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On Optimizing the routing algorithms of a GSM Infrastructure Network for supporting time critical applications

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INTRODUCTION

GSM (Groupe Speciale Mobile) is a digital cellular mobile radio system that services millions of customers and covers almost all European countries [1], [2]. It has a hierarchical structure, which consists of a number of base station controllers (BSC's) at the lower level, which are connected to a number of mobile switching centers (MSC's). MSCs are responsible for routing a call from the BSC that made the request, to the BSC that will receive the call. The routing of a call is done either through the Public Switched Digital Network (PSDN) or through the MSCs themselves. BSCs and MSCs form a mesh network, called GSM Infrastructure Network (GIN), that is based on medium and/or high speed point-to-point links.

Although GSM is mainly used for supporting voice calls and voice-related services, there is an increasing demand to support new services, with diverse characteristics, that must be routed through the GIN. These services are provided either to the GSM customers or to the GSM operators for monitoring and controlling the network infrastructure, thus increasing the reliability and maintainability of the whole system. Such applications include transferring of Maintenance and Control (M&C) data, telemetry data, compressed still-pictures and low-bit rate video etc. Due to their nature, some of these applications have strict timing response requirements, so the delay for routing their data through the GIN is very critical. Information related to the security of the BSCs equipment or the malfunction of some key devices (e.g. the BSC power supply) must be transferred really fast to the network 'maintenance and support center', so the proper actions will be taken on time. For example, during physical disasters, a lot of alarm data are generated and they must be handled reliably and efficiently by the network. The purpose of this work is to determine how the GSM infrastructure has to be organised, in order to be able to support such applications.

During the design phase of the inter-networking devices of such a GIN, the selection of the proper routing algorithm becomes the main issue. Although there is a large number of routing algorithms available in the literature and many of them are used in commercial systems, the selection of the proper algorithm, if such an algorithm exists for supporting the application requirements, was not a straight-forward process. In order to predict the network performance by using different routing algorithms, there were available only two analysis tools, mathematical analysis and simulation analysis. Due to the complexity of such a network, using mathematical methods was considered impractical (too much complicated and time consuming), so the only solution was to use simulation techniques, in order to decrease the system development time and to find the best available solution. On the other hand, real traffic measurements, which can be used for predicting the availability of temporary links, can be exploited only by using them on a simulation environment. The tool that was used for defining the most appropriate solution, is the simulation tool COMNET III of CACI.

This paper presents the model of such a GIN network, the models of its traffic sources, the network topology and the functionality of its nodes. Simulation studies are presented for determining how the various routing algorithms affect the network performance. Based on these results, the best routing algorithm for such an application was specified.

NETWORK DESCRIPTION AND MODELING

The first task in developing the optimum architecture for a given network is to accurately determine the services provided to the end users and their specifications in terms of timing and traffic load. In our case the requested services had to be routed through an already installed and operational network, with given physical characteristics and limitations. Our effort was to accurately model, not only the incoming traffic, but the physical characteristics of the network as well, using the simulation tool. Our purpose was to test the ability of the network to route alarm data from the nodes of the GIN to a central BSC with minimal delay, while the

