ENSC 427 COMMUNICATION NETWORKS
Spring 2015 Final Project
VoIP Analysis over Wi-Fi using RIVERBED Simulation

Group # 10
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Abstract

VoIP is an essential for the delivery of voice communication and sees wide applications in office and home use. VoIP over Wifi although not as high in quality as VoIP over Ethernet is connectionless and is popularized by mobile phone applications such as NetTalk and Viber. We will be examining how delay, jitter, packet loss contribute to congestion and affect the voice quality and delve deeper into what voice codecs are popular in today's world of telecommunications.

1. Introduction

WiFi is an important and popular wireless technology that supports electronic devices in computer networking. WiFi is applied not only for personal purposes but also in office. VoIP, which stands for Voice over IP, is an essential and popular application to voice communications and multimedia sessions over Internet Protocol networks. For example, both Viber and Skype applications are widely used today. In this project, we are going to simulate VoIP over WiFi and discuss the several important effects of packet loss, delay, jitter on the quality of VoIP over WiFi. Furthermore, we will also be discussing the two popular voice encoding schemes called G.711 and G.729a to compare their MOS values and their performances.

Technology Background

WiFi: Is a local area wireless technology that networks electronics devices using the 2.4 GHz and 5.0 GHz radio bands based on the IEEE 802.11 standards. Our simulations will be based on the IEEE 802.11g standard which will only operate in the 2.4 GHz band with a maximum physical layer bit rate of 54 Mbps.

Riverbed Modeler: A network simulation software tool produced by Riverbed Technology Inc. that allows users to create and analyze network topologies from a selection of protocols provided by the software. For more flexible custom designs, the user may make their own node and process models on Riverbed using C language to define their simulation objects. Our simulations will be done on Riverbed Edition 18.0.

2. Terminology

MOS: Stands for Mean Opinion Score and it is used to test and obtain the users' view of the quality of the network. MOS tests for voice are specified by ITU-T recommendation. The standard of the MOS: 5 is excellent, 4 is good, 3 is fair, 2 is poor and 1 is bad.

Jitter: Variation of packet inter-arrival time which can cause click sounds in voice streams. The OPNET defined jitter is the difference of the source jitter minus the destination jitter.

Packet loss: means the users can't receive the full data and lose some packets when the router is sending the data to the users. Packet loss is caused by the network congestion.

Throughput: The total data traffic successfully received and forwarded by the MAC layer to the high layers.

ITU-T: Stands for Standardization Sector of the International Telecommunication Union in charge of producing standards that cover all fields of telecommunications.

PSQM: Stands for Perceptual Speech Quality Measure and is a computational algorithm standardized by the ITU-T under the recommendation P.861 for evaluating voice quality of 300-3400Hz voice-band speech codes.

G.711: PCM waveform codec about standardized by the ITU-T for audio companding using a 64Kbits/s bitrate for its sampling frequency (the standard Nyquist sampling rate for a 4KHz voice channel).

G.729A: Simplified version of G.729 requiring less computational power. It is an audio compression algorithm that compresses audio into 10 ms duration packets. The operational bitrate is only 8kbits/s and is commonly used where bandwidth must be conserved.

3. Design Implementations

3.1 Wifi Setup

In wifi design, we created some scenarios on Riverbed modeler to test how jitter,delay,and packet loss affects the MOS value for voice quality. In each scenario, a server transmits voice traffic to a WLAN router which will communicate with a WLAN client. In each scenario, we placed one application, one profile, one server, one router and one wlan_wkstn on Riverbed as illustrated in the figure below.

NOTE: All scenarios and tests in this report are based on this topology

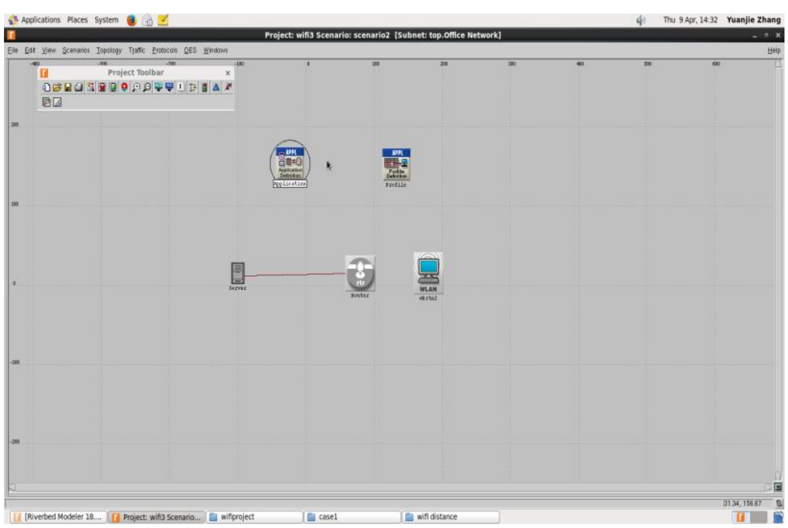
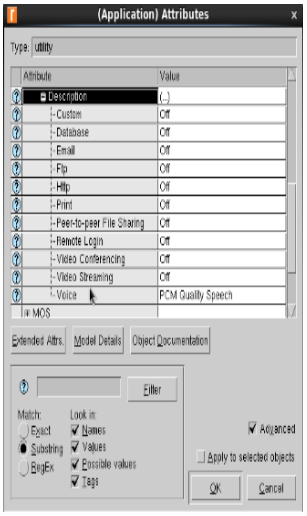
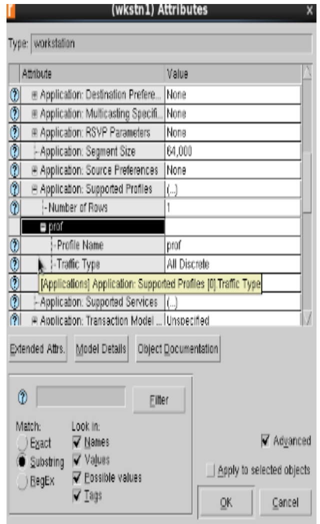


Figure 1

In each scenario, we define the application and profile for PCM Voip which will be applied to the server and wlan_wkstn(client). Their definitions are given below.





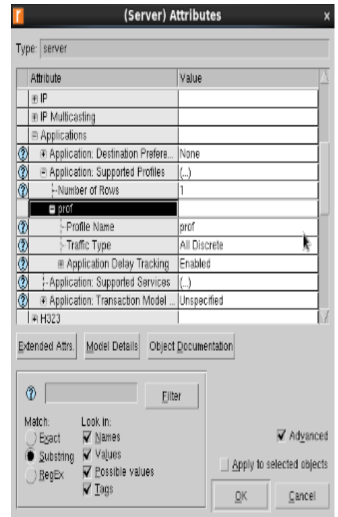


Figure 2 Figure 3 Figure 4

3.1.1 Jitter,Delay,Traffic received, MOS value, and Queue Size over different distances

First of all, we want to analyze the results when the wlan_wkstns are at the different distances. We create five scenarios and choose five distances between the router and the users. The below distances were chosen because 250 meters is the farthest distance in which the our defined voice application receives all of the packets it forwarded to the transport layer. As a result, we should test the distances farther away than 250m and observe the results of our parameters.

Wkstn1	Wkstn2	Wkstn3	Wkstn4	Wkstn5
250 meters	260 meters	265 meters	270 meters	280 meters

3.1.2 Effect of Data Rate

The data rate also will affect the results such as the delay , jitter , MOS, packets received and packets sent. We will simulate the scenarios by changing the client user workstation data rate 18Mbps, 24 Mbps and 54 Mbps and keep the router data rate and distance fixed at 24Mbps and 265 meters.

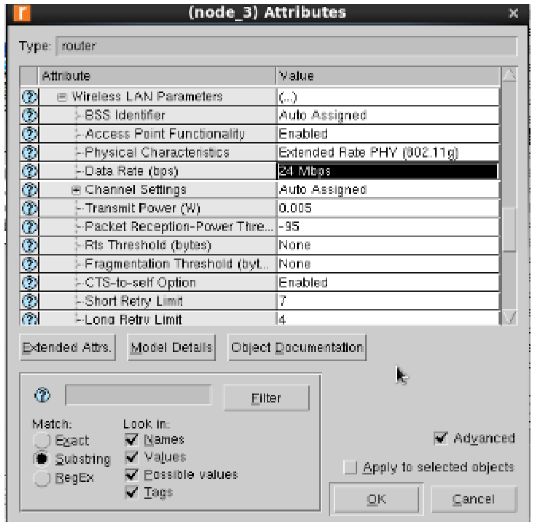


Figure 5

3.1.3 Effect of Buffer Size

We want to make sure how buffer can affect the wifi results as the delay, jitter, MOS, packets received and packets sent. Thus we choose the users' buffer as 256000 and 1024000.

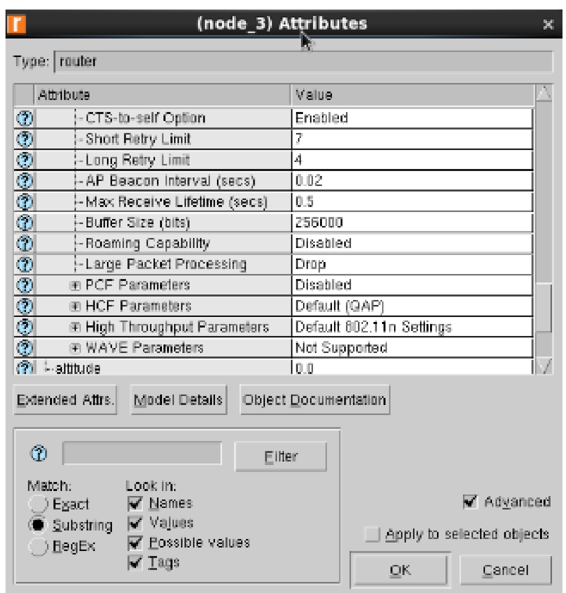


Figure 6

3.1.4 Voice Codec Comparison

We want to compare the performances of both G.711 and G.729a voice encoding schemes to see how a audio data digital compression scheme(G.729a) stacks up against a PCM audio analog-to-digital scheme(G.711). To implement G.711 on our topology on Riverbed, we select from the application attribute the voice encoder scheme which is G.711. The parameters for G.711 were chosen to be consistent with the ITU-T standard except for the fact that PLC(Packet Loss concealment) which is important for dealing with lost or discarded packets to minimize jitter is not an available option.

