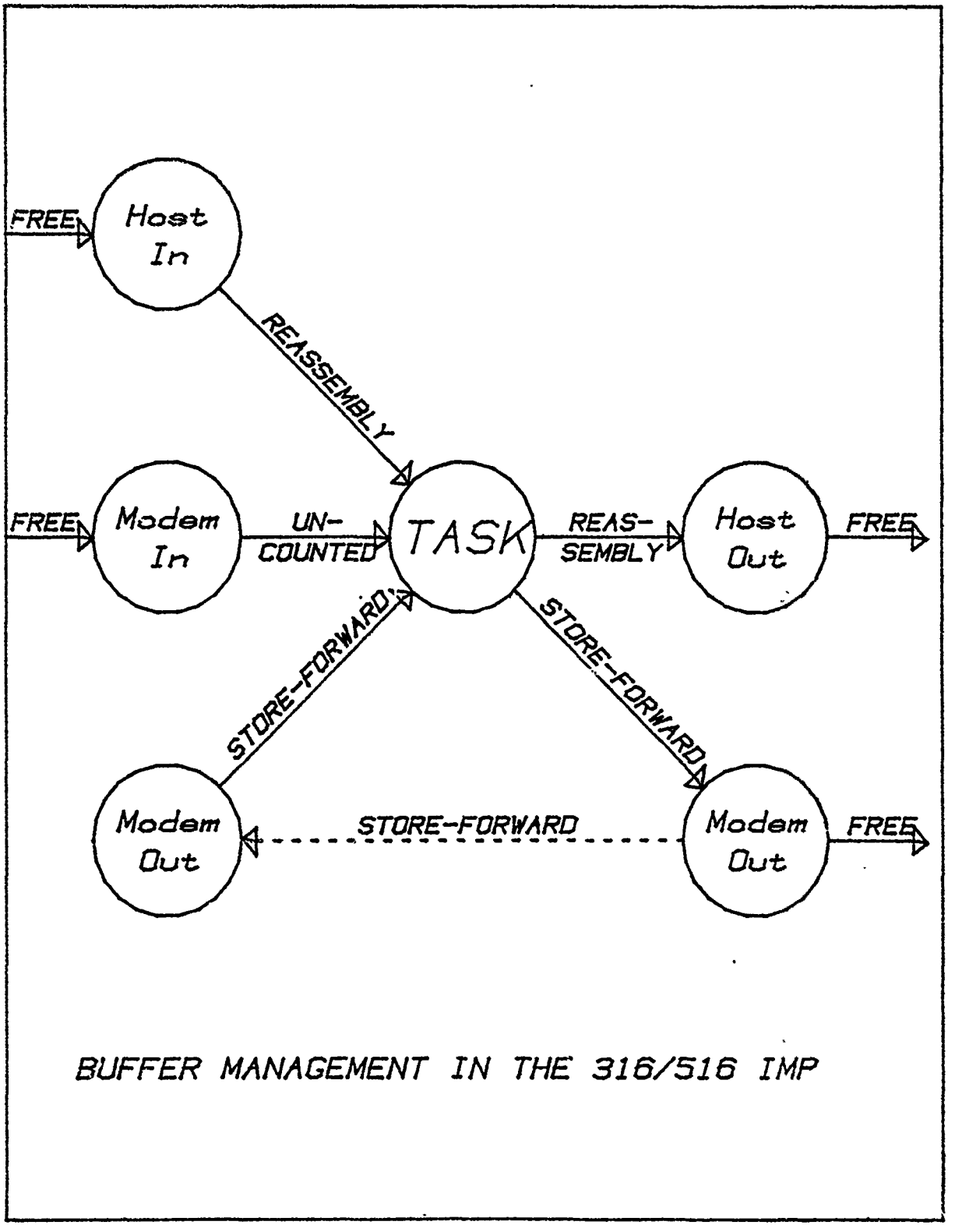

FREE
Host In
REASSEMBLY
FREE
Modem In
UN-COUNTED
TASK
REAS-SEMBLY
Host Out
FREE
STORE-FORWARD
STORE-FORWARD
Modem Out
STORE-FORWARD
Modem Out
FREE
BUFFER MANAGEMENT IN THE 316/516 IMP

Figure 7-1

for it, and places the buffer on the queue for the TASK process. The buffer is uncounted until TASK looks at it and determines how it should be counted. However, not all buffers on the TASK queue are uncounted. When packets are queued for transmission on a line which then goes down, these packets are replaced on the TASK queue for forwarding over a live line. These packets are always counted as store-and-forward even while they are on the TASK queue. Buffers which contain input from a real or fake host, or which contain end-end control packets, are placed on the TASK queue so TASK can decide how to forward them to their destinations. These packets are counted in reassembly while they are on the TASK queue.

A packet which is removed from the head of the TASK queue must either be sent to a neighboring IMP, or not. In the former case, it must be counted as store-and-forward; in the latter case as reassembly. However, there is a maximum value above which TASK is not allowed to increase these counters. If TASK cannot process a packet without increasing some counter above the maximum, TASK will refuse the packet. In the case of a packet which has arrived from a neighboring IMP (i.e., an uncounted packet), the packet will be dropped without acknowledgement, causing the neighboring IMP to send it again later. When other types of packets are refused, they are held in the IMP and resubmitted to TASK later.

7.2 Description of Buffer Counters

7.2.1 Allocated

Buffers which are allocated by the source-destination protocol are counted as follows. When the allocate request is granted, the allocate count is increased by 1 or by 8, as appropriate. However, the allocated buffers still count as free, not as reassembly (and they remain on the free queue.) When the first packet of the allocated message arrives, the allocate count is decreased by 1 or 8, and the reassembly count correspondingly increased. However, the free count is only decremented as each individual buffer is removed from the free queue, as each individual packet of the allocated message arrives.

7.2.2 Store and Forward

Let m be the number of modems (i.e. inter-IMP trunks) at a particular IMP. Then the most buffers which can be in use by modem-out at any one time (call it SFMAX) is 6 + 2m, if there is no 16-channel satellite line. If there is a 16-channel line, SFMAX is 13 + 2*(m+1). (The assumption is that a 16-channel line is entitled to 8 extra buffers for output and one extra buffer for input.)

There is also a minimum number of buffers which must be available for use by modem-out. That is, this number of buffers

(call it SFMIN) cannot, under any circumstances, be counted as reassembly or allocated. SFMIN is usually 3m, but there is one exception. Let B be the total number of buffers in the IMP. If (B-SFMIN) mod 8 = 0, then SFMIN is set to 3m-1. The reason for this will become apparent later. Note also that with both SFMIN and SFMAX, m is never taken as less than 2, even if the IMP is actually a stub.

The actual mechanism for adjusting the store-and-forward count is controlled by two parameters in the IMP known as MAXS and MINF. MAXS = SFMAX - m. MINF is set to a constant 3. When TASK processes a packet, and determines that the packet needs to go out a particular modem, it checks to see whether any other packets are either queued for, in transmission on, or awaiting acks on that modem. If not, then the packet is queued for the modem, and no adjustments are made to the store-and-forward count. This ensures that at least one packet can always be transmitted on each line, no matter how many packets are being transmitted on the other lines. (Failure to guarantee this can lead to direct store-and-forward lockup.) However, this means that one buffer per line is uncounted. If at least one other packet is either queued for or awaiting acks on that modem, then the following two checks are made:

a) Is the difference between the free count and the allocate count greater than MINF? (i.e are there MINF free buffers which have not been allocated?)

b) Is the store-and-forward count less then MAXS?

If both these questions are answered affirmatively, then the packet can be queued for the modem (if certain other checks are passed, such as availability of a logical channel.) If it is queued for the modem, then the store-and-forward count is incremented by one.

When a packet is acked, its buffer is freed, and the store-and-forward count is decremented by one. There is one exception. If the time comes to free a buffer, and at that time there are no other buffers queued for that modem or awaiting acknowledgement over that modem, then the store-and-forward count is not decremented. This special case is necessary since one buffer on each line is uncounted.

If a line goes down, so that all the packets transmitted on it or queued for it have to be re-routed, the store-and-forward count is incremented by one. This ensures that each re-routed buffer is counted as store-and-forward, even if it was previously uncounted.

In summary, the purpose of these mechanisms is to ensure that:

a) A certain minimum number of buffers are always available for modem-out.

b) A certain minimum number of buffers are never available for modem-out, so that modem-out cannot lock out other processes.

c) It is always possible to have at least one packet in flight on each inter-IMP line, regardless of the IMP's buffer utilization condition. This prevents direct store-and-forward lockups.

7.2.3 Reassembly

The reassembly count is controlled by the parameter MAXR. At initialization time, MAXR is set to B - SFMIN. Note that MAXR cannot be a multiple of 8, because of the way SFMIN is computed. Any process requesting a buffer which would have to be counted as reassembly must specify a parameter p. The following two checks must be made.

a) Is the sum of the allocate count and the reassembly count less than the difference between MAXR and p?

b) Is the difference between the free count and the allocate count less than the sum of MINF and p?