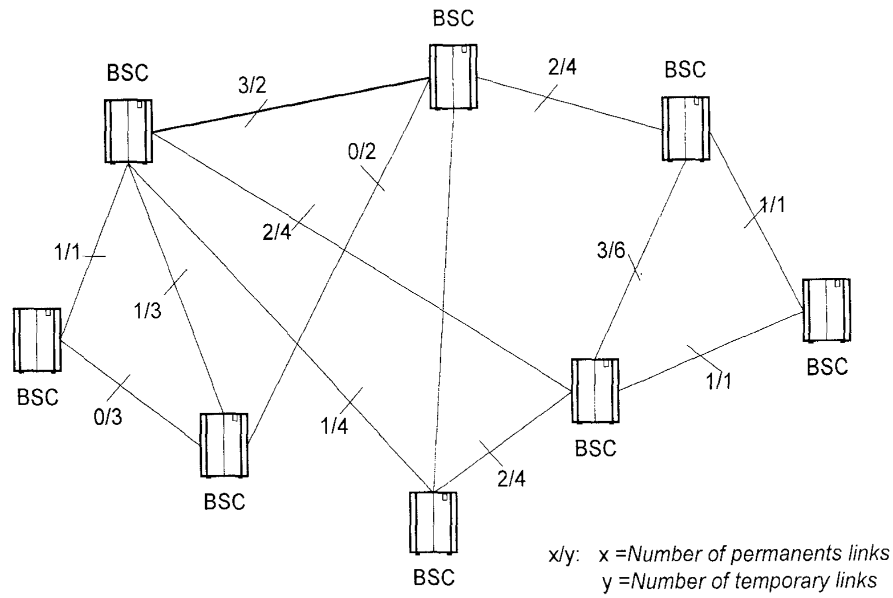

Fig. 1. A GIN example having a number of permanent and temporary channels

resources are simultaneously used for telemetry data transfer and user calls.

The GSM Infrastructure Network (GIN) consists of a number of hundreds or thousands of nodes (BSCs and MSCs) which are interconnected by medium or high-speed point-to-point links. An example of such a link, consists of a number of groups of 64 kbps channels for transferring mainly voice calls (like the 32 channels of the 2 Mbps E1 link). The connections that are available for use by the new services, are divided to permanent and temporary links. The permanent links are always available for application data, irrespective to the number of requests for voice calls, while the temporary links are available whenever there is no demand for transferring digitized voice (Fig. 1).

The connections at the backbone rely mainly on permanent links, while the sub-networks use more temporary links. The experience of the installation of such a network in Greece, says that more than 300 nodes are used for the infrastructure network, and that this number increases rapidly. This complicated structure was modeled for four basic situations that differ in the percentage of use of the temporary lines by the GSM users, and vary from 'full use' to 'no use' of the extra bandwidth. Both permanent and temporary links were modeled as 64kbps point to point lines, with 0.1ms transmission delay, and zero bit-error-rate (BER).

Concerning the BSC and MSC nodes, we were only interested to model the part responsible for routing traffic through the GIN. Since routing is a standalone function, we were free to choose the characteristics of the routing device from a great variety of commercially available routers. Therefore, we selected an internal bus rate of 3Mbps and a buffer size of 128Kbytes per port, for all peripheral stations, common for the popular Cisco2500 router. For the central station, the internal bus rate was 10Mbps.

Message Name	*IAT Distribution*	*IAT Mean*	*Size Distribution*	*Size Mean*	*Priority*	*Flow Control*
RP	exponential	1 sec	constant	128 bytes	1	Yes
RD	exponential	-	exponential	1 kbytes	1	Yes
AD	exponential	180 sec	exponential	1 kbytes	10	Yes

Table 1: Packets and packet parameters

The simulation of the data traffic includes the modeling of the sequence of events in the telemetry environment and the amount of data per transaction. In such a case, a central station requests information from each peripheral station periodically. The peripheral stations respond to the requests with data containing information of their current status. In this structure, we would like to test both the timing characteristics of the telemetry transfer, but, most significantly, we would like to test the ability of the network to transfer an alarm data from the most remote station to the central station, with an acceptable delay. Alarm data are created in case of node malfunction, and are transmitted asynchronously to the previously described polling process.

The model of transferring data through a point-to-point link is that of a M/M/k queue. This means that the inter-arrival times (IAT) between consecutive packets follow a Poison distribution [3]. Therefore, we model the time interval periods with an exponential distribution function. In order to calculate the load that each data source adds to the network, we are mainly concerned to the mean of the exponential function and the average data message size. In our case study, the three main traffic sources characteristics are shown in Table 1. RP are the request packets created in the central station, RD are the report data that contain the peripheral station status data, and AD are the alarm data.