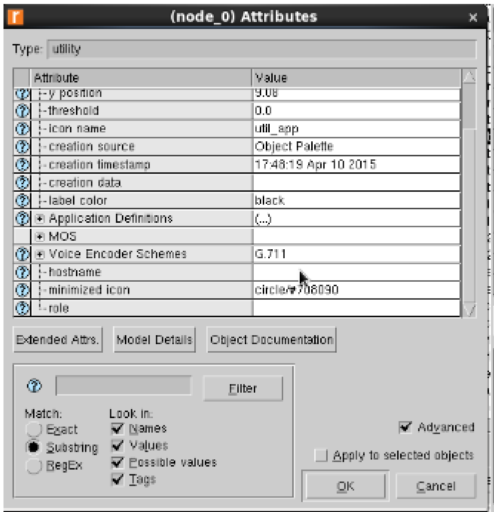


Figure 7

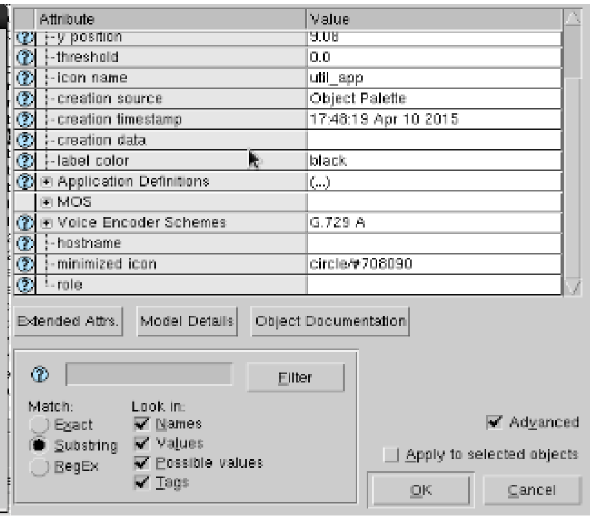
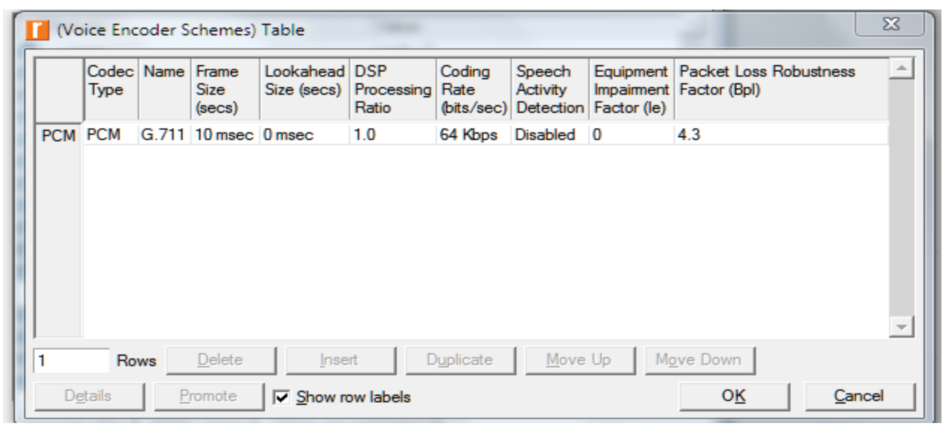


Figure 8


(Voice Encoder Schemes) Table

	Codec Type	Name	Frame Size (secs)	Lookahead Size (secs)	DSP Processing Ratio	Coding Rate (bits/sec)	Speech Activity Detection	Equipment Impairment Factor (Ie)	Packet Loss Robustness Factor (Bpl)
PCM	PCM	G.711	10 msec	0 msec	1.0	64 Kbps	Disabled	0	4.3

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Details Promote Show row labels OK Cancel

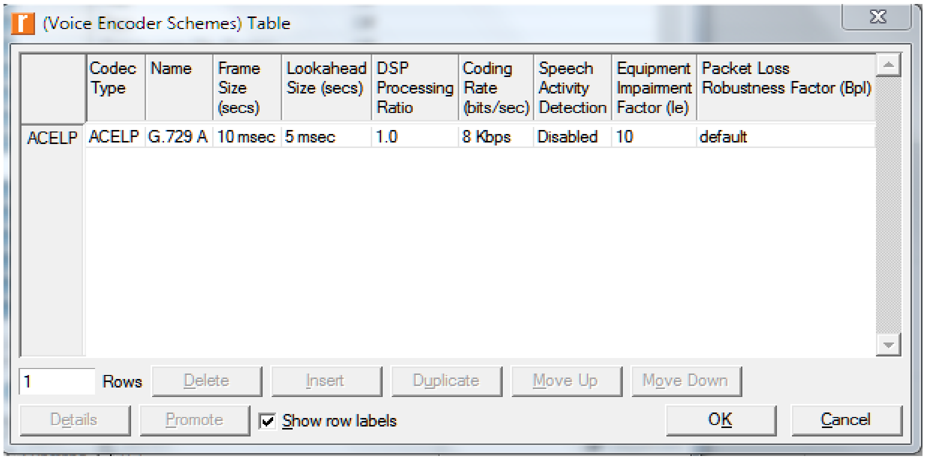
Figure 9 G.711 parameters

The parameters for G.729A were also chosen to be consistent with the ITU-T standard except there is no option to choose between the u-law version or the a-law version of the algorithm.

We chose the parameters to be consistent with the summarized ITU-T standards found on wikipedia about G.729a[1]. The frames are 10 ms because the compression algorithm of G.729 which produced 10 ms duration packets from digital voice.

The features of G.729a are:

- Sampling frequency 8 kHz/16-bit (80 samples for 10 ms frames)
- Fixed bit rate (8 kbit/s 10 ms frames)
- Fixed frame size (10 bytes for 10 ms frame)
- Algorithmic delay is 15 ms per frame, with 5 ms look-ahead delay
- G.729a is a hybrid speech coder which uses Algebraic Code Excited Linear Prediction (ACELP)



	Codec Type	Name	Frame Size (secs)	Lookahead Size (secs)	DSP Processing Ratio	Coding Rate (bits/sec)	Speech Activity Detection	Equipment Impairment Factor (Ie)	Packet Loss Robustness Factor (Bpl)
ACELP	ACELP	G.729 A	10 msec	5 msec	1.0	8 Kbps	Disabled	10	default

Figure 10

[1] Wikipedia page on G.729 http://en.wikipedia.org/wiki/G.729

4. Simulation, Results and Discussion

4.1 wifi distance analysis

All scenarios were tested with 5 distances between users and routers

Wkstn1	Wkstn2	Wkstn3	Wkstn4	Wkstn5
250 meters	260 meters	265 meters	270 meters	280 meters

Keep in mind that the maximum propagation delay allowed by the 802.11 WLAN is 1 µsec for nodes within the same BSS. This stipulation was found when browsing the DES log on Riverbed for simulation problems shown in figure 11.

This means that the maximum allowable distance is = maximum allowable propagation delay x speed of light = 1 µs x (3 × 10^8 m/s) = **300m**. Our test distances are below 300m away from the router.

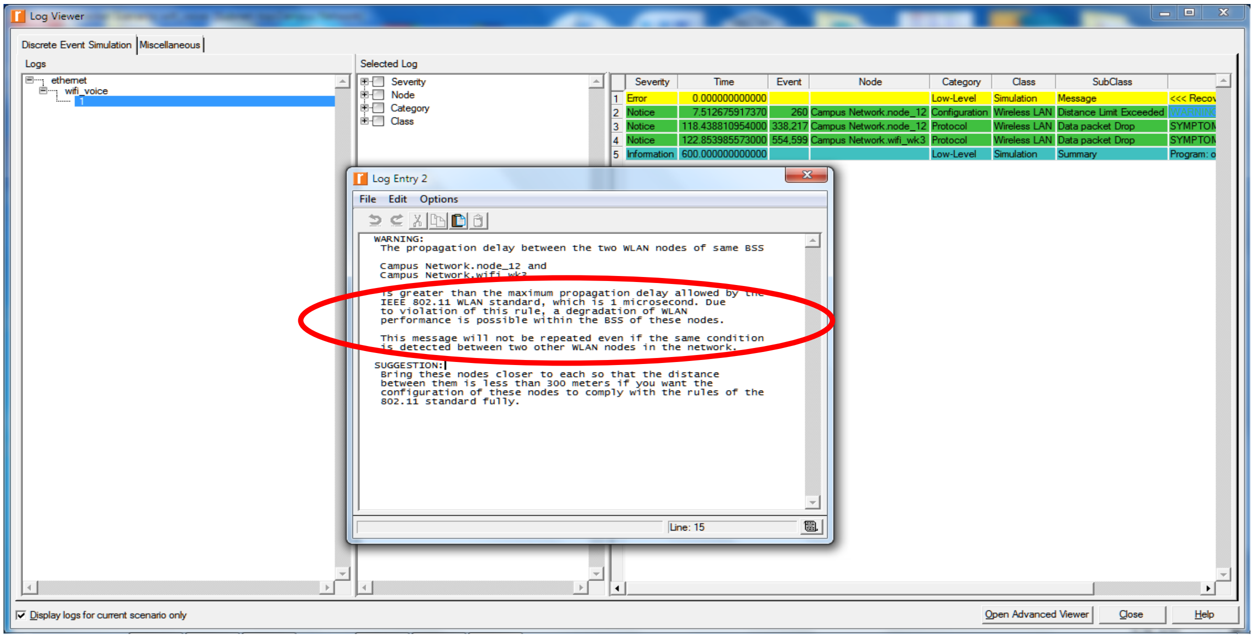


Figure 11

The delay values for the respective distances are as follows for distances 250m,260m, and 265m:

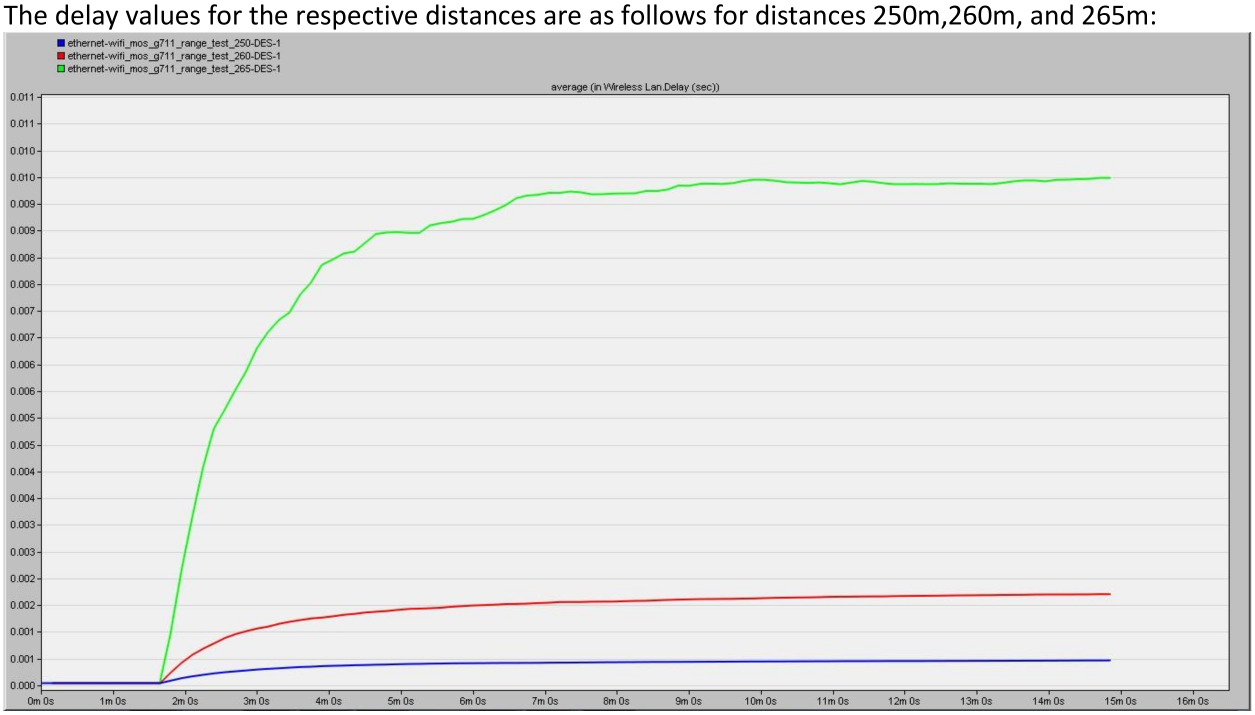


Figure 12

Also the delay values for 270m and 280m

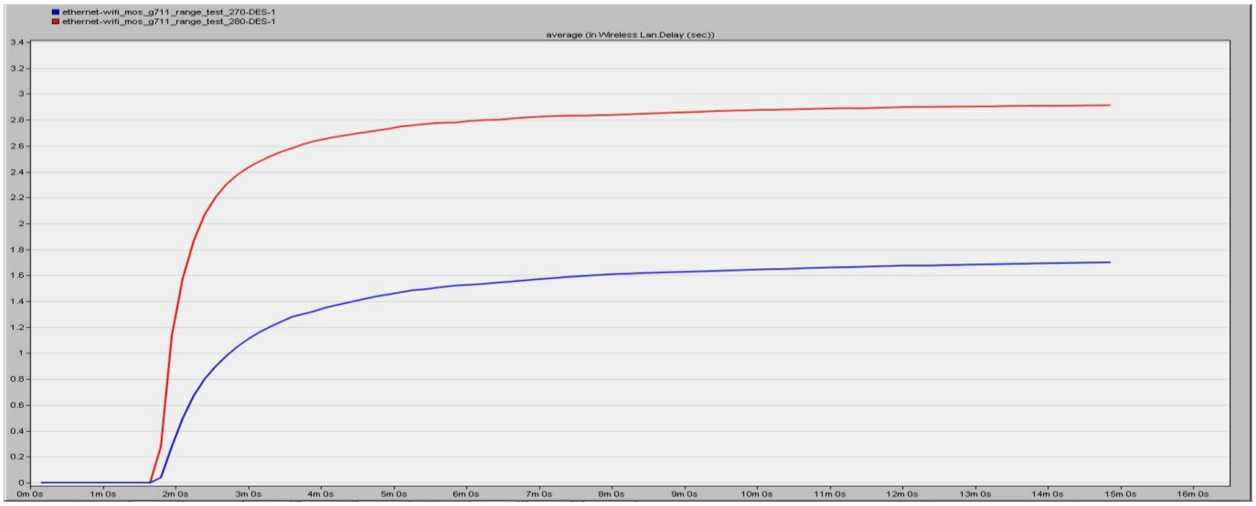


Figure 12

There is an important critical distance of about 265m where the delay increases exponentially even if the distance was further increased by a few meters as can be seen by the delay plots. This is easy to understand because if we increase the distance between the router and users, they will use more time to receive the packets.

Jitter as distance increases

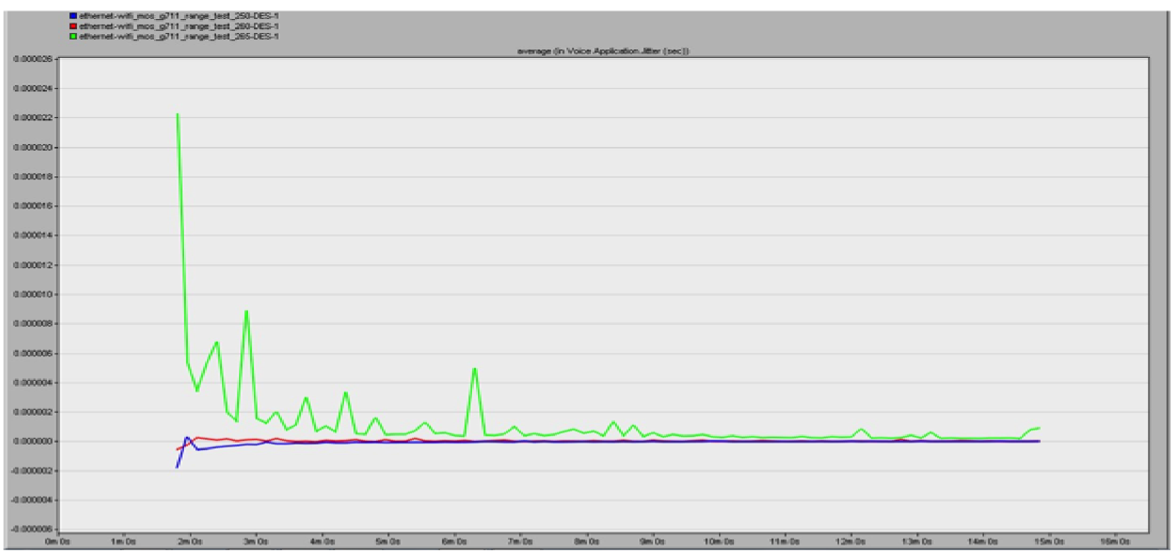


Figure 13

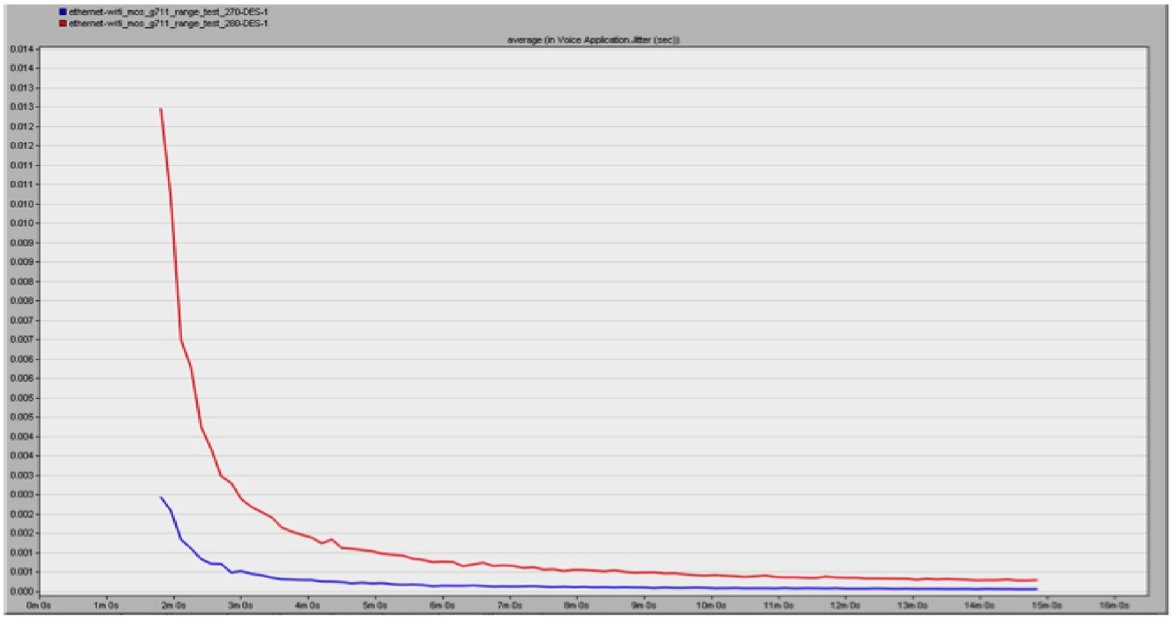


Figure 14

From the graph, if we increase the distance and we will get higher jitter. Jitter is directly related to how severe the traffic congestion is and therefore the queue delay. Initially, we thought congestion was due to the fact that the closer the client is to the router, the more packets arrive at the client per unit time, creating more congestion than if it was far away. However, from the queue size plots in figure 15, congestion is evident by looking at the queue size graphs below which increases exponentially as the distance increases proving why farther workstations receive more jitter.