Only if these two questions are answered affirmatively can a reassembly buffer (or an allocate buffer) be obtained. When the two questions can be answered affirmatively, that means that the number of free buffers is at least MINF+p, and that at least p of those can still be taken for reassembly, if that should prove necessary.

The value of p that is used by a particular process when trying to get a buffer establishes that process's priority in obtaining buffers (i.e., a process which uses p=0 can get the last buffer, subject of course to the MAXR check). The processes which need reassembly buffers, and the values of p they use, are as follows:

1) Creating control packets: p=0

   A process which needs to send an end-to-end control packet is allowed to take the last reassembly buffer.

2) Ordinary host input: p=2

   The IMP will not accept a packet from a host unless two free buffers remain in the reassembly pool.

3) Unallocated single packet messages, including raw packets: p=2

   The IMP will not accept unallocated single packet messages for host output unless two free buffers remain in the reassembly pool.

4) Allocated single packet messages: p=1

   The IMP will not pre-allocate a buffer for a single packet message unless one free buffer remains in the reassembly pool.

5) Allocated multi-packet messages: p=9

   The IMP will not pre-allocate a buffer for a multi-packet message unless there are enough free buffers available for the reassembly pool to hold all

eight packets of the message, plus an additional free buffer.

6) Packets to be sent to the teletype in the NCC: p=8
When a packet has a checksum error, it will be sent to a special diagnostic teletype in the NCC, as long as there are still eight free buffers in the reassembly pool.

7) Packet core message received from a dead IMP: p=2
"Packet core" is the name of a special protocol used for communication between the NCC and an IMP which is in its loader/dumper.

8) Packet core message to be sent to a dead IMP: p=0

7.2.4 Uncounted

Ordinarily, the receipt of a packet on one of the IMP's lines causes a buffer to go from "free" to "uncounted". There is, however, one exception. When a packet is received, it is ordinarily placed on the TASK queue. However, before this is done, the IMP tries to get a free buffer to use for receiving the next packet. If there are no free buffers, then the buffer will not be placed on the TASK queue. Rather, it will be retained for the next input. The effect of this is to discard the packet without acknowledgement, causing the neighboring IMP to retransmit it at a later time.

7.3 Possible Improvements

Whenever a packet is received from a neighboring IMP, any acknowledgements which have been piggy-backed in that packet should always be processed, since processing of the acknowledgements may enable the IMP to free some of its store-and-forward buffers. However, when a packet arrives, and takes the last free buffer, the packet is discarded, and its piggybacked acknowledgements are not looked at. This may cause buffers to remain occupied when they could be freed. That is, the time when it is most important to free buffers as soon as possible is the time when buffers are least likely to be freed. There does not seem to be any reason not to process the acknowledgements, and failure to do so degrades the IMP's performance unnecessarily. Furthermore, it is not clear that the packet should be discarded at all, even after its acknowledgements are processed. Presumably, the reason for discarding the packet is so that its buffer can be re-used to receive the next input, which otherwise would be lost due to lack of buffers. However, there seems to be little point to throwing away one input so that the next can be received.

When TASK checks the buffer counts to see whether it is permissible to "move" a buffer into store-and-forward or reassembly, it does not check to see how the buffer is currently counted. Rather, it assumes, falsely, that all buffers in the

TASK queue are uncounted. This leads to two sorts of problems. Recall that TASK will almost always refuse a buffer if there are not 3 free buffers in the IMP. The apparent assumption is that a refused buffer will be freed, bringing the free buffer pool closer to its minimum acceptable value. This assumption is true in the case of uncounted buffers, which will be freed if they are refused by TASK, but it is not true in the case of buffers which are already counted in store-and-forward or reassembly. These buffers will not be freed if refused. Rather, they will simply be held and resubmitted to TASK later. Refusing such buffers therefore actually delays their being freed, which is exactly the opposite of what is intended.

A related problem arises if the store-and-forward count has already reached its maximum value. Then TASK will refuse all buffers which have to be forwarded to a neighboring IMP (except in the special case where no packet is queued for or in flight to that IMP.) TASK will refuse a buffer for this reason, even if the buffer is already counted in store-and-forward. Again, this is a counter-productive strategy. Accepting a buffer which is already counted in store-and-forward cannot possibly increase the store-and-forward count, so there is no reason to refuse it. Since buffers on the TASK queue which are already in store-and-forward are only those buffers which had to be re-routed due to a line failure, this problem causes the network to be slower to respond to a line failure than is necessary.

APPENDIX 1 -- SAMPLE TEST OUTPUT

In chapter 5 we discussed certain measurements which were made by keeping counters in the IMP. This appendix contains some sample computer output generated after collecting these counters. These measurements were made while the network was running in state V (see chapter 5), and they cover a period of slightly more than one hour. The meaning of each measurement is discussed in chapter 4. The interpretation of the text labels is as follows:

DUPL CURR UPDATES -- number of routing update messages received which, although current, are duplicates of previously received ones.

RCVD SPURIOUS RTS -- number of spurious retransmissions received.

UPDATES GENERATED -- number of routing update messages generated by this source IMP.

RETRANS n -- number of retransmissions of routing update messages from this IMP to IMP n.

DELAY CHANGES -- number of line updates which report changes in delay.

POSS. TREE CHANGES -- number of line updates which may cause changes in this IMP's shortest-path tree.

LINE UPDATES -- total number of line updates received.

UPDATE PACKETS -- total number of update packets processed.

AVERAGE QUEUE -- average length of routing update queue when new update arrives.

MAX QUEUE -- maximum length of routing update queue.

FRACTION OF DELAY CHANGES -- number of delay changes divided by number of line updates.

FRACTION POSS. CHANGES IN TREE -- number of line updates which may cause changes in tree divided by total number of line updates.

LINES PER PACKET -- number of line updates divided by number of update packets.

The output is ordered by IMP in decreasing order of distance from the NCC.

36. HAWAII

DUPL CURR UPDATES	34757
RCVD SPURIOUS RTS	33664
UPDATES GENERATED	137
RETRANS 15	963
DELAY CHANGES	5235
POSS. TREE CHANGES	2013
LINE UPDATES	11479
UPDATE PACKETS	4975
AVERAGE QUEUE	0.03457286
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.456050172
FRACTION POSS. CHANGES IN TREE = 0.175363704
LINES PER PACKET = 2.307336568

15. AMES15

DUPL CURR UPDATES	49980
RCVD SPURIOUS RTS	1033
UPDATES GENERATED	118
RETRANS 16	21
RETRANS 45	29
RETRANS 36	33913
RETRANS 11	9
DELAY CHANGES	5240
POSS. TREE CHANGES	1940
LINE UPDATES	11508
UPDATE PACKETS	4986
AVERAGE QUEUE	0.01965503
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.455335408
FRACTION POSS. CHANGES IN TREE = 0.168578378
LINES PER PACKET = 2.308062553

45. MOFFETT

DUPL CURR UPDATES	5129
RCVD SPURIOUS RTS	40
UPDATES GENERATED	94
RETRANS 15	7
RETRANS 34	6
DELAY CHANGES	5245
POSS. TREE CHANGES	1949
LINE UPDATES	11517
UPDATE PACKETS	4992
AVERAGE QUEUE	0.02944711
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.455413728
FRACTION POSS. CHANGES IN TREE = 0.169228091
LINES PER PACKET = 2.307091236

34. LBL

DUPL CURR UPDATES	10302
RCVD SPURIOUS RTS	129
UPDATES GENERATED	98
RETRANS 21	17
RETRANS 4	51
RETRANS 45	11
DELAY CHANGES	5257
POSS. TREE CHANGES	1911
LINE UPDATES	11574
UPDATE PACKETS	5016
AVERAGE QUEUE	0.03189792
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.454207703
FRACTION POSS. CHANGES IN TREE = 0.165111452
LINES PER PACKET = 2.307416200

21. LLL

DUPL CURR UPDATES	5180
RCVD SPURIOUS RTS	44
UPDATES GENERATED	94
RETRANS 2	9
RETRANS 34	13
DELAY CHANGES	5268
POSS. TREE CHANGES	1836
LINE UPDATES	11608
UPDATE PACKETS	5031
AVERAGE QUEUE	0.02663486
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.453824937
FRACTION POSS. CHANGES IN TREE = 0.158166781
LINES PER PACKET = 2.307294726

16. AMES16

DUPL CURR UPDATES	5173
RCVD SPURIOUS RTS	48
UPDATES GENERATED	92
RETRANS 15	26
RETRANS 51	6
DELAY CHANGES	5273
POSS. TREE CHANGES	1890
LINE UPDATES	11626
UPDATE PACKETS	5039
AVERAGE QUEUE	0.03076007
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.453552380
FRACTION POSS. CHANGES IN TREE = 0.162566654
LINES PER PACKET = 2.307203769