ROUTING AND FLOW CONTROL ALGORITHMS

Routing Algorithms

The substance of all the routing algorithms used in any network, is to make a decision on the next hop of an incoming packet, based on certain information collected from the links. These are the metrics of the routing protocol, which mainly differentiate one algorithm from the other. A metric could be any measurable parameter of the network, like channel utilization, unused bandwidth, packet delay, bit error rate, packet delay, etc. Other characteristic parameters of each routing algorithm are the table update interval period, and the congestion control deviation.

The congestion control deviation is the metric that enables load balancing. When there are several hops listed in the routing table for routes of the same total weighting factor, generally only the route at the top of the list will be picked. It is an arbitrary choice that determines which of the routes will be at the top. If this metric has the value 0, then the operation is as described above. If the deviation percent is positive then hops for routes that are within the deviation percent of the shortest route are considered equivalent and routed over on a round robin basis. The main parameters of the routing protocols tested are described below.

The *Internal Gateway Routing Protocol* (IGRP) metric calculates compound route weighting factors based on bandwidth, utilization and delay metrics. When a routing table update is made, the compound penalty calculated for each link is given by the formula:

$$K1 * bandwidth\ factor + K2 * bandwidth\ factor/(256\text{-}load) + K3 * Delay\ Factor$$

were

$K1$, $K2$ and $K3$ for standard IGRP are equal to 1,0 and 1 respectively.

Bandwidth Factor = 1010 / (Bandwidth)

Load = Utilization Percentage * 255

Delay Factor = delay in units of 10 usecs

The Bandwidth for the network links is expressed in bits per second, computed automatically by COMNET III based on link parameters.

Under IGRP delay, weighting factors are additive across a route, but bandwidth factors are only calculated for the least bandwidth link on the route. The time period between updates is given in the "Routing Update Interval" field. The Deviation Percent is based on total composite metric of a route.

The *Open Shortest Path First* (OSPF) protocol calculates route weighting factors based on integer penalty values applied to links by the use of penalty tables. The deviation percent is based on the total penalty metric of a route.

The *Shortest Measured Delay* (SMD) protocol calculates route weighting factors based on packet delays seen on each link. Packet delay includes queuing time in the port output buffer, the transmission time, and the propagation delay. It does not include port input buffer delays or node switching delays. The deviation percent is based on this metric.

Concerning the *Predefined Routing Tables* (PRD), all possible routes from each node to each destination are pre-entered to the nodes, and are not altered throughout the testing period. The routing protocol selects a route from 1 or more of these routes. Multiple alternate routes may be entered. Partial routes may also be entered which route the packet to some intermediate node. At each intermediate node, one or more routes must be available to forward the packet towards the destination. The tables may be constructed on the basis of any constant metric of the network. In our case, Node-By-Node routing was chosen, and the tables were formed on the basis of minimum hop routing, since the propagation delay and BER metrics were chosen to be the same for all the links.

When a packet arrives at a node, and there are several routes that lead to the destination, a route selection criterion is required to determine which alternate to use. These criteria include:

1. First Available
2. Maximum Unused Bandwidth
3. Minimum Queue
4. Random List
5. Round Robin

The alternate routes can de either primary, or secondary. When a routing decision is made, the primary routes are inspected and a route is found. If no route can be found from the primary routes, then the secondary routes are evaluated.

Flow Control Algorithms

Flow control is a mechanism for adjusting the flow of information from the source to the destination. Flow control algorithms' characteristics will not be discussed in detail here. Emphasis is given to those aspects that affect our study.

There are three error control options available to the flow-control mechanisms. The selective repeat will retransmit just the packet that is blocked. The go-back-N option will retransmit the entire window of unacknowledged packets, which for fixed and SNA-pacing windows is the entire window, but for sliding windows it is just that portion of the window that is still outstanding. The fast-recovery option applies just to sliding windows (such as TCP/IP) that would retransmit the entire window unless it detects multiple duplicate acknowledgments which indicate that only one packet needs to be retransmitted.