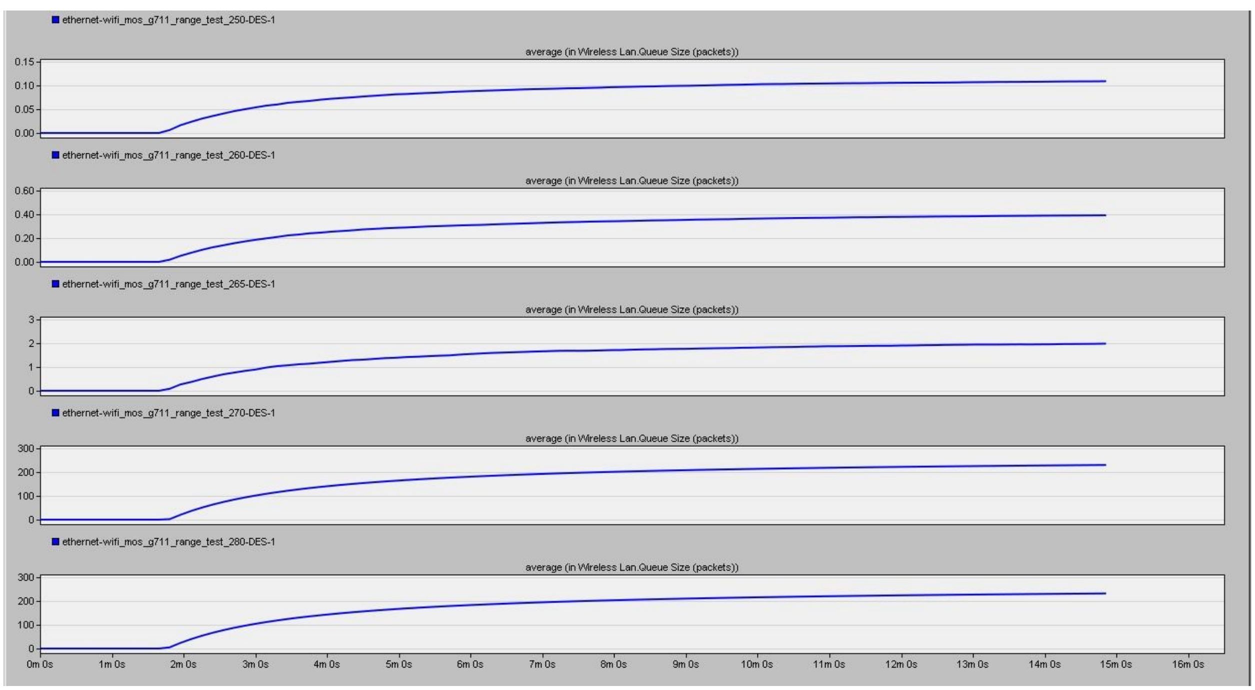


Figure 15

Packets loss as distance increases

Figure 16

From the graph, we can easily to see when we increase the distance, we will lose more packets due to weaker signal strength. To show why this is the case, if we increase the distance between the router and the user, it will lose more packets in. Furthermore, the MOS network loss rate which is the ratio of the packets lost due to network factors/out of sequence problems to the total number of packets increases in figure__ as the distance increases. Using the MOS network loss rate we can calculate the rough estimate of the number of lost packets by multiplying the MOS network loss rate by the total packets which is 200.

Distance(meters)	Packets Lost = MOS network loss rate x total packets	Packets Received = Packets Forwarded – Packets Lost
250	200 x 0 = 0 Packets	200 Packets
260	200 x 0.25 = 5 Packets	195 Packets
265	200 x 0.1 = 20 Packets	180 Packets
270	200 x 0.45 = 90 Packets	110 Packets
280	200 x 0.82 = 162 Packets	38 Packets

The table of calculated packets received values correspond quite similarly to the values of packets received in figure 16. The MOS network loss rates for different distances are in figure 17 below.

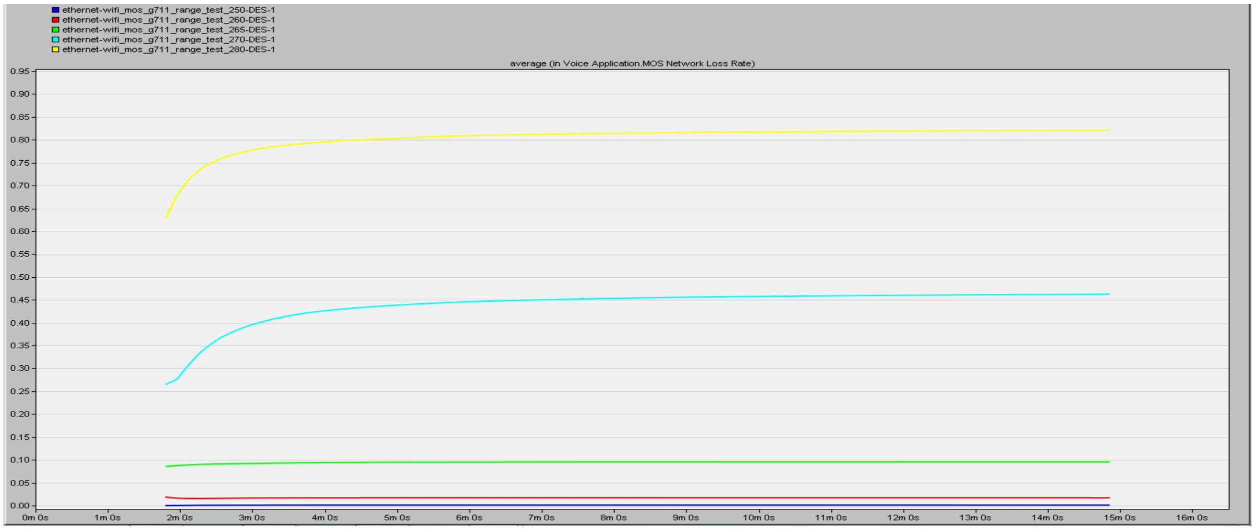


Figure 17

Another factor affecting voice packet loss is Dejitter loss rate due to the interarrival time being larger than the dejitter delay cause the receiving buffer to be overrun. However we have decided that the factor was not significant as the loss rate ratio was in the order of 10^{-3}. Keep in mind that the packet loss is also related to the jitter increase we just discussed in the previous section. When packets are dropped or discarded due to network factors (in this case the distance is stretching the network thin), the time difference of the packet arrival times of the packets after and previous to the dropped packet will have increased because the dropped packet does not exist anymore but the time interval allotted to it still remains. The change in the difference of the packet arrival times is the defitinition of jitter.

Throughput as distance increases

The point-to-point throughput of the 1000 BaseX ethernet connection between the server and the router of our topology in figure 1 is roughly 235000 bits/sec shown in figure 18.

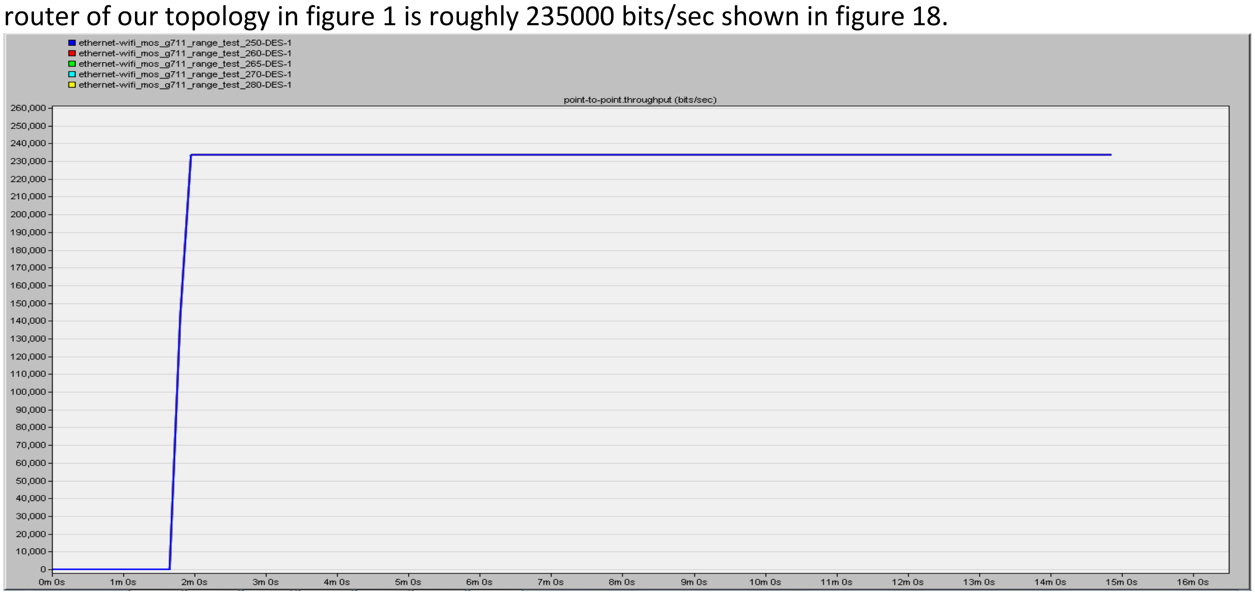


Figure 18

The throughput is determined by the utilization of the 1000 BaseX ethernet connection which is about 0.0235% shown below in figure 19.
Therefore the throughput above is justified by the utilization times the maximum data rate of the ethernet cable.
Throughput = Utilization x Maximum Data Rate = 0.0235/100 x 1000000000 bits/sec = 235000 bits/sec
Note that this is the same point-to-point throughput as displayed in the graph in figure 18.

