An Experimental Study Examining the Ethernet Local Computer Network: Towards Enhancing Packet-Voice Communication

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Abstract--- In data applications, there has been remarkable success in the use of local computer networks. An example of such applications is a case of file transfers. In the recent past, there has been growing effort to utilize such networks in voice applications. In this study, the main aim was to describe a voice protocol that could be utilized on packet-switching local networks. The experimental setting involved the use of a 3 Mbps Ethernet network. From the findings, it was evident that when parameters are selected appropriately, the selected Ethernet can support approximately 40 simultaneous voice conversations occurring at 64-Kbps, with the resultant quality also proving acceptable.

Keywords--- Voice Communication, Packet-voice, Measurement, Local Computer Network, Ethernet.

I. INTRODUCTION

IN the recent past, shared resource and computer interconnection local computer networks have increased in popularity. The increase in popularity has been felt in campuses or buildings and other small areas [1]. Also, the networks have supported various applications, including electronic mail and file transfers between autonomous computers [2]. Other applications that the networks have supported include distributed computation and the shared use of expensive peripherals and large computers [3, 4]. The bandwidths of the networks have been affirmed to span distances of 0.1 to 10 km, with ranges being 0.1 to 10 Mbps [5, 6].

For local networks, different architectures have been proposed and implemented [7]. One of the earliest of these architectures is the Ethernet [8]. For various data transfer applications, the experimental Ethernet's feasibility has been confirmed at different installations employed by large user communities, especially regarding the Ethernet's ability to perform superiorly under different conditions [9, 10]. Given the superior performance, a 10 Mbps has been proposed as a commercial product [2, 7].

It is also worth noting that in the last few years, local computer networks have been used for real-time applications, including voice applications [6]. Indeed, the role of local computer networks lies in supporting voice communications, hence the supplementation or elimination of internal telephone exchanges that many campuses and offices use. To achieve new levels of functionality, many studies document that the digitized voice can be integrated into data systems. Examples of the data systems include electronic mail systems and text editors [3-6]. In this study, measurement results were presented, with the aim of evaluating how feasible a 3 Mbps experimental Ethernet is, especially in relation to packet-voice applications. The motivation was to discern whether, given certain bounds, the network could aid in carrying voice traffic. Similarly, the motivation of the study was to determine the degree to which the findings could aid in the validation of theoretical models proposed for predicting other Ethernet-like applications' implementation performance.

II. METHODOLOGY

Regarding the setting up and running of the experiment, this study relied on a special control program. The role of the program lay in establishing idle hosts on the network, as well as loading test programs into the respective hosts. In turn, the control program was utilized in setting up parameters responsible for the description of the test programs' generated traffic patterns. The next step was to allow test programs generate and record data based on the experimental run's duration. After the completion of the run, the resultant statistical data accruing from the target hosts was collected. For any loss of packets arising from collisions that arose from noise and were also unlikely to be accounted by the transmitter, they were ignored. Thus, the study assumed that a successful transmission of a packet ensured further that it was received successfully. During throughout computation, an assumption that was held was that apart from the case of the checksum and the 6-byte Ethernet, the whole packet constituted useful data. The implication is that the study's findings were a representation of the performance upper bound because the majority of the real applications included more protocol data on individual packets.

For each run, the experimental duration was set at 60 seconds. From the previous literature, statistics do not differ significantly when run times are set in the range of 10 to 600 seconds [7-9]. It is also notable that up to 32 hosts can be controlled by the control program conveniently [3]. Should there be a need to ensure that more hosts are controlled, the control program needs to be run on at least two machines [4]. Also, each machine needs to control at most 32 hosts [1]. The latter arrangement increases the required effort for designing and running an experiment considerably, as well as statistical data collection and analysis. In this study, most of the experimental scenarios utilized at most 32 hosts.

Of importance to note is that this experimental study was different from most of the previous scholarly investigations in such a way that it stretched beyond the parameter of throughput to measure packet delays. Also, different traffic patterns were used, especially those capable of modeling voice conversations.

III. Results and Discussion

The purpose of this study was to determine the impact of P_{max}, P_{min}, R, and N parameters on communication quality and network performance. In the majority of the experimental scenarios, R was set at 105 Kbps. At P_{max} set at 1024 bytes, the relationship between delay and throughput was established as shown in Figure 1 below.