Some sliding window protocols have a provision to delay the acknowledgment in order to improve the chance to piggy-back data on the acknowledgment. These delays were modeled with the "Hold" method. The Hold-1 option allows for the delay to be interrupted if there is a second acknowledgment waiting, while the Hold-all option accumulates all acknowledgments until the delay has expired. When an acknowledgment is delayed, the transmitted acknowledgment will acknowledge all packets that have been received during that delay.

Since the sliding-window algorithms acknowledge each packet (or at least most packets), they have an opportunity to measure round-trip packet delay and use that delay to adjust the retransmission time-out timer. There are two options for estimating the round-trip time (RTT) and using that estimate for the retransmission time-out. The first option estimates the round-trip time and uses a multiple of that time for the retransmission time. The second option estimates both the average round trip time and its standard deviation and then uses the average plus a multiple of the standard deviation for the retransmission time.

SIMULATION RESULTS

Concerning the topology and physical layer characteristics of the network, we chose to model a certain case study, which is shown in Fig.1. The respective network model in the COMNET III simulator is shown in Fig. 2. Those network characteristics

No	*Protocol Name*	*Metrics (K1,K2,K3)*	*Table Update Interval*	*Congestion Control Deviation*
1	Standard IGRP	1,0,1	0.1sec	0%
2	Standard IGRP	1,0,1	0.1sec	25%
3	Standard IGRP	1,0,1	0.1sec	34%
4	OSPF	-	auto	0%
5	OSPF	-	auto	25%
6	OSPF	-	auto	34%
7	Shortest Measured Delay	-	0.1sec	0%
8	Shortest Measured Delay	-	0.1sec	25%
9	Shortest Measured Delay	-	0.1sec	34%
10	PRD - First Available	-	auto	-
11	PRD - Maximum Unused Bandwidth	-	auto	-
12	PRD - Minimum Queue	-	auto	-
13	PRD - Random List	-	auto	-
14	PRD - Round Robin	-	auto	-
15	IGRP - UD	1,2,1	0.1sec	0%
16	IGRP - UD	1,2,1	0.1sec	25%
17	IGRP - UD	1,2,1	0.1sec	34%