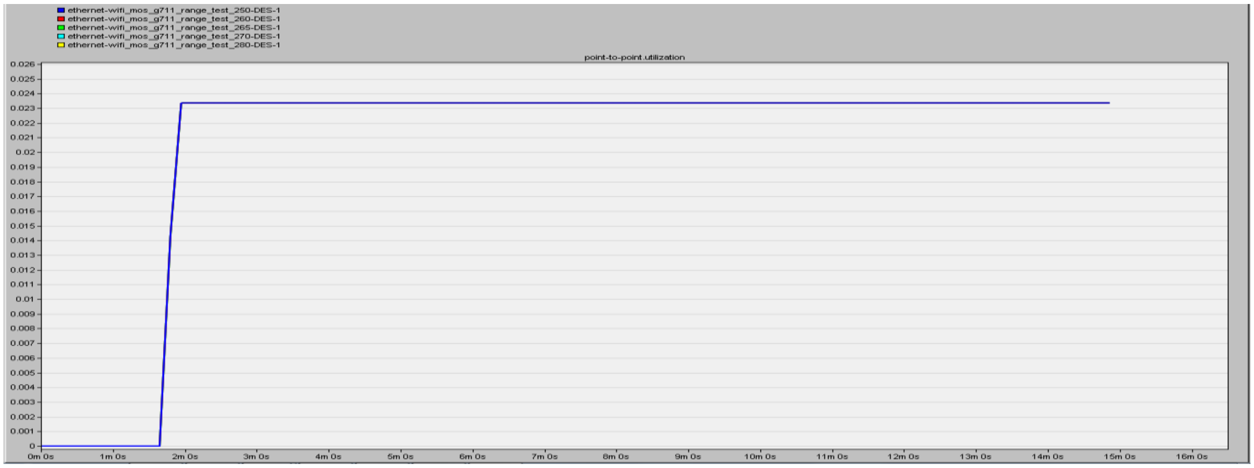


Figure 19

This throughput is further reduced in the MAC layer of our router node in the topology to about 190000 bits/sec. As the distance increases, thoughtput of the decreases after more and more bits are dropped due to buffer overflow and getting discarded after being retransmitted many times, exceeding the retransmission threshold.

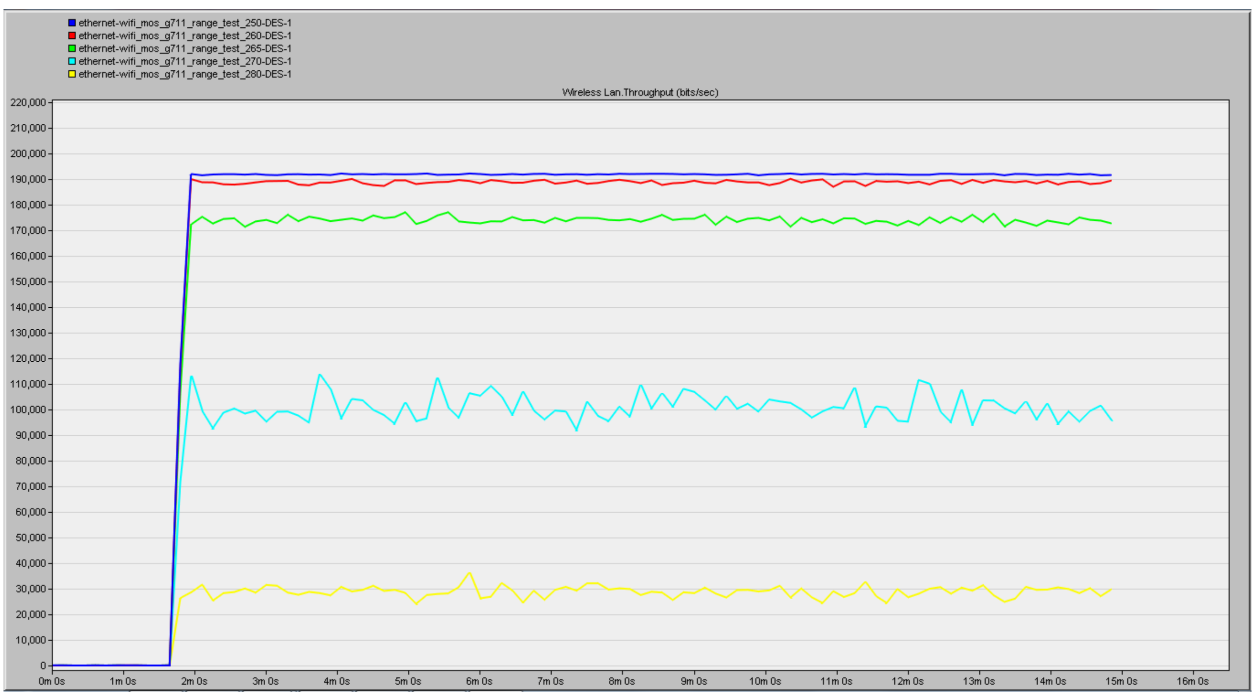


Figure 20

For example at 270 meters, the dropped due to buffer overflow , retry threshold exceeded, and reduced throughput due to dropped packets in displayed in figure 21 add up to about the 190000 bits/sec which is about the throughput transmitted to the network layer by the MAC layer in the router.

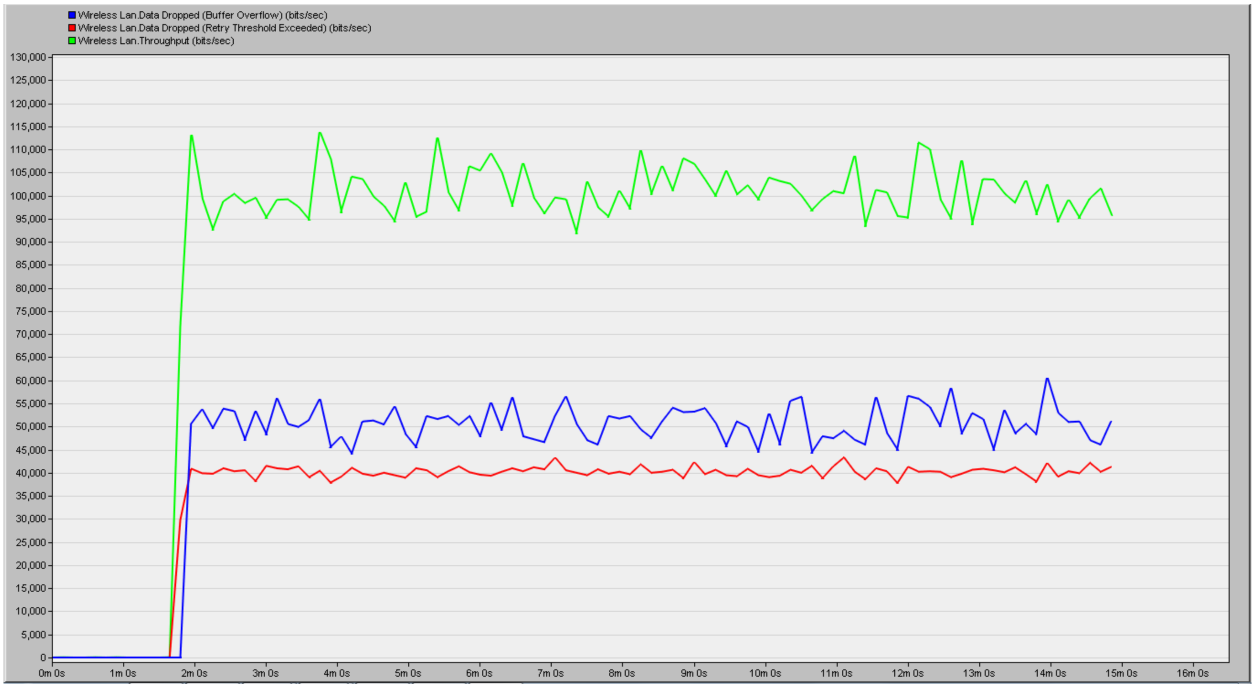


Figure 21

MOS values as distance increases

The MOS voice quality values are affected mainly by jitter ,delay, and packet loss. The farther distances have high values in ALL those factors and as a result will have the lowest mos values.