51. SRI51

DUPL CURR UPDATES	5160
RCVD SPURIOUS RTS	16
UPDATES GENERATED	83
RETRANS 16	29
RETRANS 2	28
DELAY CHANGES	5278
POSS. TREE CHANGES	1820
LINE UPDATES	11630
UPDATE PACKETS	5043
AVERAGE QUEUE	0.02716636
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.453826308
FRACTION POSS. CHANGES IN TREE = 0.156491830
LINES PER PACKET = 2.306166887

2. SRI2

DUPL CURR UPDATES	10311
RCVD SPURIOUS RTS	84
UPDATES GENERATED	89
RETRANS 51	11
RETRANS 32	11
RETRANS 21	27
DELAY CHANGES	5283
POSS. TREE CHANGES	1767
LINE UPDATES	11671
UPDATE PACKETS	5059
AVERAGE QUEUE	0.01858074
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.452660426
FRACTION POSS. CHANGES IN TREE = 0.151400901
LINES PER PACKET = 2.306977629

32. XEROX

DUPL CURR UPDATES	5255
RCVD SPURIOUS RTS	53
UPDATES GENERATED	90
RETRANS 2	47
RETRANS 43	1
DELAY CHANGES	5292
POSS. TREE CHANGES	1744
LINE UPDATES	11711
UPDATE PACKETS	5077
AVERAGE QUEUE	0.03210557
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.451882839
FRACTION POSS. CHANGES IN TREE = 0.148919813
LINES PER PACKET = 2.306677103

56. SUMEX

DUPL CURR UPDATES	5229
RCVD SPURIOUS RTS	41
UPDATES GENERATED	93
RETRANS 43	3
RETRANS 11	15
DELAY CHANGES	5299
POSS. TREE CHANGES	1857
LINE UPDATES	11717
UPDATE PACKETS	5081
AVERAGE QUEUE	0.01574493
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.452248856
FRACTION POSS. CHANGES IN TREE = 0.158487662
LINES PER PACKET = 2.306042075

43. TYMSHARE

DUPL CURR UPDATES	10386
RCVD SPURIOUS RTS	63
UPDATES GENERATED	78
RETRANS 32	42
RETRANS 33	11
RETRANS 56	30
DELAY CHANGES	5302
POSS. TREE CHANGES	1707
LINE UPDATES	11733
UPDATE PACKETS	5089
AVERAGE QUEUE	0.01434466
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.451887831
FRACTION POSS. CHANGES IN TREE = 0.145487084
LINES PER PACKET = 2.305560946

11. STANFORD

DUPL CURR UPDATES	10569
RCVD SPURIOUS RTS	59
UPDATES GENERATED	98
RETRANS 15	49
RETRANS 22	159
RETRANS 56	11
DELAY CHANGES	5310
POSS. TREE CHANGES	2010
LINE UPDATES	11806
UPDATE PACKETS	5120
AVERAGE QUEUE	0.04472656
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.449771299
FRACTION POSS. CHANGES IN TREE = 0.170252412
LINES PER PACKET = 2.305859327

22. ISI22

DUPL CURR UPDATES	10865
RCVD SPURIOUS RTS	211
UPDATES GENERATED	98
RETRANS 11	35
RETRANS 48	31
RETRANS 52	277
DELAY CHANGES	5328
POSS. TREE CHANGES	2041
LINE UPDATES	11831
UPDATE PACKETS	5130
AVERAGE QUEUE	0.01968810
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.450342312
FRACTION POSS. CHANGES IN TREE = 0.172512888
LINES PER PACKET = 2.306237697

35. ACCAT

DUPL CURR UPDATES	155
RCVD SPURIOUS RTS	87
UPDATES GENERATED	68
DELAY CHANGES	5336
POSS. TREE CHANGES	2029
LINE UPDATES	11854
UPDATE PACKETS	5141
AVERAGE QUEUE	0.03676327
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.450143396
FRACTION POSS. CHANGES IN TREE = 0.171165846
LINES PER PACKET = 2.305777072

52. ISI52

DUPL CURR UPDATES	10536
RCVD SPURIOUS RTS	41
UPDATES GENERATED	92
RETRANS 7	29
RETRANS 22	16
RETRANS 35	87
DELAY CHANGES	5345
POSS. TREE CHANGES	2017
LINE UPDATES	11894
UPDATE PACKETS	5160
AVERAGE QUEUE	0.02403100
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.449386239
FRACTION POSS. CHANGES IN TREE = 0.169581294
LINES PER PACKET = 2.305038690

7. RAND

DUPL CURR UPDATES	5361
RCVD SPURIOUS RTS	67
UPDATES GENERATED	89
RETRANS 23	23
RETRANS 52	14
DELAY CHANGES	5351
POSS. TREE CHANGES	1933
LINE UPDATES	11932
UPDATE PACKETS	5176
AVERAGE QUEUE	0.03265069
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.448457926
FRACTION POSS. CHANGES IN TREE = 0.162001334
LINES PER PACKET = 2.305254936

23. USC

DUPL CURR UPDATES	5355
RCVD SPURIOUS RTS	45
UPDATES GENERATED	86
RETRANS 7	38
RETRANS 1	10
DELAY CHANGES	5357
POSS. TREE CHANGES	1821
LINE UPDATES	11951
UPDATE PACKETS	5185
AVERAGE QUEUE	0.03529411
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.448247000
FRACTION POSS. CHANGES IN TREE = 0.152372181
LINES PER PACKET = 2.304917931

1. UCLA

DUPL CURR UPDATES	5569
RCVD SPURIOUS RTS	266
UPDATES GENERATED	84
RETRANS 23	22
RETRANS 33	15
DELAY CHANGES	5358
POSS. TREE CHANGES	1706
LINE UPDATES	11958
UPDATE PACKETS	5187
AVERAGE QUEUE	0.02024291
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.448068231
FRACTION POSS. CHANGES IN TREE = 0.142665997
LINES PER PACKET = 2.305378794

33. NPS

DUPL CURR UPDATES	11692
RCVD SPURIOUS RTS	631
UPDATES GENERATED	79
RETRANS 43	59
RETRANS 1	256
RETRANS 59	323
DELAY CHANGES	5363
POSS. TREE CHANGES	1665
LINE UPDATES	11973
UPDATE PACKETS	5194
AVERAGE QUEUE	0.03754331
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.447924494
FRACTION POSS. CHANGES IN TREE = 0.139062888
LINES PER PACKET = 2.305159687

48. AFWL

DUPL CURR UPDATES	5861
RCVD SPURIOUS RTS	304
UPDATES GENERATED	90
RETRANS 22	36
RETRANS 62	225
DELAY CHANGES	5372
POSS. TREE CHANGES	2079
LINE UPDATES	12024
UPDATE PACKETS	5218
AVERAGE QUEUE	0.03583748
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.446773111
FRACTION POSS. CHANGES IN TREE = 0.172904185
LINES PER PACKET = 2.304331064

4. UTAH

DUPL CURR UPDATES	5693
RCVD SPURIOUS RTS	200
UPDATES GENERATED	88
RETRANS 34	110
RETRANS 25	72
DELAY CHANGES	5376
POSS. TREE CHANGES	2029
LINE UPDATES	12041
UPDATE PACKETS	5225
AVERAGE QUEUE	0.03502392
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.446474537
FRACTION POSS. CHANGES IN TREE = 0.168507598
LINES PER PACKET = 2.304497599

25. DOCB

DUPL CURR UPDATES	5768
RCVD SPURIOUS RTS	238
UPDATES GENERATED	84
RETRANS 4	150
RETRANS 24	70
DELAY CHANGES	5381
POSS. TREE CHANGES	2033
LINE UPDATES	12070
UPDATE PACKETS	5237
AVERAGE QUEUE	0.03532556
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.445816069
FRACTION POSS. CHANGES IN TREE = 0.168434128
LINES PER PACKET = 2.304754614

24. GWC

DUPL CURR UPDATES	5705
RCVD SPURIOUS RTS	184
UPDATES GENERATED	88
RETRANS 12	35
RETRANS 25	169
DELAY CHANGES	5389
POSS. TREE CHANGES	1879
LINE UPDATES	12080
UPDATE PACKETS	5241
AVERAGE QUEUE	0.03186114
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.446109265
FRACTION POSS. CHANGES IN TREE = 0.155546352
LINES PER PACKET = 2.304903626

62. TEXAS

DUPL CURR UPDATES	5600
RCVD SPURIOUS RTS	250
UPDATES GENERATED	84
RETRANS 13	15
RETRANS 48	25
DELAY CHANGES	5400
POSS. TREE CHANGES	2056
LINE UPDATES	12118
UPDATE PACKETS	5260
AVERAGE QUEUE	0.01806083
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.445618078
FRACTION POSS. CHANGES IN TREE = 0.169664956
LINES PER PACKET = 2.303802251