Table 2. Routing algorithms parameters

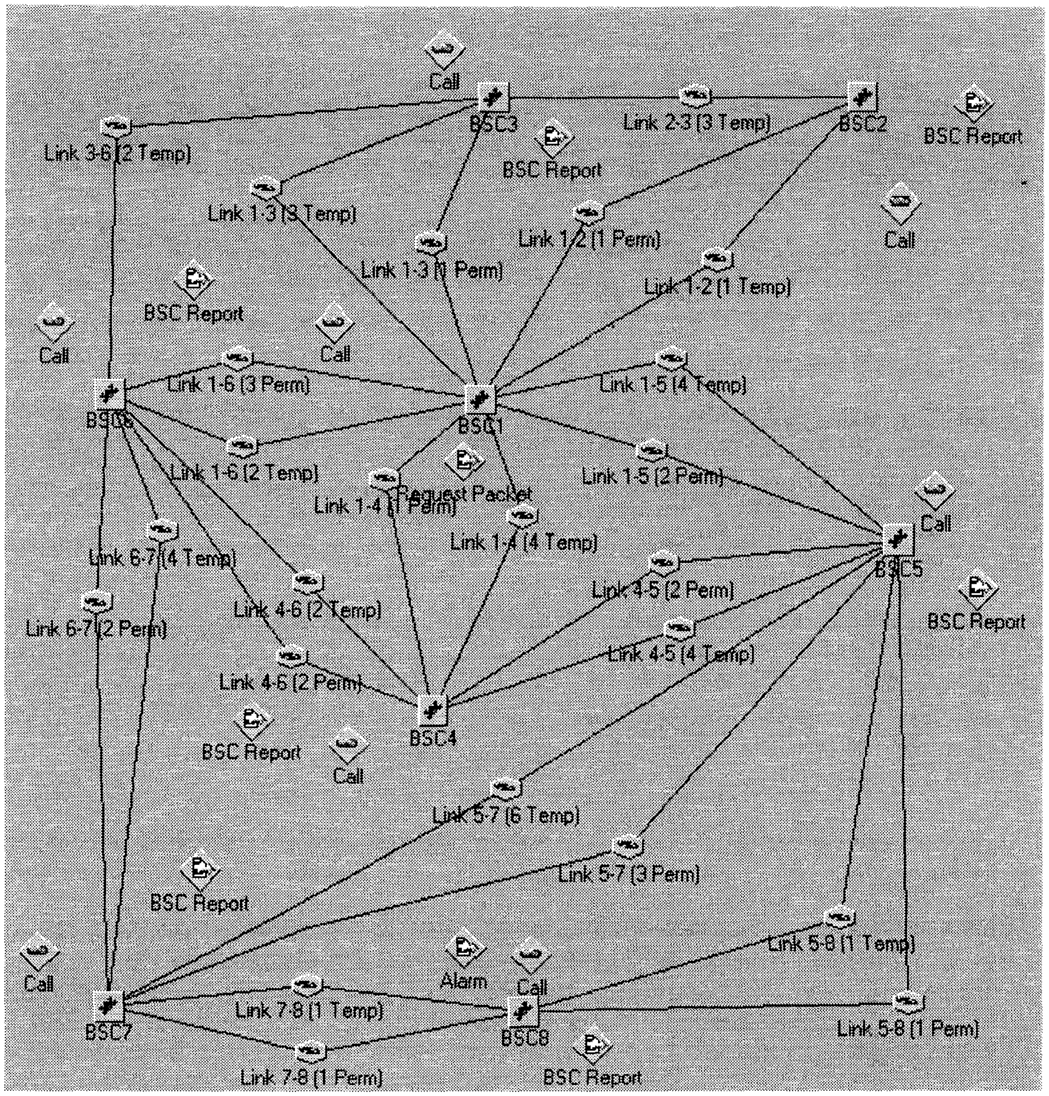

Fig.2: Simulation Model

remained unaltered for all the tests that we performed during the development of this study. We experimented on routing algorithms, having to choose among a great variety of algorithms, like standard and non-standard Interior Gateway Routing Protocol (IGRP), Open Shortest Path First (OSPF), shortest measured delay routing, minimum queue routing etc. We also tested the influence the characteristics of each algorithm, such as metrics, table update intervals, and congestion control mechanisms. These tests were made, with constant transport layer characteristics, in order to be able to compare the different routing protocols on a constant basis. The full catalogue of the algorithms simulated, together with their main parameters is shown in Table 2.

With the test of some primary simulation results, we concluded that with the use of UDP/IP with additional flow control for the request packets and TCP/IP for the report and alarm data, we had acceptable end-to-end delays for packet transfer. We also modeled the error control acknowledgments for the transport layer and set the retransmission time-out to 1500ms. For packet delays at nodes we chose an average delay of 0,022 ms per packet, which is common for IP routing.

The test for the selection of the optimum routing protocol, among the ones described above, were performed for four basic situations that differ in the percentage of use of the temporary lines by the GSM users, and vary from full to no use of the extra bandwidth.

a. *No Traffic Model.* Utilization percentage of the network's bandwidth for calls is 0%.

b. *Low Traffic Model.* Utilization percentage of the temporary lines' bandwidth for calls is 27,7%.

c. *Medium Traffic Model.* Utilization percentage of the temporary lines' bandwidth for calls is 71,65%.

d. *High Traffic Model.* Utilization percentage of the temporary lines' bandwidth for calls is 100%.

The timing characteristics of each protocol tested in each of the cases described above, are shown in Figures 3 to 6.

