# An Experimental Study of Multimedia Traffic Performance in Mesh Networks

This paper was published in the MobiSys International Workshop on Wireless Traffic Measurements and Modeling in 2005. The objective is to present an experimental study of multimedia traffic performance in mesh networks. Most of the existing wireless researches have been conducted using simulations which left out the consideration of random interference and background noise while utilized unrealistic traffic traces. An experimental study conducted in real testbed would draw more accurate conclusions. As a result, such study would be beneficial in both wireless network capacity planning and protocol design.

The introduction section of this paper provided solid information in regard to the motivation behind such experimental study. The problem of multimedia on wireless network, which could be summarized as applications that demand seamless real-time delivery placed on top of a highly variable network channel, was properly introduced. The authors mentioned specifically the importance of enabling repeatable tests and results, however, it was not clear whether the test results presented in this paper were repeatable but the authors only stated that measurements were collected at night in a relatively stable environment with less interference. In fact, when discussing the intra-flow variation in section 4.3, figure 4 shows a sudden burst of latency in the eight concurrent flow network and no explanation was given as to whether such sudden burst is environmentally caused or can be repeated.

The paper reveals a novel solution for measurement to the highly variable wireless network. Rather than using simulators like the majority of the research, the UCSB MeshNet testbed which consists of 25 nodes was used for the experiment. While the authors disclosed the measurement environment as a five floor Engineering I building, no further details were given in regard to the specification of the environment. Disclosing the environment and specifically the distance between each node in approximate distance will provide a better picture and clarification than using phrases such as “across the hallway” or “neighboring labs. This is especially true with 802.11b as it provides only an approximate range of 120 feet indoor, so the variable distance could lead to additional latency or an increase in loss rate.

The two tools developed specifically for this study showed the careful consideration from the authors in regard to the highly variable wireless network. The authors were able to consider the asymmetric links nature in wireless network, therefore realizing the need for the time synchronization tool to measure one-way delay as round trip latency does not provide accurate measurement like it does for wired connection. While the authors did not specify many details about the mesh network architecture, the utilization of wired management links for measuring one0way delay was pretty neat.

From the discussion of the time synchronization tool, it appears that the multimedia traffic utilized UDP as the transport layer protocol rather than TCP. Such choice is rather interesting considering TCP’s higher deployment rate in comparison to UDP even in video/audio streaming applications. While one possibility of choosing UDP over TCP could be to prevent the congestion control mechanism in TCP to interfere with the wireless network trace collection, the use of UDP as the transport layer protocol is certainly not representative to the general wireless applications. In addition, weaknesses of UDP in content distribution can been found throughout most of the Multicast articles read previously including the MPEG paper by D. Le Gall and the application layer multicast protocols paper by Banerjee and Bhattacharjee.

One of the most interesting results from the experiment is the large influence that packet sending rate has on network capacity in comparison to the small effect from packet size. The surprising result contradicted to regular intuition. The insignificant effect of packet size on network capacity could means a potential performance improvement by combining smaller packets which would result in less overhead from smaller header size, and also better bandwidth utilization. However, such benefits would occur only if it was able to overcome the challenges of reliability and retransmission resulted from larger packet size.

All the graphs presented in this paper supported the intuitive assumption where the latency and loss rate increases with the increase of hops and the number of video streams. It was interesting to see the result of auto-rate adaptation from the experiments. While the auto-rate mechanism follows a slow-start-like process, it was able to achieve an optimal throughput of about 6Mbps after succeeds in adaptation. However, if 6Mbps is the near optimal throughput as claimed by the authors, I wonder why the fixed data rate was chosen at 2Mbps rather than a more optimal scenario. One possibility could be that 2Mbps represented a more average scenario but it is not clarified in the paper.

Overall, this paper is similar to the “Aspects of Networking in Multiplayer Computer Games” paper by Kaukoranta and Hakonen in terms of their scope. Both papers were intended to provide a solution that is beyond the specific problem discussed in the papers. In fact, both papers have a broader scope in which they intended to provide experiences that would be beneficial in future design and studies. The objective of this paper was successfully achieved through the measurement of wireless network performance from actual testbed with the consideration of background noise and interference successfully. The paper overall, was able to carefully consider many aspects of the highly variable wireless network. Specifically, the consideration of asymmetric links resulting inaccurate measurement of delays gave the paper great credibility in my opinion.