13. GUNTER

DUPL CURR UPDATES	8111
RCVD SPURIOUS RTS	219
UPDATES GENERATED	157
RETRANS 62	34
RETRANS 53	87
RETRANS 17	55
DELAY CHANGES	5411
POSS. TREE CHANGES	2623
LINE UPDATES	12154
UPDATE PACKETS	5276
AVERAGE QUEUE	0.05079605
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.445203214
FRACTION POSS. CHANGES IN TREE = 0.215813718
LINES PER PACKET = 2.303639054

53. EGLIN

DUPL CURR UPDATES	5726
RCVD SPURIOUS RTS	185
UPDATES GENERATED	152
RETRANS 13	10
RETRANS 38	105
DELAY CHANGES	5423
POSS. TREE CHANGES	2616
LINE UPDATES	12186
UPDATE PACKETS	5290
AVERAGE QUEUE	0.03440453
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.445018872
FRACTION POSS. CHANGES IN TREE = 0.214672572
LINES PER PACKET = 2.303591609

28. ARPA

DUPL CURR UPDATES	179
RCVD SPURIOUS RTS	84
UPDATES GENERATED	95
DELAY CHANGES	5434
POSS. TREE CHANGES	2554
LINE UPDATES	12232
UPDATE PACKETS	5312
AVERAGE QUEUE	0.04442770
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.444244593
FRACTION POSS. CHANGES IN TREE = 0.208796598
LINES PER PACKET = 2.302710771

17. MITRE

DUPL CURR UPDATES	11127
RCVD SPURIOUS RTS	82
UPDATES GENERATED	129
RETRANS 13	194
RETRANS 19	11
RETRANS 28	85
DELAY CHANGES	5441
POSS. TREE CHANGES	2515
LINE UPDATES	12258
UPDATE PACKETS	5322
AVERAGE QUEUE	0.04998120
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.443873375
FRACTION POSS. CHANGES IN TREE = 0.205172128
LINES PER PACKET = 2.303269386

38. BRAGG

DUPL CURR UPDATES	5817
RCVD SPURIOUS RTS	230
UPDATES GENERATED	152
RETRANS 26	11
RETRANS 53	97
DELAY CHANGES	5452
POSS. TREE CHANGES	2604
LINE UPDATES	12290
UPDATE PACKETS	5336
AVERAGE QUEUE	0.05284857
MAX QUEUE	3

FRACTION OF DELAY CHANGES = 0.443612679
FRACTION POSS. CHANGES IN TREE = 0.211879573
LINES PER PACKET = 2.303223371

26. PENTAGON

DUPL CURR UPDATES	5668
RCVD SPURIOUS RTS	80
UPDATES GENERATED	124
RETRANS 38	123
RETRANS 19	2
DELAY CHANGES	5462
POSS. TREE CHANGES	2495
LINE UPDATES	12329
UPDATE PACKETS	5354
AVERAGE QUEUE	0.04650728
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.443020507
FRACTION POSS. CHANGES IN TREE = 0.202368393
LINES PER PACKET = 2.302764177

19. NBS

DUPL CURR UPDATES	10941
RCVD SPURIOUS RTS	28
UPDATES GENERATED	106
RETRANS 17	28
RETRANS 29	22
RETRANS 26	69
DELAY CHANGES	5476
POSS. TREE CHANGES	2388
LINE UPDATES	12348
UPDATE PACKETS	5363
AVERAGE QUEUE	0.04586984
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.443472623
FRACTION POSS. CHANGES IN TREE = 0.193391636
LINES PER PACKET = 2.302442550

12. DTI

DUPL CURR UPDATES	11105
RCVD SPURIOUS RTS	98
UPDATES GENERATED	110
RETRANS 59	11
RETRANS 24	122
RETRANS 55	30
DELAY CHANGES	5476
POSS. TREE CHANGES	1829
LINE UPDATES	12367
UPDATE PACKETS	5371
AVERAGE QUEUE	0.01582573
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.442791298
FRACTION POSS. CHANGES IN TREE = 0.147893585
LINES PER PACKET = 2.302550673

55. ANL

DUPL CURR UPDATES	5691
RCVD SPURIOUS RTS	94
UPDATES GENERATED	107
RETRANS 47	96
RETRANS 12	23
DELAY CHANGES	5482
POSS. TREE CHANGES	1936
LINE UPDATES	12384
UPDATE PACKETS	5378
AVERAGE QUEUE	0.02975083
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.442667946
FRACTION POSS. CHANGES IN TREE = 0.156330741
LINES PER PACKET = 2.302714705

47. WPAFB

DUPL CURR UPDATES	5741
RCVD SPURIOUS RTS	128
UPDATES GENERATED	108
RETRANS 14	68
RETRANS 55	65
DELAY CHANGES	5489
POSS. TREE CHANGES	2031
LINE UPDATES	12409
UPDATE PACKETS	5390
AVERAGE QUEUE	0.03302411
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.442340224
FRACTION POSS. CHANGES IN TREE = 0.163671523
LINES PER PACKET = 2.302226305

14. CMU

DUPL CURR UPDATES	11134
RCVD SPURIOUS RTS	114
UPDATES GENERATED	128
RETRANS 47	33
RETRANS 50	18
RETRANS 18	47
DELAY CHANGES	5497
POSS. TREE CHANGES	2089
LINE UPDATES	12429
UPDATE PACKETS	5401
AVERAGE QUEUE	0.01870024
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.442272096
FRACTION POSS. CHANGES IN TREE = 0.168074660
LINES PER PACKET = 2.301240444

18. RADC

DUPL CURR UPDATES	5665
RCVD SPURIOUS RTS	84
UPDATES GENERATED	109
RETRANS 10	47
RETRANS 14	25
DELAY CHANGES	5508
POSS. TREE CHANGES	1992
LINE UPDATES	12459
UPDATE PACKETS	5412
AVERAGE QUEUE	0.03215077
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.442090049
FRACTION POSS. CHANGES IN TREE = 0.159884415
LINES PER PACKET = 2.302106380

10. LINCOLN

DUPL CURR UPDATES	11086
RCVD SPURIOUS RTS	62
UPDATES GENERATED	102
RETRANS 44	44
RETRANS 18	37
RETRANS 37	11
DELAY CHANGES	5511
POSS. TREE CHANGES	1832
LINE UPDATES	12485
UPDATE PACKETS	5423
AVERAGE QUEUE	0.01512078
MAX QUEUE	1

FRACTION OF DELAY CHANGES = 0.441409677
FRACTION POSS. CHANGES IN TREE = 0.146736077
LINES PER PACKET = 2.302231192

44. MIT44

DUPL CURR UPDATES	5584
RCVD SPURIOUS RTS	44
UPDATES GENERATED	79
RETRANS 6	36
DELAY CHANGES	5513
POSS. TREE CHANGES	1859
LINE UPDATES	12497
UPDATE PACKETS	5428
AVERAGE QUEUE	0.03389830
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.441145867
FRACTION POSS. CHANGES IN TREE = 0.148755699
LINES PER PACKET = 2.302321195

6. MIT6

DUPL CURR UPDATES	108
RCVD SPURIOUS RTS	36
UPDATES GENERATED	72
DELAY CHANGES	5519
POSS. TREE CHANGES	1857
LINE UPDATES	12525
UPDATE PACKETS	5441
AVERAGE QUEUE	0.02297371
MAX QUEUE	1

FRACTION OF DELAY CHANGES = 0.440638720
FRACTION POSS. CHANGES IN TREE = 0.148263469
LINES PER PACKET = 2.301966547

20. DCEC

DUPL CURR UPDATES	208
RCVD SPURIOUS RTS	111
UPDATES GENERATED	97
DELAY CHANGES	5525
POSS. TREE CHANGES	2290
LINE UPDATES	12539
UPDATE PACKETS	5450
AVERAGE QUEUE	0.03669724
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.440625235
FRACTION POSS. CHANGES IN TREE = 0.182630188
LINES PER PACKET = 2.300733923

50. DARCOM

DUPL CURR UPDATES	11205
RCVD SPURIOUS RTS	27
UPDATES GENERATED	134
RETRANS 14	23
RETRANS 29	11
RETRANS 20	111
DELAY CHANGES	5534
POSS. TREE CHANGES	2263
LINE UPDATES	12578
UPDATE PACKETS	5466
AVERAGE QUEUE	0.04024880
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.439974546
FRACTION POSS. CHANGES IN TREE = 0.179917313
LINES PER PACKET = 2.301134228

29. ABERDEEN

DUPL CURR UPDATES	11204
RCVD SPURIOUS RTS	63
UPDATES GENERATED	143
RETRANS 46	30
RETRANS 19	18
RETRANS 50	8
DELAY CHANGES	5544
POSS. TREE CHANGES	2290
LINE UPDATES	12625
UPDATE PACKETS	5486
AVERAGE QUEUE	0.01239518
MAX QUEUE	1