As it is shown in these graphs, there is a great number of routing protocols suitable for 'no traffic' and low traffic models. However, since our most crucial criteria in selecting the appropriate algorithm is to have suitable delays when no temporary line is available, only few of these are acceptable. The routing algorithm that we finally agreed on, is the First Available algorithm, which was based on predefined routing tables. These tables were formed on the basis of minimum hop routing, since the propagation delay and BER metrics are the same for all the links. Additionally, the permanent links were placed higher in the routing tables, in order to be given higher priority in routing decisions. The "First Available" algorithm seems to have a moderately good response in all simulated models, and moreover, one of the best timing characteristics in the high traffic model.

Since we have concluded on the most appropriate routing algorithm, additional test should be made on the whole protocol suite (flow control, retransmission time-outs, window size), in order to optimize the message delay characteristics of the transfer. Two of the most common protocols were tested, TCP/IP and UDP/IP. A great number of different flow-control parameters are simulated, including fixed window, sliding window and SNA pacing algorithms.

The first set of experiments were made on the no traffic model, were the performance of the selected routing algorithm is moderate (Fig.7). From a first analysis of these tests, we see that the UDP/IP protocol with no end-to-end flow control gives by far the best results (protocol 4, Fig.7). Unfortunately, end-to-end flow control is obligatory in this application. Since many of the remaining algorithms have moderately the same response in this model, we selected the algorithms listed in Table 3, since they had the best timing characteristics. Then we tested them in the medium call traffic model, were the average of the bandwidth of the temporary lines used by calls, is 71.65%. The results of these tests (Fig.8), lead us to the conclusion that the best solution is to maintain the UDP/IP packet size, and add

Number	*Basic Protocol*	*Window*	*Window Size*	*Method*	*Time-out*	*Holding Method*
1	UDP	Sliding Window	17	Fast Recovery	RTT+M*Deviation	1
2	UDP	Sliding Window	24	Fast Recovery	RTT+M*Deviation	1
3	UDP	Sliding Window	11	Fast Recovery	RTT+M*Deviation	All
4	UDP	Sliding Window	14	Fast Recovery	RTT+M*Deviation	All
5	UDP	Sliding Window	17	Fast Recovery	RTT+M*Deviation	All
6	UDP	Sliding Window	24	Fast Recovery	RTT+M*Deviation	All
7	UDP	Sliding Window	17	Fast Recovery	2*RTT	1
8	UDP	Sliding Window	24	Fast Recovery	2*RTT	1
9	UDP	Sliding Window	17	Fast Recovery	2*RTT	none
10	UDP	Sliding Window	24	Fast Recovery	2*RTT	none
11	UDP	Fixed Window	24	Go Back N	500ms	-

Table 3: Transport protocols parameters

flow control with the following characteristics: Sliding window algorithm, window size 17, fast recovery method, time-out twice the round-trip time, and no holding method.

CONCLUSIONS

This paper presented how simulation techniques were used to define the implementation parameters of a large-scale telecommunication system, thus minimizing the time and cost of such an implementation. Simulation was used for measuring the performance of various routing algorithms on a specific network and for estimating the network response time under extreme conditions.

A commercially available network simulation tool was used and the model of an existing network was developed. Then the network was examined under various traffic loads and based on the simulation results, it was measured how various routing algorithms affect its performance. Based on the simulation results and the application requirements, the proper routing algorithm was selected and most of the transport protocol aspects were determined.

References

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[2] Nathan J. Muller, 1995, "Wireless Data Networking", Artech House Publishers, Boston, USA.

[3] Jeremiah F. Hayes, 1984, "Modeling and Analysis of Computer Communications Networks", Plenum Press, New York, USA.

