An Experimental Study of Multimedia Traffic Performance in Mesh Networks Review

This paper appeared in the International Workshop on Wireless Traffic Measurements and Modeling Workshop, which was one of two workshops in conjunction with MobiSys 2005. MobiSys is an ongoing annual international conference on mobile systems, applications, and services and at the time was the third installment of the conference.

Mobile systems were relatively new at the time, hence the young age of MobiSys. This explains why something as basic as traffic measurement warranted an entire workshop dedicated to it.

The abstract provides a fairly detailed description of the evaluations performed in the paper—performance of video and voice traffic through multi-hop wireless paths and capacity of the mesh network, impact of different traffic and network characteristics on application performance, the impact of various wireless network interface card configurations, and how to improve performance. The authors claim their work to be beneficial to wireless network capacity planning as well as wireless protocol design.

As wireless technology expands, so does the opportunity for a new variety of applications. Such applications, specifically in the multimedia field, include video streaming, VoIP, and online games. However, these applications come with stricter network requirements that need more flexible and robust network support. In order to better understand these characteristics, evaluation and analysis of application performance over wireless networks is necessary, and can even be used to design robust protocols for the future wireless Internet.

While wireless research existed at the time, it was conducted through simulations which assumed simplified protocols, ideal wireless channels without background noise and interference, and unrealistic traffic traces. This proved quite inaccurate, sparking the need for physical testbeds to conduct more accurate measures.

Setting up such a testbed is not an easy task, and the results depend upon specific configurations and protocol settings that can be hard to fine tune. However, a correctly setup multi-hop mesh network testbed can provide far more accurate results than previous simulations.

The UCSB MeshNet testbed is UCSB's wireless mesh network, deployed on 25 nodes running IEEE 802.11b wireless radios located over five floors of the Engineering building. The goal of the MeshNet is to evaluate protocols and system designs for multi-hop wireless networks. The MeshNet is comprised of two different types of nodes; with the experiments being conducted on the type called Mesh Gateways. The existing tools iwpriv and iptables are used to set and lock the pseudo BSSID, and filter packets and route configurations, respectively. The two newly developed tools—link reliability test tool and time synchronization tool, are network monitoring and diagnosing tools to facilitate repeatable experiments with accurate analysis written by the authors.

The first tool—link reliability test tool, performs tests between nodes to determine link reliability. It has two goals; to measure link quality of each hop, and to identify asymmetric links. To measure reliability, broadcast packets are transmitted periodically in both directions, and the number of successfully received packets in a given period of time determines the level of reliability. A link is deemed symmetric if the calculated reliability rate is above 70%. These tests are performed multiple times, and the node pairs with reliable bi-directional links are used for the experiment.

The latter tool—time synchronization tool, calculates the time synchronization between mesh nodes. This is needed for the delay and bandwidth calculations in the performance analysis. These values are calculating by taking the time difference of two machines by sending time stamped probes over the wired management links of the mesh nodes. These wired links are connected to the Engineering building's LAN. The authors' measures show that these calculations have less than ten microseconds of error.

The testbed's network topology consists of five nodes (chosen from the reliable nodes identified by the previously mentioned tool) forming a four-hop path. The nodes are scattered over the second and third floors of the building. The routing tables are updated with static route entries to force four hop paths.

The performance of UDP video and voice streams over the mesh network are examined. In the MAC layer, all nodes operate in ad hoc mode on channel 6, using the static routing topology previously mentioned. The configuration parameter that varies during experiments is setting the card to a fixed data rate vs. auto rate.

The experiments are performed at night to reduce the amount of background noise and random interference. Results are also collected during the day to show the effect of these factors. These experiments are as follows: measuring the impact of auto-rate adaptation of the wireless card vs. fixed rate, measuring the impact that enabling or disabling RTS/CTS has on performance, and finally measuring the impact that varying maximum retransmission value has on the number of transmissions.

The results show the capacity of the mesh network for both video and voice streams. As shown in the table, as the number of hops increases the capacity shrinks. Unexpectedly, the network has a higher capacity for voice than for video streams. This is due to the higher packet sending rate of voice streaming.

The result of fixed vs. auto rate is a marginal effect on capacity, only significant in the single hop scenario of video streaming. This is because the network capacity is constrained by the number of hops.

In addition, when using auto rate the fairness between competing video and voice traffic is significantly unfair. Finally, their results indicate that with different channel conditions, jitter could have a noticeably negative effect on the received data quality.

Simply put, the result of testing RTS/CTS for multimedia traffic is that it is a bad idea to enable it. The graphs clearly show that RTS/CTS enabled flows actually increases latency and loss over those without it enabled, and even limit the capacity of the network.

The result of changing the maximum number of MAC layer retransmission, is that with small values the transmission latency over each hop reduces which in turn increases the capacity of the network as well as improves the packet delivery rate. With retransmission enabled, the loss rate is significantly lower, however an ideal value does not exist as it highly depends on the network characteristics such as congestion and number of hops. Further research on the relationship between maximum retransmission and the number of hops would help find an optimal retransmission value to increase performance.

The summarizing points are: the capacity of the network is constrained by the number of hops. The number of supported flows is heavily influenced by packet sending rate not data rate or packet size. Auto-rate does not always improve capacity. Unfairness among flows is a result of channel capture. Packet jitter can be significant in 802.11b networks, requiring a solution to dampen the variation. Enabling RTS/CTS does not improve performance of real-time traffic. And finally, finding an optimal value for the maximum retransmission number could improve performance.

The abstract of this paper was a bit too detailed, as the exact information was repeated almost identically in the introduction. However, it accurately described the papers goals and results.

The introduction claims previous simulations are inaccurate because they don't take into account background noise and random interference. Yet the experiments take place only at night when background noise and interference are at an unrealistic minimum. Though some experiments ran day and night, their results did not receive as much attention as the other tests, and the differences were only discussed briefly. In addition, the paper states “asymmetric links frequently occur in wireless networks”, yet they chose the five nodes in the testbed based on them having the highest level of bi-directional reliability. Both of these factors contradict their reasoning for using a testbed in the first place, as apposed to the traditional simulation. On top of these contradictions, the scale of the testbed is far too small. Compared to the entire internet, five computers located in the same building streaming to one another at night time is microscopic.

A diagram of the network topology would have been aesthetically pleasing.

It says there are two different types of nodes in the UCSB MeshNet, but only names one.

The references are all incomplete, labeled as “[?]”.

The graphs are hard to read due to their small size, and distinguishing between small markings on the lines is visually difficult. In addition, there is a wealth of measurements and values both in the graphs and text, which are quite difficult to keep up with.

Compared to testbeds today, this is very imprecise. But at the time of this paper, these experiments and results were a significant improvement over previous simulations, and paved the way for larger and more accurate future testbeds.

Since this paper was presented at a workshop, it makes sense that it wasn't fully completed and things like incomplete references and small graphs are forgivable. Though the underlying contradictions between the reasoning and implementation are not. Overall it was a significant step forward for wireless network analysis at the time, but has plenty of room for improvement.