FRACTION OF DELAY CHANGES = 0.439128711
FRACTION POSS. CHANGES IN TREE = 0.181386135
LINES PER PACKET = 2.301312327

46. RUTGERS

DUPL CURR UPDATES	5705
RCVD SPURIOUS RTS	62
UPDATES GENERATED	110
RETRANS 29	30
RETRANS 60	17
DELAY CHANGES	5551
POSS. TREE CHANGES	2144
LINE UPDATES	12644
UPDATE PACKETS	5496
AVERAGE QUEUE	0.03074963
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.439022451
FRACTION POSS. CHANGES IN TREE = 0.169566586
LINES PER PACKET = 2.300582170

60. CORADCOM

DUPL CURR UPDATES	5707
RCVD SPURIOUS RTS	46
UPDATES GENERATED	123
RETRANS 46	32
RETRANS 58	11
DELAY CHANGES	5557
POSS. TREE CHANGES	1957
LINE UPDATES	12660
UPDATE PACKETS	5502
AVERAGE QUEUE	0.02653580
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.438941538
FRACTION POSS. CHANGES IN TREE = 0.154581353
LINES PER PACKET = 2.300981402

58. NYU

DUPL CURR UPDATES	5759
RCVD SPURIOUS RTS	85
UPDATES GENERATED	128
RETRANS 9	17
RETRANS 60	29
DELAY CHANGES	5565
POSS. TREE CHANGES	1725
LINE UPDATES	12685
UPDATE PACKETS	5513
AVERAGE QUEUE	0.02122256
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.438707128
FRACTION POSS. CHANGES IN TREE = 0.135987386
LINES PER PACKET = 2.300925016

37. DEC

DUPL CURR UPDATES	5700
RCVD SPURIOUS RTS	58
UPDATES GENERATED	105
RETRANS 10	15
RETRANS 9	5
DELAY CHANGES	5573
POSS. TREE CHANGES	1659
LINE UPDATES	12713
UPDATE PACKETS	5525
AVERAGE QUEUE	0.01520361
MAX QUEUE	1

FRACTION OF DELAY CHANGES = 0.438370168
FRACTION POSS. CHANGES IN TREE = 0.130496338
LINES PER PACKET = 2.300995469

59. SCOTT

DUPL CURR UPDATES	12459
RCVD SPURIOUS RTS	504
UPDATES GENERATED	105
RETRANS 12	41
RETRANS 9	135
RETRANS 33	642
DELAY CHANGES	5579
POSS. TREE CHANGES	1639
LINE UPDATES	12737
UPDATE PACKETS	5536
AVERAGE QUEUE	0.02366329
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.438015222
FRACTION POSS. CHANGES IN TREE = 0.128680221
LINES PER PACKET = 2.300758600

9. HARVARD

DUPL CURR UPDATES	17132
RCVD SPURIOUS RTS	150
UPDATES GENERATED	98
RETRANS 40	26
RETRANS 58	75
RETRANS 37	47
RETRANS 59	150
DELAY CHANGES	5584
POSS. TREE CHANGES	1482
LINE UPDATES	12750
UPDATE PACKETS	5542
AVERAGE QUEUE	0.02959220
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.437960773
FRACTION POSS. CHANGES IN TREE = 0.116235293
LINES PER PACKET = 2.300613403

63. BBN63

DUPL CURR UPDATES	208
RCVD SPURIOUS RTS	133
UPDATES GENERATED	74
RETRANS 40	1
DELAY CHANGES	5586
POSS. TREE CHANGES	1423
LINE UPDATES	12769
UPDATE PACKETS	5550
AVERAGE QUEUE	0.01675675
MAX QUEUE	1

FRACTION OF DELAY CHANGES = 0.437465727
FRACTION POSS. CHANGES IN TREE = 0.111441772
LINES PER PACKET = 2.300720691

40. BBN40

DUPL CURR UPDATES	11353
RCVD SPURIOUS RTS	22
UPDATES GENERATED	78
RETRANS 49	23
RETRANS 9	1
RETRANS 63	133
DELAY CHANGES	5594
POSS. TREE CHANGES	1419
LINE UPDATES	12783
UPDATE PACKETS	5556
AVERAGE QUEUE	0.03779697
MAX QUEUE	5

FRACTION OF DELAY CHANGES = 0.437612444
FRACTION POSS. CHANGES IN TREE = 0.111006803
LINES PER PACKET = 2.300755858

49. RCC49

DUPL CURR UPDATES	5656
RCVD SPURIOUS RTS	19
UPDATES GENERATED	72
RETRANS 5	124
DELAY CHANGES	5596
POSS. TREE CHANGES	1396
LINE UPDATES	12791
UPDATE PACKETS	5559
AVERAGE QUEUE	0.03004137
MAX QUEUE	2

FRACTION OF DELAY CHANGES = 0.437495112
FRACTION POSS. CHANGES IN TREE = 0.109139237
LINES PER PACKET = 2.300953388

5. RCC5

DUPL CURR UPDATES	196
RCVD SPURIOUS RTS	123
UPDATES GENERATED	73
RETRANS 49	1
DELAY CHANGES	5602
POSS. TREE CHANGES	1403
LINE UPDATES	12805
UPDATE PACKETS	5566
AVERAGE QUEUE	0.02317642
MAX QUEUE	1

FRACTION OF DELAY CHANGES = 0.437485352
FRACTION POSS. CHANGES IN TREE = 0.109566573
LINES PER PACKET = 2.300574898

APPENDIX 2 -- TRAFFIC TESTS

This appendix contains the results of 5 experiments performed on 1/4/79. The experiments are discussed in chapter 6. In all of these experiments, node 64 was sending 20 packets per second to node 59, with each packet containing 1192 bits. Node 59 was set to refuse to acknowledge a certain fraction of the traffic arriving from node 64. Nodes 56 and 43 sent tagged packets to node 9.

a) In this experiment, node 43 sent 10 packets per second to node 9, with each packet containing 1192 bits. Node 59 was set to reject 80% of the traffic arriving from node 64.

SOURCE: 43 COUNT: 1836 DELAY: 317.51 MS.

PATH: 43-32- 2-21-34- 4-25-24-12-59- 9
COUNT: 879 DELAY: 312.30 48%

PATH: 43-56-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 801 DELAY: 316.80 44%

PATH: 43-56-11-22-48-62-13-53-38-26-19-29-46-58- 9
COUNT: 75 DELAY: 358.07 4%

PATH: 43-56-11-15-45-34- 4-25-24-12-59- 9
COUNT: 80 DELAY: 342.64 4%

PATH: 43-32- 2-51-16-15-45-34- 4-25-24-12-59- 9
COUNT: 1 DELAY: 423.20 0%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH =11.08 HOPS