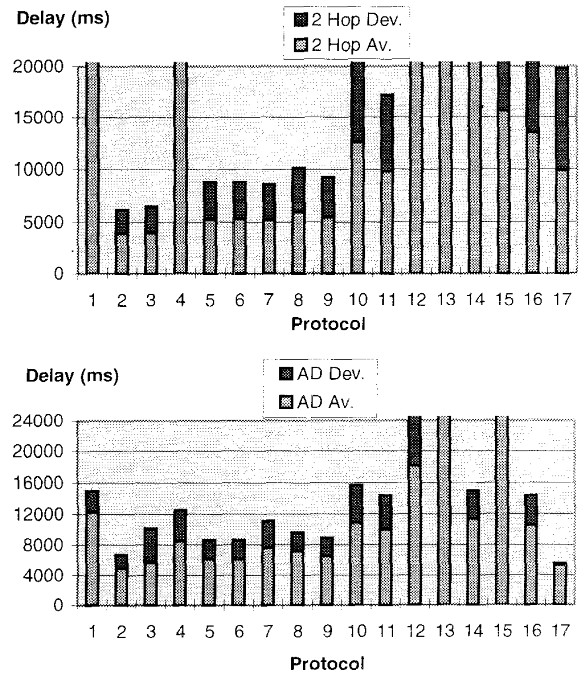

Fig. 3: No Traffic Model Delay Characteristics: 2 hop and alarm delays

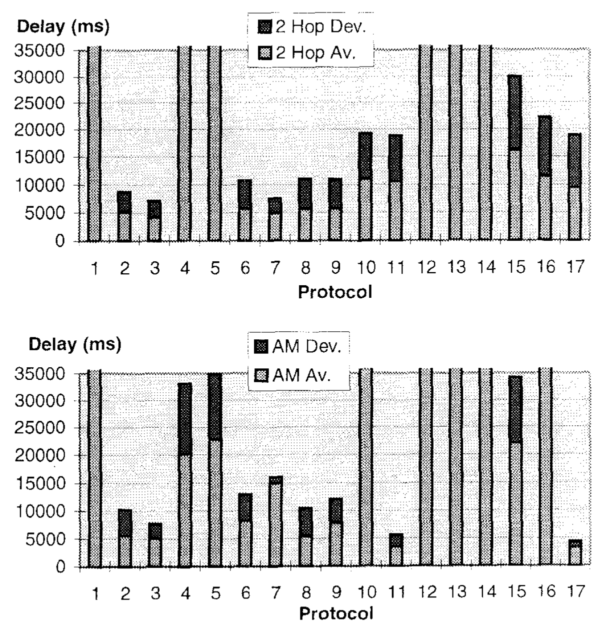

Fig.4: Low Traffic Model Delay Characteristics: 2hop and alarm delays

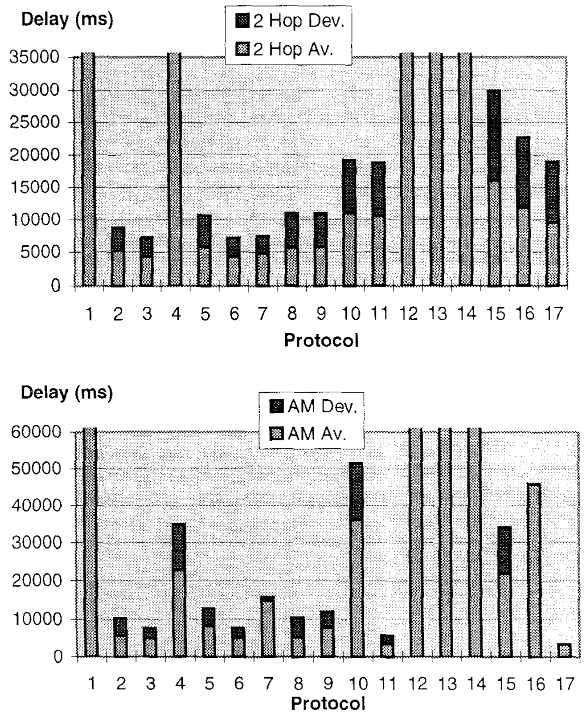

Fig.5: Medium Traffic Model Delay Characteristics: 2 hop and alarm delays

