Assessment of Wireless Link Transport Layer Protocols Performance using Throughput, Packet Loss and Round Trip Time Metrics

by

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ABSTRACT
This research is a measurement based study of throughput, packet loss, round trip time, to analyze the performance of transmission control protocol and user datagram protocol in providing QoS to WLAN users. The results of experiments to assess the performance of the transport protocols in providing guaranteed QoS to WLAN users showed that packet loss increased as the transmission rate increased. Also the packet loss increased as the distance between the sending and the receiving devices increased. Another observation is that the packet throughput drops with increase in distance between the transmitter and receiver. Finally, the results obtained establish that increase in time for a packet to travel the distance between transmitter and receiver occurs when throughput drops.

Keywords: Assessment, Wireless Link, Transport Layer Protocols, Performance & Throughput

1. INTRODUCTION

Wireless Local Area Network (WLAN) has become a most promising and successful technology in recent years. Wireless transmission of voice and data over radio waves allows its users to communicate with each other without requiring a physical connection to the network. Wireless devices include anything that uses a wireless network to either send or receive data. The first digital network based on packet radio, ALOHANET, was developed at the University of Hawaii in 1971. Wireless networks are superior to wired networks with regard to ease and flexibility of installation (Akella et al., 2005). They do, however, suffer from lower bandwidth, higher delays, higher bit-error rates, and higher costs than wired networks. With the release of the free and unrestricted ISM bands (2.4 GHz and 5.8 Ghz) for Wireless Local Area Networks, bandwidth has increased and prices have decreased on Wireless networking solutions. These factors have made WLANs a very popular Wireless networking solution. The demands of end users and their applications are daily increasing as a wider variety of new applications are being invented. These applications impose different demands on the underlying network protocol suite. High bandwidth Internet connectivity has become a basic requirement to the success of almost all of these areas.
2. WIRELESS LOCAL AREA NETWORK ARCHITECTURE

The basic building block of a WLAN network is the Basic Service Set (BSS), which is a group of stations that communicate with each other. Communication takes place within the ‘basic service area’, and a station within the basic service area can communicate with other members of the BSS. Generally, BSSs may be independent networks, infrastructure networks or extended service networks. Figure 1 is an Independent BSS (IBSS), which is also called an ad-hoc network. Stations in an IBSS can communicate directly with each other within a. Typically, IBSSs are composed of a small number of stations set up for a specific purpose and for a short period of time.

In order to achieve wider coverage in wireless networks, an infrastructure BSS (Figure 2) is more common in which case APs are used for all communications, including communication between mobile nodes in the same basic service area. If station A needs to communicate with station B, the communication must take two hops. Although multi-hop transmission takes more transmission power and time than a direct packet transmission from the sender to the receiver, it has two major advantages. First, an infrastructure BSS is defined by the distance from the AP. All mobile stations are required to be within reach of the AP, but no restriction is placed on the distance between mobile stations themselves. Thus a large communication range is available for infrastructure BSS. Second, allowing direct communication between mobile stations would save transmission power, but requires them to maintain neighbor relationships with all other mobile stations within the service area. APs in infrastructure networks are in a position to assist with stations attempting to save power. APs can note when a station enters a power-saving mode and buffer packets for it. Battery-operated stations can turn the wireless transceiver off and power it up only to transmit and retrieve buffered packets from the AP, which can give the battery-operated stations a longer service time.

BSSs can create coverage in small offices and homes, but they cannot provide network coverage to larger areas. However, it is possible to link BSSs into an Extended Service Set (ESS). An ESS is created by chaining BSSs together with a backbone network. Figure 3 ESS is the union of the BSSs 1, 2 and 3, in which each BSS AP connects to each station wirelessly. AP 1, AP 2 and AP 3 are connected by the backbone network, which can be either a wireless or a wired network. As the user moves, the signal strength of the signal from AP-1 will decrease. At some point, the signal strength from AP-2 will exceed AP-1, and the wireless bridge will establish a new connection with AP-2. This is called a hand-off. This is an automatic process for the wireless client adapter in 802.11, and the term used to describe this is roaming (Stajmenovic, 2002).