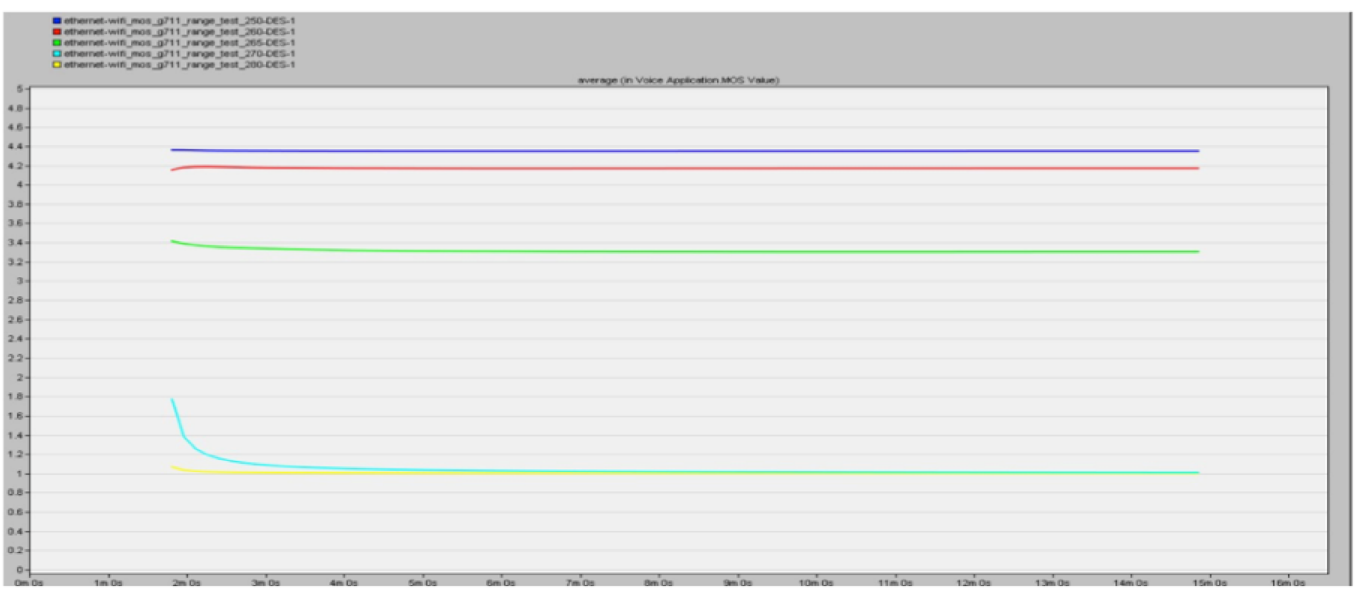


Figure 18

MOS value stands for Mean Opinion Score and is used to obtains the user's view of the quality of the network. Its range is 1 to 5 with 1 being the lowest quality:
MOS > 4.3 (Very Good) Range 30- 250 meters
3.5 < MOS < 4.3 (Good) Range 250 - 260 meters
3 < MOS < 3.5 (Fair) Range 260-265 meters
MOS < 1.3 (Bad) more than 270 meters

4.2

Wifi Router data rate effect anlaysis

We created 3 scenarios and choose 3 LAN router data rates which are 18Mbps 24Mbps and 54Mbps while keeping the user's data rate constant at 24 Mbps, the distances the same at 265 meters and the buffer at 256000 bits.

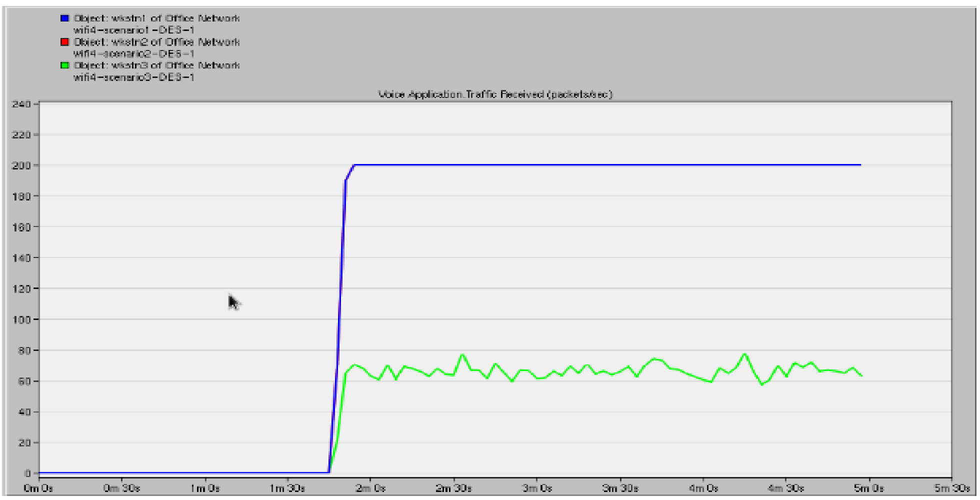


Figure 19

From the traffic received graph, when the router's data rate is 54 Mbps, the packet loss will increase. This is because the router sends the packets too fast at 54 Mbps, overrunning the receiver buffer and causing packets to be discarded.

We also provide the jitter results for the three different data rates. Due to the packets being discarded, the jitter is more severe for 54 Mbps scenario because packets being discarded causes a time gap due to the missing data and messes up the ideally constant interarrival timing of the packets necessary for low jitter. From figure 20 you can see the jitter spikes for the 54 Mpbs scenario.

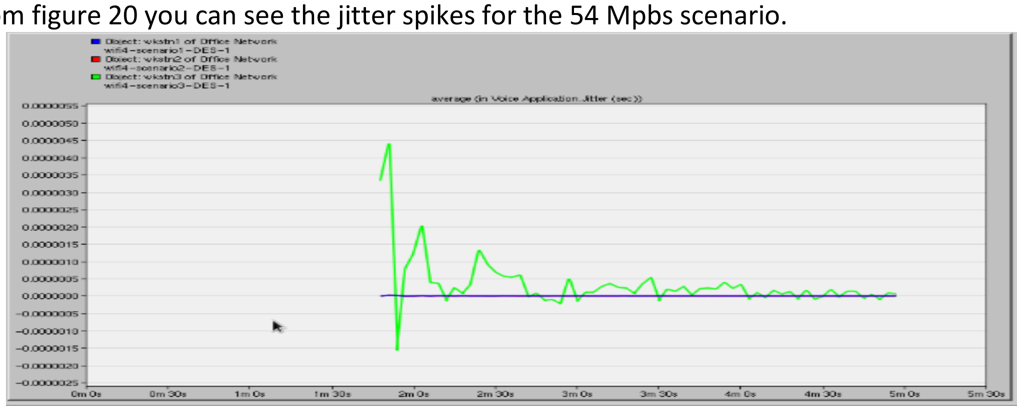


Figure 20

From the graph, the routers data rate at 54 Mbps created lot of congestion at the user end, causing a bit more jitter than the lower data rate cases. The queue size for the 54 bps data rate in comparison to others corroborates the evidence of congestion in figure 21.

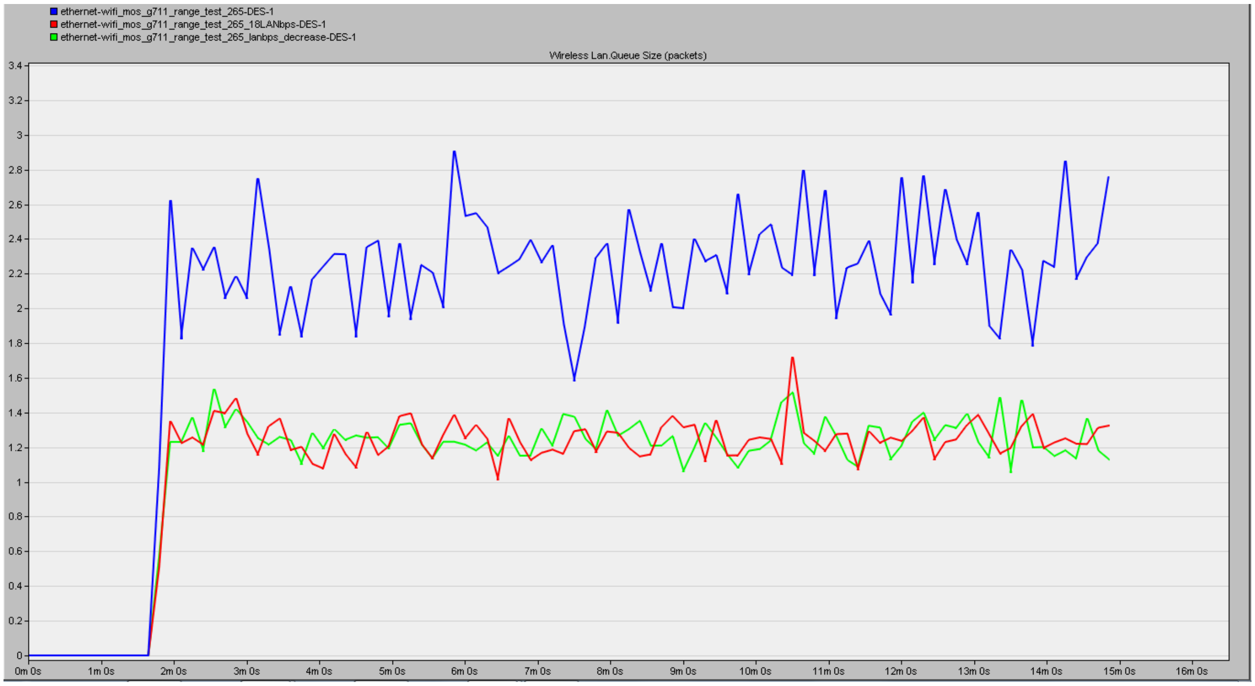


Figure 21

Finally, the delay of the 54 router bps case is the longest as shown in figure 22. Contributions to the delay by the much higher media access delay and retransmission delay can be seen from our results. The media access delay (shown in figure 22) which is created by contention between packets for channel access seems to make up more than 50% of the total delay in figure 23