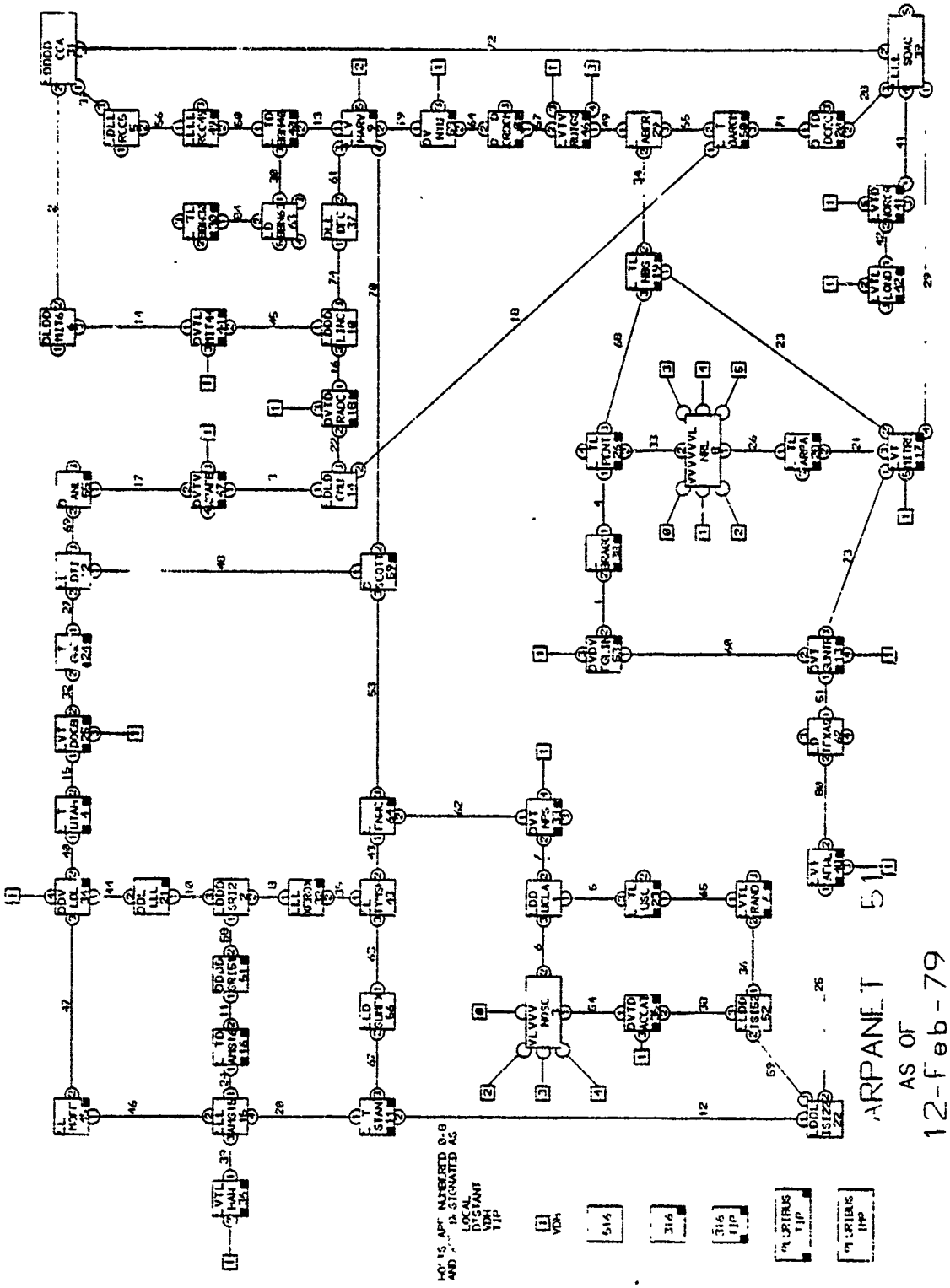


Figure A2-1

b) In this experiment, node 43 sent 10 packets per second to node 9, and node 59 was set to reject 67% of the traffic arriving from node 64.

SOURCE: 43 COUNT: 2010 DELAY: 499.68 MS.

PATH: 43-64-59- 9
COUNT: 557 DELAY:1103.97 28%

PATH: 43-32- 2-21-34- 4-25-24-12-59- 9
COUNT: 772 DELAY: 247.20 38%

PATH: 43-56-11-15-45-34- 4-25-24-12-59- 9
COUNT: 320 DELAY: 272.36 16%

PATH: 43-56-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 324 DELAY: 303.74 16%

PATH: 43-56-11-22-48-62-13-53-38-26-19-29-46-58- 9
COUNT: 36 DELAY: 347.82 2%

PATH: 43-32- 2-32-43-64-59- 9
COUNT: 1 DELAY: 522.40LOOP 0%

PERCENTAGE OF LOOPING PACKETS = 0.05%
AVERAGE PATH LENGTH = 8.61 HOPS

c) In this experiment, node 43 sent 20 packets per second to node 9, and node 59 was set to reject 80% of the traffic arriving from node 64.

SOURCE: 43 COUNT: 3798 DELAY: 422.02 MS.

PATH: 43-64-59- 9
COUNT: 84 DELAY:2140.69 2%

PATH: 43-32- 2-21-34- 4-25-24-12-59- 9
COUNT: 1192 DELAY: 387.08 31%

PATH: 43-56-11-15-45-34- 4-25-24-12-59- 9
COUNT: 573 DELAY: 386.76 15%

PATH: 43-56-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 1310 DELAY: 358.25 34%

PATH: 43-56-11-22-48-62-13-53-38-26-19-29-46-58- 9
COUNT: 525 DELAY: 409.79 14%

PATH: 43-64-33- 1-23- 7-52-22-48-62-13-17-19-29-46-58- 9
COUNT: 82 DELAY: 505.93 2%

PATH: 43-32- 2-51-16-15-45-34- 4-25-24-12-59- 9
COUNT: 29 DELAY: 440.19 1%

PATH: 43-32- 2-51-16-15-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 3 DELAY: 433.60 0%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH =11.40 HOPS

d) In this experiment, node 43 sent 20 packets per second to node 9, and node 59 was set to reject 67% of the traffic arriving from node 64.

SOURCE: 43 COUNT: 4151 DELAY: 531.07 MS.

PATH: 43-56-11-22-52- 7-23- 1-33-64-59- 9
COUNT: 79 DELAY:2015.56 2%

PATH: 43-32- 2-21-34- 4-25-24-12-59- 9
COUNT: 1751 DELAY: 272.28 42%

PATH: 43-56-11-22-52-22-48-62-13-17-19-29-46-58- 9
COUNT: 1 DELAY: 380.80LOOP 0%

PATH: 43-56-11-22-52- 7-23- 7-52-22-48-62-13-17-19-29-46-58- 9
COUNT: 1 DELAY: 480.00LOOP 0%

PATH: 43-56-11-22-52- 7-23- 1-23- 7-52-22-48-62-13-17-19-29-46-58- 9
COUNT: 4 DELAY: 913.60LOOP 0%

PATH: 43-64-59- 9
COUNT: 787 DELAY:1223.50 19%

PATH: 43-56-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 778 DELAY: 367.79 19%

PATH: 43-56-11-15-45-34- 4-25-24-12-59- 9
COUNT: 717 DELAY: 317.14 17%

PATH: 43-32- 2-32-43-64-59- 9
COUNT: 2 DELAY: 640.80LOOP 0%

PATH: 43-56-11-22-52- 7-23- 7-52-22-48-62-13-53-38-26-19-29-46-58- 9

```
                COUNT:    8     DELAY:2820.00LOOP    0%

      PATH: 43-56-11-22-52- 7-23- 1-23- 7-52-22-48-62-13-
                                    53-38-26-19-29-46-58- 9
                COUNT:    7     DELAY:3164.91LOOP    0%

      PATH: 43-56-11-22-52- 7-52-22-48-62-13-53-38-26-
                                          19-29-46-58- 9
                COUNT:    1     DELAY:2482.40LOOP    0%

      PATH: 43-56-11-22-52- 7-23- 1-33- 1-23- 7-52-22-
                          48-62-13-53-38-26-19-29-46-58- 9
                COUNT:    6     DELAY:3726.40LOOP    0%

      PATH: 43-64-43-32- 2-21-34- 4-25-24-12-59- 9
                COUNT:    8     DELAY:1875.20LOOP    0%

      PATH: 43-56-11-56-43-64-59- 9
                COUNT:    1     DELAY:1712.80LOOP    0%

PERCENTAGE OF LOOPING PACKETS = 0.94%
AVERAGE PATH LENGTH = 9.32   HOPS
```

e) In this experiment, nodes 56 and 43 each sent 10 packets per second to node 9, and node 59 was set to reject 80% of the traffic arriving from node 64.

SOURCE: 43 COUNT: 2673 DELAY: 357.01 MS.

PATH: 43-64-59- 9
COUNT: 4 DELAY:1933.60 0%

PATH: 43-56-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 892 DELAY: 378.25 33%

PATH: 43-32- 2-21-34- 4-25-24-12-59- 9
COUNT: 1572 DELAY: 333.87 59%

PATH: 43-56-11-15-45-34- 4-25-24-12-59- 9
COUNT: 162 DELAY: 412.83 6%

PATH: 43-56-11-22-48-62-13-53-38-26-19-29-46-58- 9
COUNT: 43 DELAY: 405.30 2%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH =10.78 HOPS

SOURCE: 56 COUNT: 2673 DELAY: 352.32 MS.

PATH: 56-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 1631 DELAY: 325.54 61%

PATH: 56-43-56-11-22-48-62-13-17-19-29-46-58- 9
COUNT: 4 DELAY:5269.60LOCP 0%

PATH: 56-43-64-59- 9
COUNT: 3 DELAY:6365.33 0%

PATH: 56-43-32- 2-21-34- 4-25-24-12-59- 9
COUNT: 89 DELAY: 397.29 3%

PATH: 56-11-15-45-34- 4-25-24-12-59- 9
COUNT: 903 DELAY: 353.33 34%

PATH: 56-11-22-48-62-13-53-38-26-19-29-46-58- 9
COUNT: 43 DELAY: 376.82 2%

PERCENTAGE OF LOOPING PACKETS = 0.15%
AVERAGE PATH LENGTH =10.69 HOPS

TOTAL MESSAGES FROM ALL SOURCES = 5346
PERCENT LOOPING PACKETS = 0.07%

APPENDIX 3 -- INSTABILITY TESTS

This appendix contains the results of 5 experiments performed on 11/21/78. The experiments are discussed in chapter 6. They were performed on our lab network, pictured in Figure A3-1.