Figure 1: Independent Basic Services Set (IBSS) or Adhoc Networks
2.1 Transport layer Protocols
Transmission control Protocol (TCP) is a connection oriented reliable protocol. Reliability is ensured by acknowledgements from the receiver and retransmissions of the lost packets at the sender. Congestion control and flow control mechanisms in TCP make sure that the fast sender does not swamp too many packets on to the slow receiver. Whenever a packet is lost, TCP reduces its congestion window size (a sender can only send a maximum bytes indicated by the window size) to reduce congestion (Gopalakrishnan, 2004). TCP is traditionally designed for wired networks where in the packet losses are mostly due to congestion. But in wireless networks (in 802.11 (a, b and g) networks) the packet loss due to collisions and corruption (because of poor signal strength) is very high. In such cases, TCP reduces its congestion window when a packet loss is reported and eventually the throughput becomes zero. The 802.11 MAC employs MAC level retransmissions to hide these errors from TCP layer (Gu and Zhang, 2003).

User Datagram Protocol (UDP), on the other hand, is a connection less unreliable protocol because there will be no acknowledgement from the receiver of a correct packet reception and hence no retransmissions. Please note that these retransmissions (transport layer) are different from MAC layer retransmissions.
3. QoS RELATED STANDARDS AND TECHNOLOGIES

“QoS is defined as the ability of the network to provide a service at an assured service level” (Lindgren et al., 2003). Traffic of different classes or traffic with different requirements receives different levels of QoS assurance. QoS support mechanism refers to any mechanism that is equipped by any kind of QoS support while QoS guarantee refers to a mechanism that can provide guaranteed support. The objectives of QoS provision can be categorized into prioritized QoS support and parameterized QoS support (Gu and Zhang, 2003). With the increasing demand and penetration of wireless services, users of wireless network now expect quality of service and performance comparable to what is available from fixed networks. Some factors that influence QoS of Wireless Network include network throughput, retransmission attempts, dropped data and medium access delay.

3.1 QoS PERFORMANCE METRICS

For any QoS analysis to be performed on wireless links, a measurement of each QoS performance metric must be carried out. The analysis of the measured values obtained will then provide an insight into the state of health of the wireless link. Performance of wireless LAN may be evaluated with respect to signal to noise ratio (SNR), file size, number of simultaneous users and the direction of file transfer in (Demir et al., 2000). The practical performance of IEEE 802.11 depends on the combined MAC and PHY environment at both sender and receiver on each 802.11 link. Thus, their impact cannot be estimated purely on the basis of PHY measurements such as signal-to-noise ratio (SNR) at the receiver. Alternative high-level characterizations, such as throughput and delay statistics, can also detect lost transmission opportunity. In fact, the nature and number of transmission opportunities are a combination of MAC- and PHY-layer effects (Giustiniano et al, 2010). It has been observed that the throughput per user is quite low when there is more than one user in a LAN, but the overall or net throughput increases giving an impression that the lower layers of the 802.11b protocol stack reserves some resources for future users. It is also observed that fairness is preserved by the underlying mechanism of 802.11b protocol. The results show that the number of simultaneous users has a high impact on the throughput compared to that due to SNR while file size has little or no impact.

The effects of path loss and building loss measurements due to residences for outdoor networks were described in (Durgin et al., 1998). Effects of various shadowing objects like trees and houses and receiver positions measured and quantified. Zahur et.al., in (Zahur et al., 2003) measured the performance of 802.11 networks. The effect of various parameters including distance, power, and RTS/CTS on throughput was measured with varied packet sizes using simulation. Hidden node problem was also considered to analyze the effect of RTS/CTS. In (Kamerman and Aben, 2000) net throughput is assessed by modeling the physical layer and medium access.

Throughput
This concerns the maximum data rate of a communications link or network access medium. A typical method of performing throughput measurement is to transfer a ‘large’ file and measure the time taken to do so. The throughput is then calculated by dividing the file size by the time in order to determine the throughput in megabits, kilobits or bits per second. Unfortunately, such an exercise will only result in the ‘good-put’ which is less than the maximum throughput, leading to the erroneous belief that their wireless link is not operating correctly. In fact, there are many overheads in transmission that mean the calculated good-put does not reflect the maximum throughput.

Latency
Latency is to the delay between sending and receiving a message over a packet-switched network. Generally, the overall application latency is comprised of the time to process messages at both the sending and receiving hosts (host latency) and the delays which occur inside the network (network latency). The overall performance of distributed applications depends on the hosts and the network connectivity between them. Network latency is the delay that is introduced by the network; it excludes the hosts’ software processing at the source and destination end-points. Network latency can be
measured either one-way (the time from the source sending a packet to the destination receiving it), or round-trip (which is the sum of the one-way latency from source to destination plus the one-way latency from the destination back to the source). In general, it is easier to measure round-trip latency than one-way latency, because both the send and receive events occur at the same physical location. Users of wireless LAN services begin noticing network latency as an issue when round-trip latency is greater than 250 milliseconds (ms). The International Telecommunications Union recommends that latency should not exceed 150 ms one way (i.e. from transmitting node to receiving node).