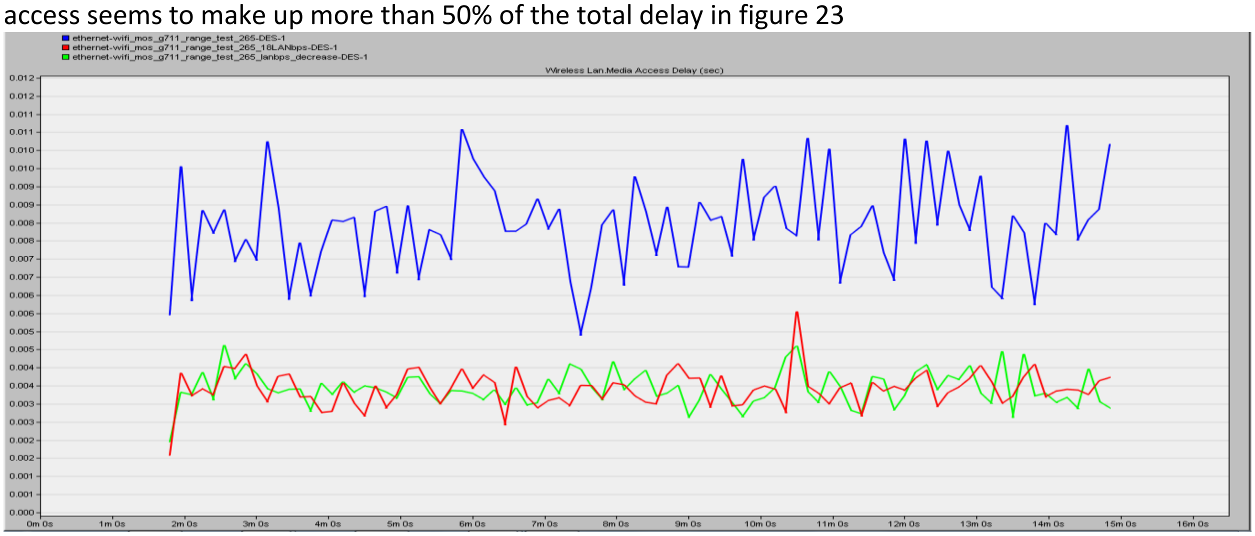


Figure 22

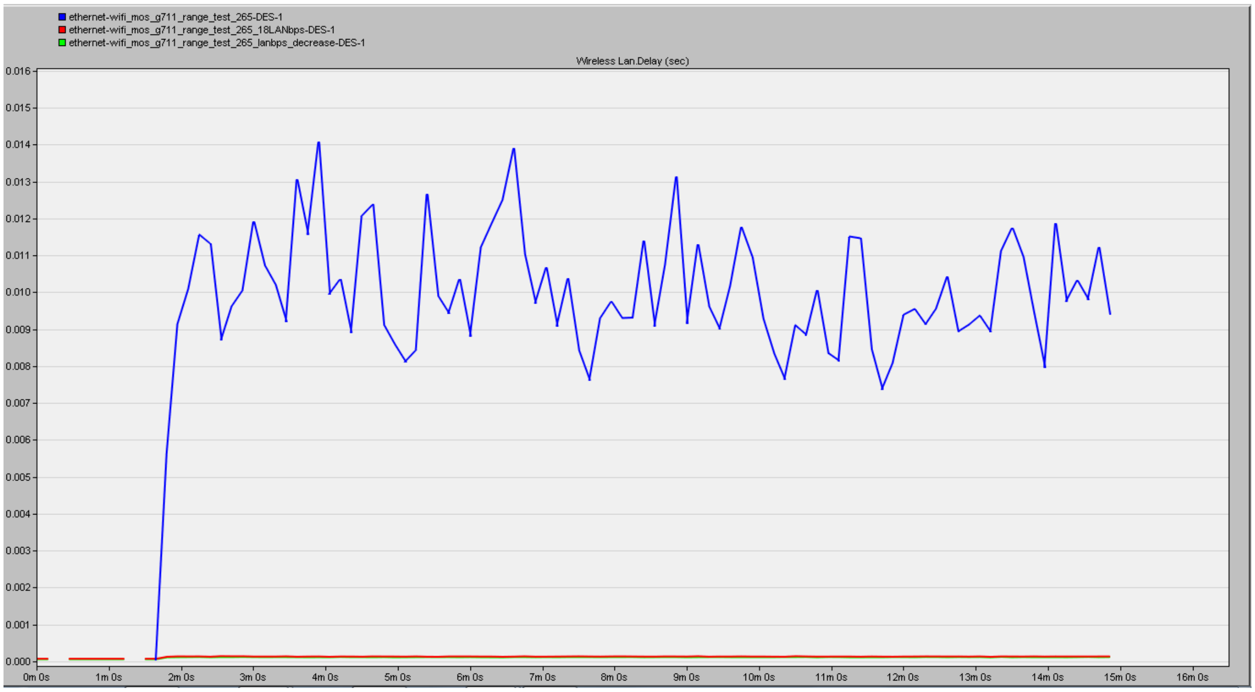


Figure 23

4.3 Wifi Buffer Analysis

We create 2 scenarios and choose 2 different receiver buffers which are 256000 bits and 1024000 bits for the client workstation. We keep the distance the same as 265 meters and both the user workstation data rate and the router data rate at 54 bps.

We will see below that increasing the receiver buffer will remedy the problem of packet loss and jitter. For the lower buffer scenario, we can see that from figure 24 that the voice traffic received is less than traffic sent. Of course getting fewer packets received than sent for the voice application doesn't necessarily mean that the packets not received are lost. However if we look at the MOS network loss rate in figure 25, we will see that the scenario with the lower receiver buffer has a 0.1 ratio of network-related packet loss to total packets received which means that those packets were indeed discarded. In the high buffer scenario, the voice traffic sent are 100% received by the application and the MOS network loss rate is minimal.