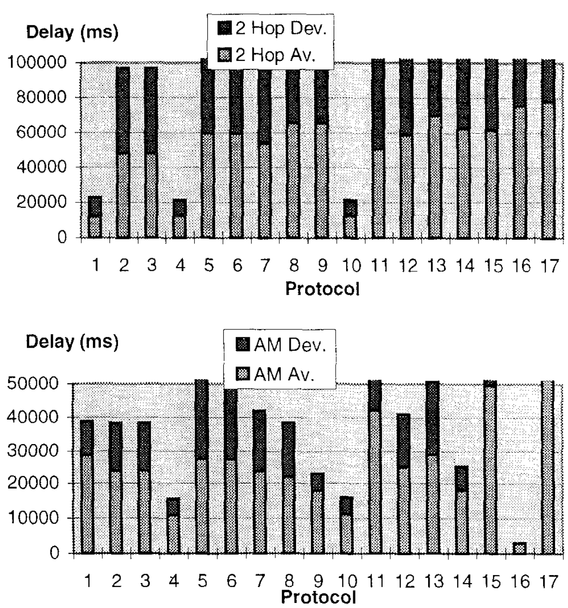

Fig.6: High Traffic model Delay Characteristics: 2 hop and alarm delays

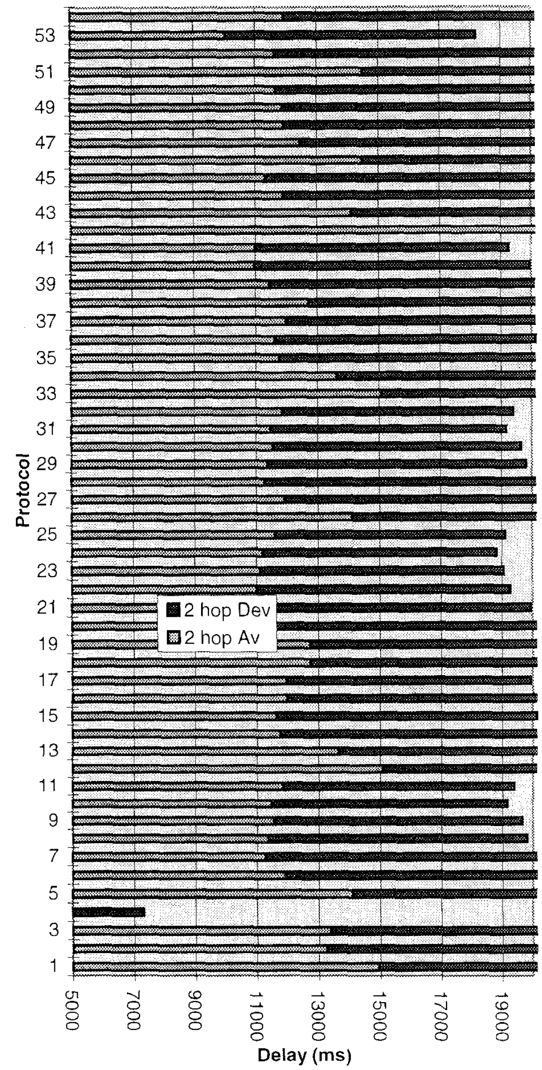

Fig.7: Simulation results on transport layer protocols for the low traffic model.

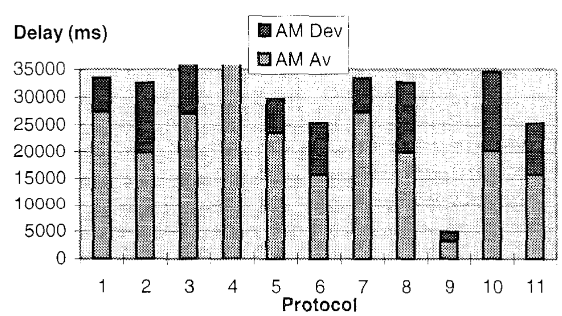

Fig.8: Simulation results on transport layer protocols for the low traffic model.