Packet Loss
Packet loss occurs when packets sent are not properly received by the other end, causing them to be discarded by the receiver. Packet loss can be caused by many different reasons: overloaded links, overload in the receiving device, excessive collisions in the wireless link, physical media errors due to interference or low link quality, and others. However, packet loss starts to be a real problem when the percentage of the lost packets exceeds a certain threshold (roughly 5% of the packets), or when packet losses are grouped together in large packet bursts. There are two kinds of packet loss correlation, namely temporal correlation and spatial correlation. The temporal loss correlation measures the packet loss pattern over time. It is often related to the issue of consecutive packet losses or the burstiness of packet loss by a single receiver. However, for multicast/broadcast communication, in which a single packet has multiple receivers, it is also useful to measure the correlation of losses by different receivers, which is referred to as spatial loss correlation.

3.1 Single-point end-to-end QoS Performance Measurement
The simplest form of end-to-end performance measurements can be executed as single-point measurements directly from the terminal, which uses some service. The measurement software can be an application on top of the protocol hierarchy, in which case the measurement gives directly an idea about the application layer performance that is usually the desired case in QoS. As it can be assumed, with this kind of measurement setup, it is only possible to obtain information about the total round trip performance of the system. Often it is assumed that the links in both directions have about the same performance and, therefore, for example, one way delay is got simply by dividing the total delay by two. However, this can lead to serious errors since, as it is known, many communication systems are asymmetric in nature.

Figure 4: Single-Point end-to-end Performance Measurement
3.2 Multi-point end-to-end QoS Measurements
Bidirectional end-to-end measurements give good insight into the overall performance in both directions. However, if one is interested in the reasons behind the behaviour of the performance, more complicated measurement setup is needed. In the presented directional measurements, information in packet headers (addresses, ports, packet identification numbers, timestamp etc) is used to identify individual packets transferred between stations, at the multiple measurement points in order to be able to calculate QoS metrics (delays, losses, etc.). In addition, the actual execution of the measurement can be a problem, because of confidentiality concerns.

![Figure 5: Multi-Point QoS Performance Measurement](image)

3.3 Link-to-link QoS Measurement
Ultimately multi-point measurements lead to link-by-link measurements or hop-by-hop measurements, where the performance of every link or every hop in the communication chain is determined, respectively. This is very difficult and time-consuming, but is needed, if for example some broken or single bottleneck network components are needed to be found. In terms of QoS-handling, a more common multi-point measurement, like network segment measurement, is usually enough. Another problem is that the performance gets worse with increasing hop count, and also, the number of required probes increases quickly.

![Figure 6: Link-to-link Performance Measurement](image)
3.4 Measurement Analysis
In accurate QoS measurements, the effect of the network traffic itself must be included. High traffic load of the measured network creates extra queuing delays, which might give false information about the real network performance. Therefore, one should be aware about the current load of the network when performing measurements. Otherwise no accurate decisions can be drawn from the basis of the measurements. On the other hand, the end user is only interested in end-to-end quality of service, regardless of the underlying technologies, i.e., the user is only interested that does the service work or not, and does not care what are the reasons behind bad service level. In this respect, simple end-to-end performance measurements in a public network with all the other network traffic included are well-grounded and valuable. Measurement of QoS can either be sampled or full. In sampled measurement, data is collected intermittently and collated at the end of the experiment while in full measurement data is collected only at the end of the experiment. In this research we assess the quality of service of wireless local area network infrastructure with respect to key performance indicators such as throughput, packet loss and round trip time.

4. MATERIALS AND METHODS
This research has been carried out using data gathered by a survey method in the test environment. A common method of testing network performance involves initiating a file transfer from one end to another end was used for the tests while walking. In this method the researcher chooses a route that has clear line of sight between the access point and the Laptop. The walking speed does not really matter but it is advisable for the test Laptop and software to be allowed time to settle, and hence get a good signal and reading.