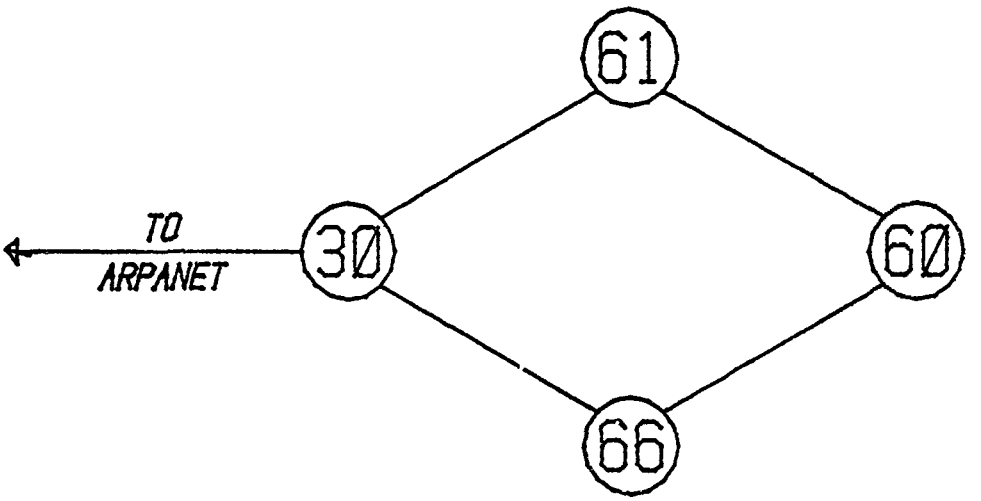


Figure A3-1

a) For this experiment, nodes 60, 61, and 66 each sent 10 packets per second to node 30. Each packet contained 1192 bits.

SOURCE: 60 COUNT: 1331 DELAY: 108.96 MS.

PATH: 60-61-30
COUNT: 525 DELAY: 145.19 39%

PATH: 60-66-30
COUNT: 803 DELAY: 74.86 60%

PATH: 60-61-60-66-30
COUNT: 3 DELAY:2897.07LOOP 0%

PERCENTAGE OF LOOPING PACKETS = 0.23%
AVERAGE PATH LENGTH = 2.00 HOPS

SOURCE: 61 COUNT: 1317 DELAY: 63.04 MS.

PATH: 61-30
COUNT: 1310 DELAY: 57.30 99%

PATH: 61-60-66-30
COUNT: 7 DELAY:1135.77 1%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.01 HOPS

SOURCE: 66 COUNT: 1349 DELAY: 55.58 MS.

PATH: 66-30
COUNT: 1339 DELAY: 51.53 99%

PATH: 66-60-61-30
COUNT: 10 DELAY: 598.72 1%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.01 HOPS

TOTAL MESSAGES FROM ALL SOURCES = 3997
PERCENT LOOPING PACKETS = 0.08%

b) For this experiment, nodes 60, 61, and 66 each sent 20 packets per second to node 30. Each packets contained 1192 bits.

```
SOURCE:  60      COUNT:     372      DELAY:1555.02  MS.

         PATH: 60-66-30
                    COUNT:  243      DELAY:1386.86        65%

         PATH: 60-61-30
                    COUNT:  124      DELAY:1841.39        33%

         PATH: 60-66-60-61-30
                    COUNT:    1      DELAY:1913.60LOOP     0%

         PATH: 60-61-60-66-30
                    COUNT:    4      DELAY:2803.20LOOP     1%

PERCENTAGE OF LOOPING PACKETS = 1.34%
AVERAGE PATH LENGTH = 2.03   HOPS


SOURCE:  61      COUNT:     493      DELAY: 643.88  MS.

         PATH: 61-30
                    COUNT:  464      DELAY: 563.89        94%

         PATH: 61-60-66-30
                    COUNT:   29      DELAY:1923.70         6%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.12   HOPS
```

SOURCE: 66 COUNT: 594 DELAY: 438.29 MS.

PATH: 66-30
COUNT: 592 DELAY: 436.79 100%

PATH: 66-60-61-30
COUNT: 2 DELAY: 881.60 0%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.01 HOPS

TOTAL MESSAGES FROM ALL SOURCES = 1459
PERCENT LOOPING PACKETS = 0.34%

c) For this experiment, nodes 60 and 66 each sent 10 packets per second to node 30. Each packet contained 1192 bits.

SOURCE: 60 COUNT: 1197 DELAY: 50.60 MS.

PATH: 60-66-30
COUNT: 586 DELAY: 52.15 49%

PATH: 60-61-30
COUNT: 611 DELAY: 49.12 51%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 2.00 HOPS

SOURCE: 66 COUNT: 1192 DELAY: 25.06 MS.

PATH: 66-30
COUNT: 1192 DELAY: 25.06 100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.00 HOPS

TOTAL MESSAGES FROM ALL SOURCES = 2389
PERCENT LOOPING PACKETS = 0.00%

d) For this experiment, nodes 60 and 66 each sent 20 packets per second to node 30. Each packet contained 1192 bits.

SOURCE: 60 COUNT: 885 DELAY: 820.94 MS.

PATH: 60-66-30
COUNT: 481 DELAY: 729.95 54%

PATH: 60-61-30
COUNT: 389 DELAY: 906.83 44%

PATH: 60-66-60-61-30
COUNT: 8 DELAY:1148.00LOOP 1%

PATH: 60-61-60-66-30
COUNT: 7 DELAY:1926.40LOOP 1%

PERCENTAGE OF LOOPING PACKETS = 1.69%
AVERAGE PATH LENGTH = 2.03 HOPS

SOURCE: 66 COUNT: 893 DELAY: 371.48 MS.

PATH: 66-30
COUNT: 849 DELAY: 336.34 95%

PATH: 66-60-61-30
COUNT: 44 DELAY:1049.60 5%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.10 HOPS

TOTAL MESSAGES FROM ALL SOURCES = 1778
PERCENT LOOPING PACKETS = 0.84%

e) For this experiment, we used the network pictured in Figure A3-2. Nodes 60, 61, and 66 each sent 10 packets per second to node 30.

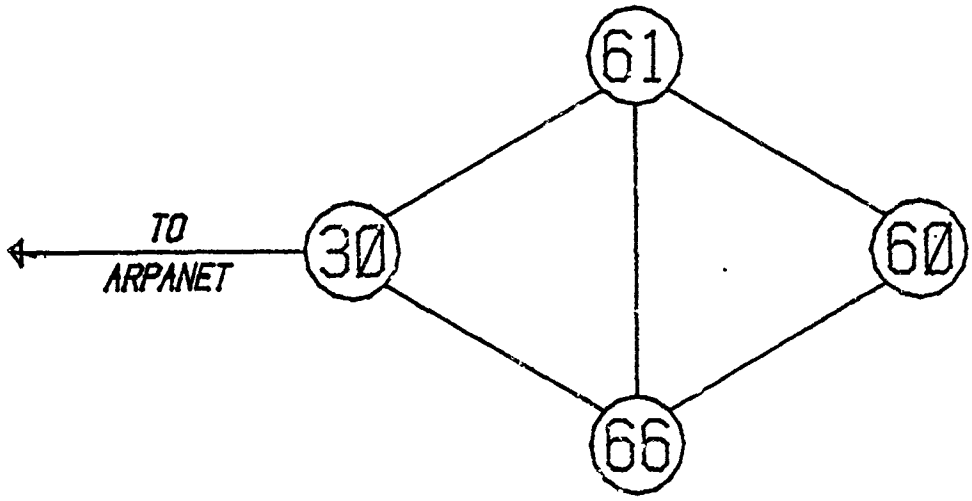


Figure A3-2

SOURCE: 60 COUNT: 1051 DELAY: 104.66 MS

PATH	COUNT	DELAY	
60-61-66-30	60	148.91	6%
60-66-30	719	72.19	68%
60-61-30	196	138.25	19%
60-61-60-66-30	3	364.80LOOP	0%
60-66-61-30	71	277.93	7%
60-66-61-60-66-30	2	516.00LOOP	0%

PERCENTAGE OF LOOPING PACKETS = 0.48%
AVERAGE PATH LENGTH = 2.14 HOPS

SOURCE: 61 COUNT: 1034 DELAY: 77.34 MS.

PATH	COUNT	DELAY	
61-66-30	293	112.17	28%
61-30	715	60.17	69%
61-60-66-30	26	157.17	3%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.33 HOPS

SOURCE: 66 COUNT: 1051 DELAY: 80.58 MS.

PATH: 66-30
COUNT: 879 DELAY: 44.34 84%

PATH: 66-61-30
COUNT: 134 DELAY: 272.53 13%

PATH: 66-61-60-66-30
COUNT: 3 DELAY: 514.13LOOP 0%

PATH: 66-60-61-30
COUNT: 35 DELAY: 218.56 3%

PERCENTAGE OF LOOPING PACKETS = 0.29%
AVERAGE PATH LENGTH = 1.20 HOPS

TOTAL MESSAGES FROM ALL SOURCES = 3136
PERCENT LOOPING PACKETS = 0.26%

APPENDIX 4 -- INSTABILITY/OVERLOAD TESTS

This appendix contains the results of an experiment done on 2/27/79. Nodes 26, 38, and 17 each sent 10 packets per second to node 19. Nodes 13 and 53 each sent 20 packets per second to node 19. All packets were 1192 bits long. During this experiment, the line between nodes 13 and 62 was removed from operation. The network, as it was during our experiments, is pictured in Figure A4-1.