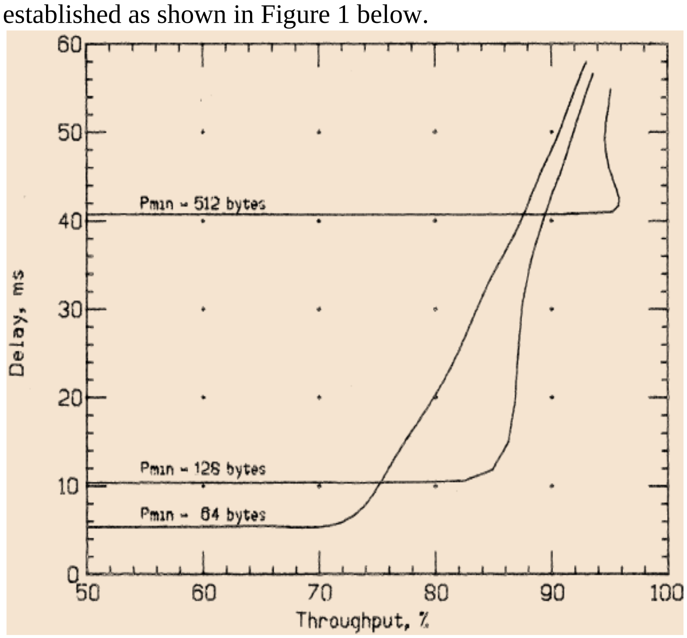

Figure 1 - The Relationship between Delay and Throughput

At P_{min}, other variables that were investigated included the relationship between delay and the offered load. The results that were obtained are summarized in Figure 2 below.

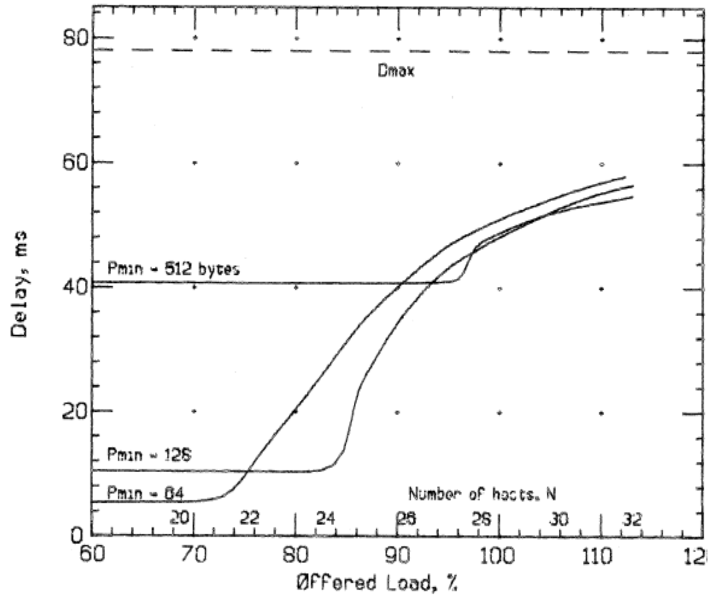

Figure 2 - A Summary of the Study's Finding on the Relationship between Delay and the Offered Load

From the figure above, P_{min} was set at 512. Based on this design, the eventuality was that when additional load was introduced, L stretched beyond 1%.

From the study's specific findings, this study established that whether centralized or distributed, higher-level access control plays an important role in ensuring that network overload is avoided. Notably, network overload degrades voice connection performance in the entirety. With an increase in the minimum packet length, this study established that there is a need for the algorithm to respond to load changes more rapidly. In so doing, acceptable voice quality is likely to be maintained.

Lastly, this study examined the impact of the maximum packet length on network performance. To achieve this objective, the experimental was designed in such a way that the maximum packet length was varied and involved 512, 256, 128, and 64 bytes, with D_{max} for each of these values set at 39, 19.5, 9.8, and 4.9 ms, respectively. From the results, this study established that when the offered loads were of low to medium ranges, there was no loss. However, it was noted that when an integrated data or voice network was used, the data traffic was not connection-based. The bursty nature of the network implied that the control algorithm was unlikely to remove high-load spikes entirely. Therefore, further investigation is needed to understand whether a priority-less Ethernet might solve the problem satisfactorily. From the previous literature, a possible approach that has been proposed to address this challenge concerns the decision to employ only connection-based traffic [3, 4]. In other studies, it has been documented that the problem could be addressed by introducing packet-level priorities [7, 8].

IV. Conclusion and Future Implications

In this study, a simple voice communication protocol was proposed, especially in the context of packet-switching computer networks. From the results that had focused on an experimental Ethernet set at 3 Mbps via the proposed protocol as a framework, findings demonstrate that the network can support voice traffic at a performance level deemed acceptable. A specific example is a case in which about 35 conversations can be supported by the network if the maximum packet length is set at 1024 bytes and the minimum packet length set at 64 bytes. For all voice connections, this study established that there is likely to be a degradation of performance if the Ethernet protocol's fairness experiences high offered loads. The implication is that if acceptable quality is to be achieved, there is a need to implement a high-level access control model, especially due to its ability to avoid the offered load's peaks. In future, it is recommended that studies focus on how the latter interaction might be altered if a computer network setup is set in such a way that packet-level priorities are eliminated.

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