The hardware and software used for the data collection are as follows:
1. Two Laptops (with windows 7 Operating System)- or other similar device (wireless enabled)
2. Linksys WRT54G V2 access point
3. Tamosoft Throughput Software (server and client),
4. 50m length surveyor tape

The schematic of the measurement arrangement is shown in Figure 7.
4.1 Research Architecture

The schematic diagram in Figure 7 shows the setup for the measuring campaign and consists of the:
1. Laptops (2 nos.) – equipped with Tamosoft Throughput test software which provides a test platform for the experiments performed. Tamosoft is a windows OS based system.
2. Linksys WRT54G V2 access point – This serves as the hub for connecting the Laptops via the wireless radio link so that the Probe can record test data.
3. Measurement tape

4.2 Measurement Procedures

We set up a temporary WLAN access point and performed investigation of the covered area. The area of study is the Life Science Shopping Complex at Ugbowo Campus of the University of Benin. The server is a Sony Viao Laptop, with 802.11 b/g Realtek Wireless LAN Card and IP address 192.168.1.100. The client is a HP Pavilion Laptop, with 802.11 a/g/n Wifi link 5100 Wireless LAN Card and IP address 192.168.1.102. The access point is a Linksys router with IP address 192.168.1.1. We installed Tamosoft Throughput Test tool on the laptops and performed test for throughput, packet loss and round trip time in the coverage area of the access point. The wireless receiver systems on the Laptops are designed specifically for sweeping, analyzing and optimizing Wireless Local Area Networks. The tests were performed at fixed distance separations apart and the QoS parameters recorded. Data processing and analysis was done using Microsoft Excel software.

Figure 8: Walk Test Environment (google earth)
5. RESULTS AND DISCUSSION

Readings were collected on several occasions for two weeks period around the study area within a radius of 45m. The readings obtained for the experiments performed in the walk test environment were then averaged and is presented in Table 1.

Table 1: Averaged readings obtained during the experiments

<table>
<thead>
<tr>
<th>distance (m)</th>
<th>Throughput Average (MBps)</th>
<th>Packet loss Average</th>
<th>Round Trip Time Average (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>tcp up</td>
<td>tcp down</td>
<td>udp up</td>
</tr>
<tr>
<td>0</td>
<td>40.95</td>
<td>39.19</td>
<td>51.45</td>
</tr>
<tr>
<td>5</td>
<td>14.52</td>
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<td>25</td>
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<td>5.07</td>
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<td>30</td>
<td>16.12</td>
<td>24.56</td>
<td>18.95</td>
</tr>
<tr>
<td>35</td>
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</tr>
<tr>
<td>40</td>
<td>17.5</td>
<td>17.71</td>
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<tr>
<td>45</td>
<td>19.38</td>
<td>19.11</td>
<td>24.58</td>
</tr>
</tbody>
</table>

Results show that as the distance separating client and server computer/ AP increase, both the upstream and downstream TCP and UDP throughput decrease. This is in agreement with literature studied. This trend will continue to the extent that link speed and QoS will degrade. A possible solution will be to deploy multiple APs in the coverage area. This will increase the capacity of the network and hence improve QoS.

Figure 9: Average upstream throughput vs Distance between Server and Client computer
Results obtained show that Packet loss is higher at higher transmission rate for UDP traffic. TCP has retransmission mechanism which is used to check if the sent packets were received in good state, hence it does not suffer packet loss as throughput increase, both the downstream and upstream UDP throughput as shown in Figure 11 and Figure 12. This is also in agreement with literature studied. This trend will continue to the extent that link speed and QoS will degrade. A possible solution will be to increase the transmit power, but studies have shown that this will increase packet loss ratio eventually and hence impact negatively on the QoS.
Experimental results indicate reduction in throughput produces increase in round trip time. This is clearly seen in Figures 13 and Figure 14. The relevance of this result is that if excessive round trip time is observed in accessing a network service, then it is a certain proof that data throughput is suspect. This can be invaluable information for the network user or administrator to localize the cause of poor QoS in the network.
The experimental result showed distinct increase in packet loss rate with increase in distance separation between access point and client during packet transmission.

6. CONCLUSIONS

This research concludes by recommending that measurement based analysis of WLAN link be carried out to establish the unique link characteristics before rolling out wireless based services. Flaws can be detected on time and adjustments made in the installation. The quality of service of multimedia transmission over live WiFi networks depend on the transport protocols like TCP and UDP. In this research work we have observed among other observations that UDP packet throughput and packet loss is higher when compared to TCP packet throughput and packet loss. TCP protocol bandwidth is also very high when compared to UDP bandwidth for the reason that packet drop is negligible (Konyeha and Konyeha, 2015).

REFERENCES