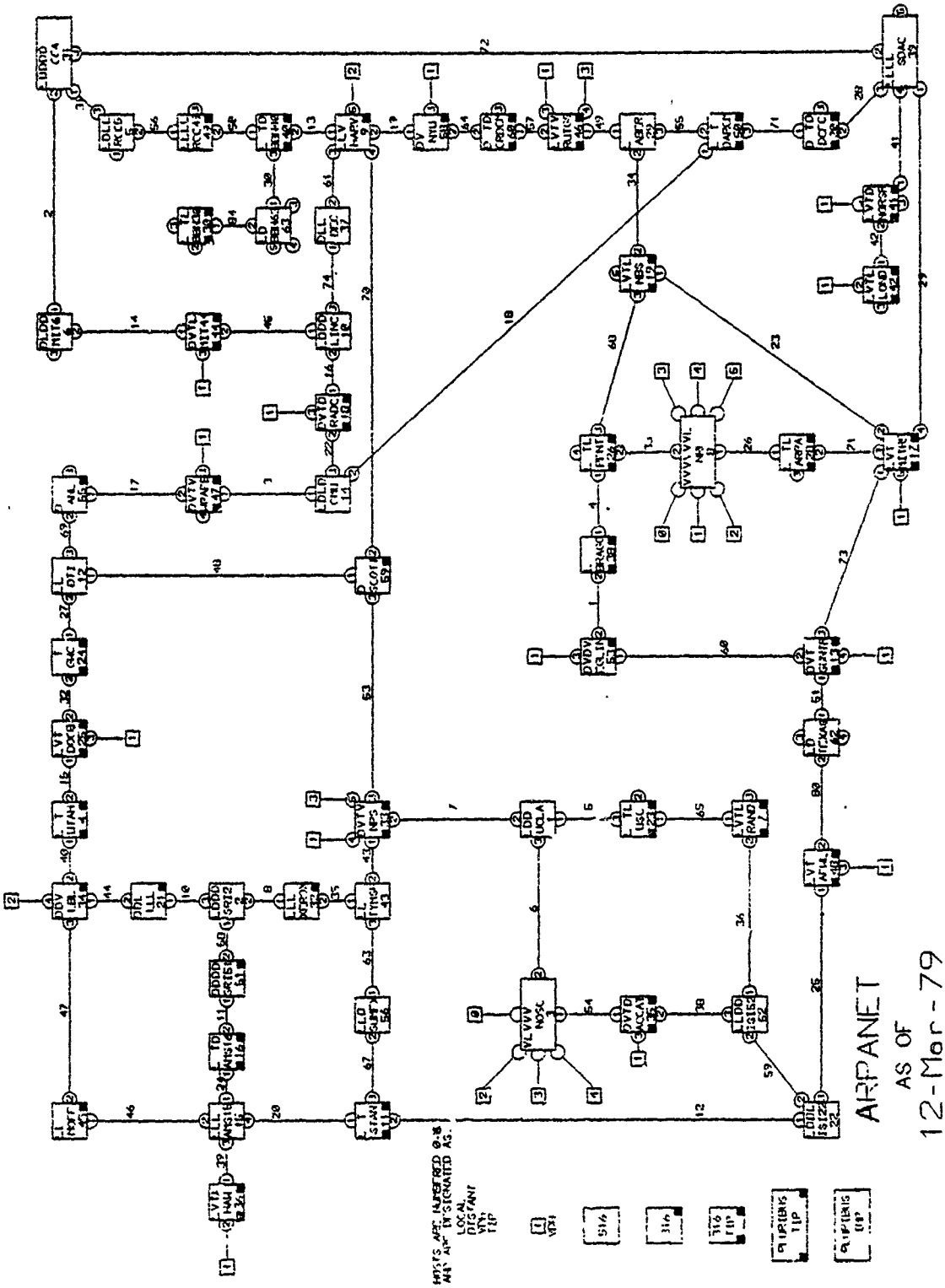


Figure A4-1

SOURCE: 13 COUNT: 3503 DELAY: 357.75 MS.

PATH: 13-17-19
COUNT: 2869 DELAY: 186.07 82%

PATH: 13-53-38-26-19
COUNT: 634 DELAY:1134.63 18%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 2.36 HOPS

SOURCE: 26 COUNT: 2084 DELAY: 219.85 MS.

PATH: 26-19
COUNT: 1950 DELAY: 121.12 94%

PATH: 26-38-53-13-17-19
COUNT: 130 DELAY:1690.02 6%

PATH: 26-38-26-19
COUNT: 4 DELAY: 567.20LOOP 0%

PERCENTAGE OF LOOPING PACKETS = 0.19%
AVERAGE PATH LENGTH = 1.25 HOPS

SOURCE: 17 COUNT: 2536 DELAY: 86.24 MS.

PATH: 17-19
COUNT: 2533 DELAY: 85.89 100%

PATH: 17-13-53-38-26-19
COUNT: 3 DELAY: 382.40 0%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.00 HOPS

SOURCE: 53 COUNT: 3878 DELAY: 555.83 MS.

PATH	COUNT	DELAY	
53-38-26-19	2630	471.59	68%
53-13-17-19	1225	729.06	32%
53-13-53-38-26-19	17	1049.27LOOP	0%
53-38-53-13-17-19	6	718.40LOOP	0%

PERCENTAGE OF LOOPING PACKETS = 0.59%
AVERAGE PATH LENGTH = 3.01 HOPS

SOURCE: 38 COUNT: 1471 DELAY: 515.56 MS.

PATH: 38-53-13-17-19
COUNT: 286 DELAY:1387.12 19%

PATH: 38-26-19
COUNT: 1178 DELAY: 297.89 80%

PATH: 38-26-38-53-13-17-19
COUNT: 2 DELAY: 373.60LOOP 0%

PATH: 38-53-38-26-19
COUNT: 3 DELAY:1793.07LOOP 0%

PATH: 38-53-13-53-38-26-19
COUNT: 2 DELAY:2315.20LOOP 0%

PERCENTAGE OF LOOPING PACKETS = 0.48%
AVERAGE PATH LENGTH = 2.40 HOPS

APPENDIX 5 -- MODERATE LOAD TESTS

This appendix contains the results of two experiments, one performed on 3/1/79 and the second performed on 12/21/78.

a) For this experiment, nodes 38, 26, 13, and 17 each sent 20 packets per second to node 19. Each packet contained 1112 bits. Only the data from nodes 38 and 17 are shown here. During this experiment, the line between nodes 13 and 62 was removed from operation. The network is pictured in Figure A4-1.

```
SOURCE:  17      COUNT:    2437      DELAY:  35.60  MS.

         PATH: 17-19
                        COUNT: 2437      DELAY:  35.60       100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.00    HOPS


SOURCE:  38      COUNT:    2481      DELAY:  55.50  MS.

         PATH: 38-26-19
                        COUNT: 2481      DELAY:  55.50       100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 2.00    HOPS


TOTAL MESSAGES FROM ALL SOURCES = 4919
PERCENT LOOPING PACKETS = 0.00%
```

b) For this experiment, each of nodes 2, 16, 21, 34, 45, and 51 sent approximately 2.5 packets per second to node 15. Each packet contained 1192 bits.

SOURCE: 2 COUNT: 937 DELAY: 45.52 MS.

PATH: 2-51-16-15
COUNT: 937 DELAY: 45.52 100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 3.00 HOPS

SOURCE: 16 COUNT: 949 DELAY: 7.13 MS.

PATH: 16-15
COUNT: 949 DELAY: 7.13 100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.00 HOPS

SOURCE: 21 COUNT: 937 DELAY: 73.77 MS.

PATH: 21- 2-51-16-15
COUNT: 453 DELAY: 71.27 48%

PATH: 21-34-45-15
COUNT: 484 DELAY: 76.12 52%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 3.48 HOPS

SOURCE: 34 COUNT: 938 DELAY: 48.94 MS.

PATH: 34-45-15
COUNT: 938 DELAY: 48.94 100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 2.00 HOPS

SOURCE: 45 COUNT: 945 DELAY: 23.27 MS.

PATH: 45-15
COUNT: 945 DELAY: 23.27 100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 1.00 HOPS

SOURCE: 51 COUNT: 941 DELAY: 33.66 MS.

PATH: 51-16-15
COUNT: 941 DELAY: 33.66 100%

PERCENTAGE OF LOOPING PACKETS = 0.00%
AVERAGE PATH LENGTH = 2.00 HOPS

TOTAL MESSAGES FROM ALL SOURCES = 5647
PERCENT LOOPING PACKETS = 0.00%