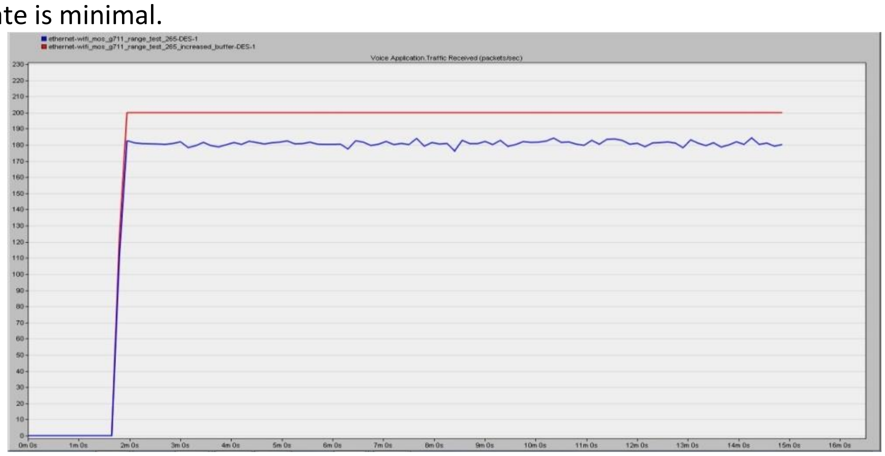


Figure 24

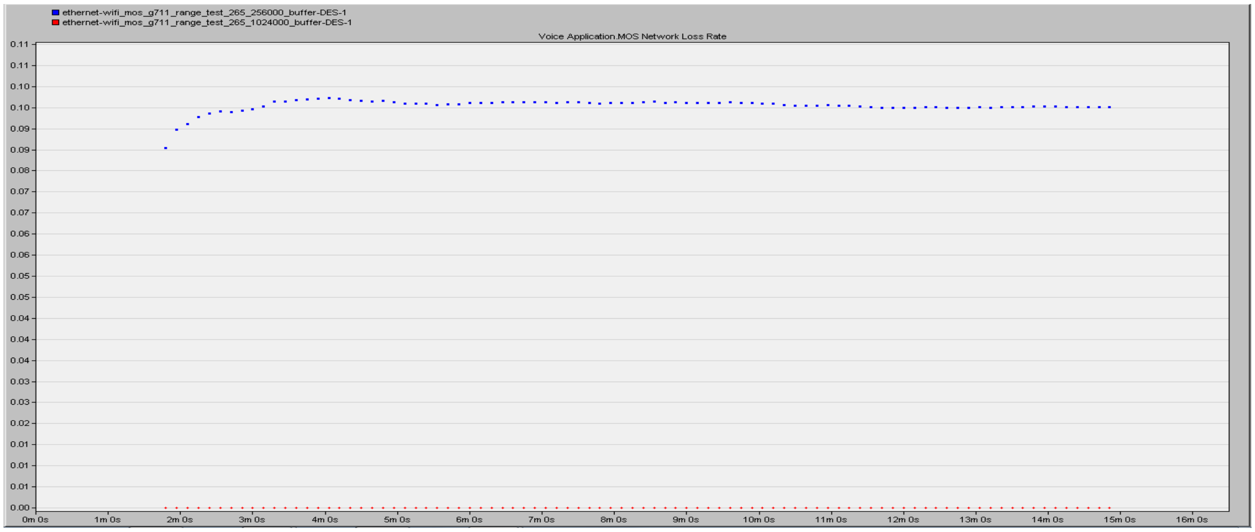


Figure 25

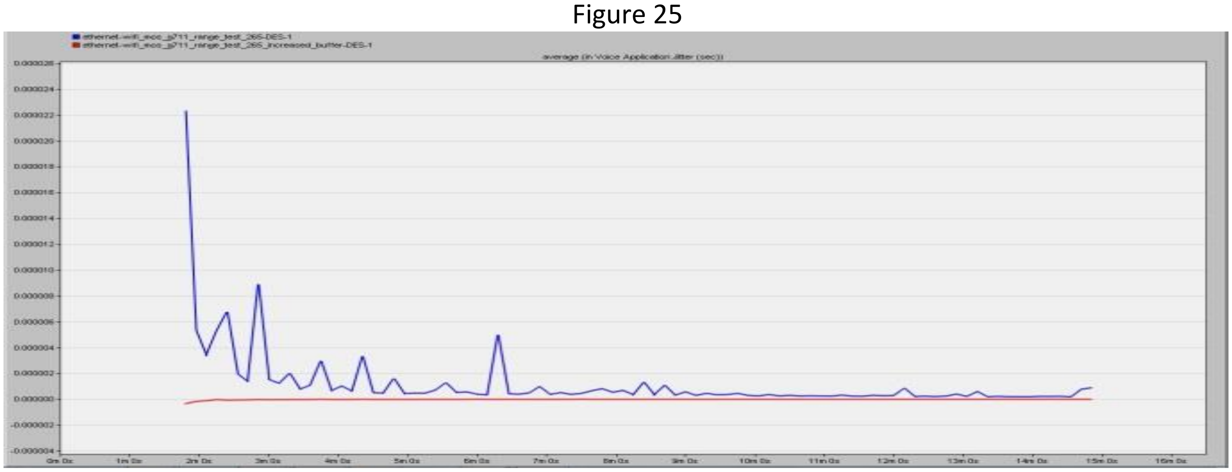


Figure 26

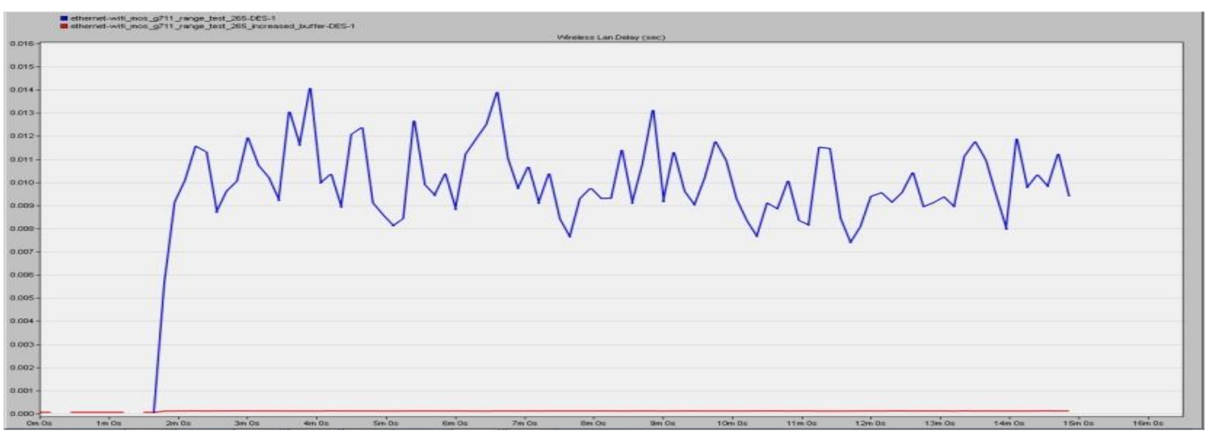


Figure 27

Finally we can see that the MOS value for the 1024000 buffer scenario is higher due to low jitter and delay.

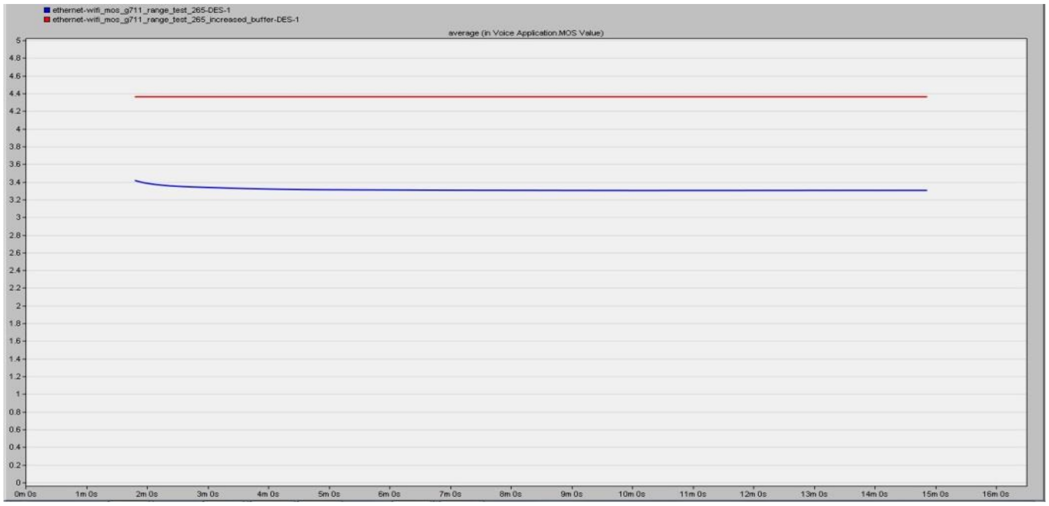


Figure 28

4.4 G.711 vs. G.729a

The resulting MOS values of the G.711 and G.729a when the client workstation is close to the router were similar to the MOS values obtained using ITU-T standard PSQM testing algorithm under ideal conditions. The discrepancies may be due to the lack of detailed parameters on Riverbed regarding the two voice encoding schemes and also the fact that the simulations were not under ideal conditions (some jitter and delay exists).

	G.711 Mos Value	G.729a Mos Value
Riverbed Result	4.36	4.04
PSQM standard MOS	4.45(A-law)	4.02

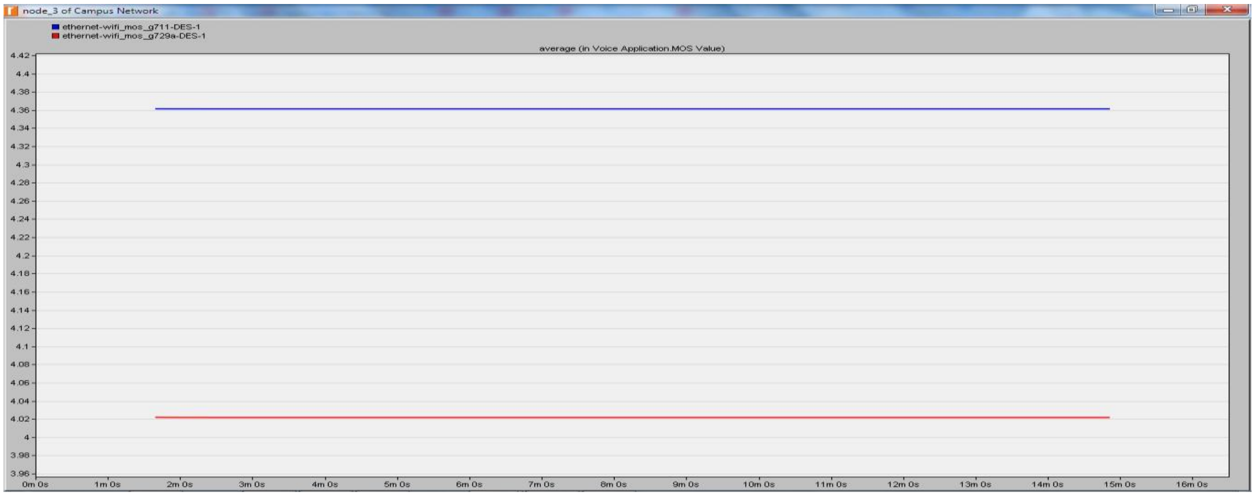


Figure 29

When we placed the client workstation 270 meters away from the router and the user client exxperiences some packet loss, we find that G.711 is able to maintain higher MOS voice quality at 270 meters than G.729a since G.729a is a compression codec as illustrated in figure 31. In other words, G.711 is more tolerable to packet loss than G.729a.

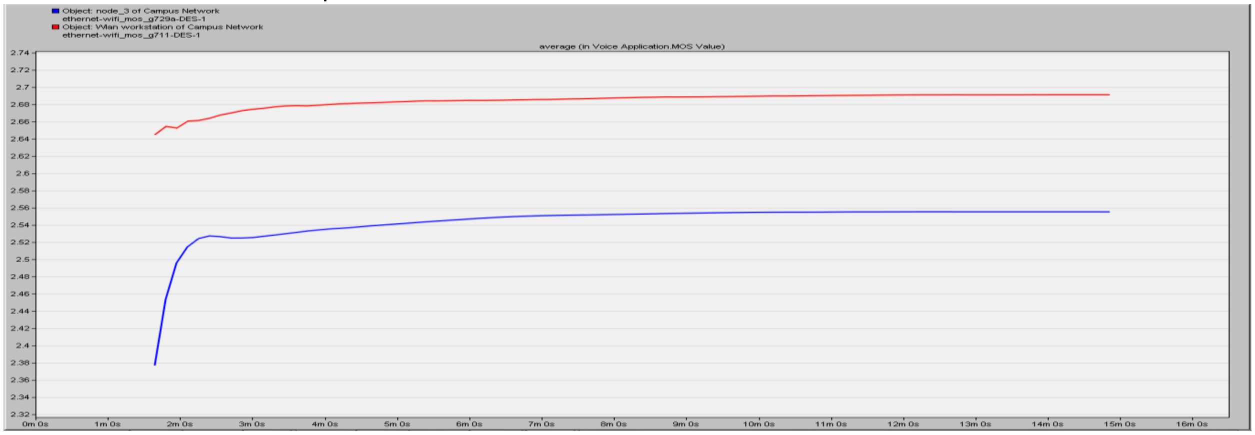


Figure 30

The tradeoff with using the G.729a compression codec is that although the MOS performance is not as good as G.711, the bandwidth it takes up only 8kbits/s instead of the 64 kbits/s like the G.711 codec. This means the bandwidth comsumed is 8 times less as shown in figure 31 where the G.711 application sends 16000 bytes while the G.729 application sends only 2000 bytes.

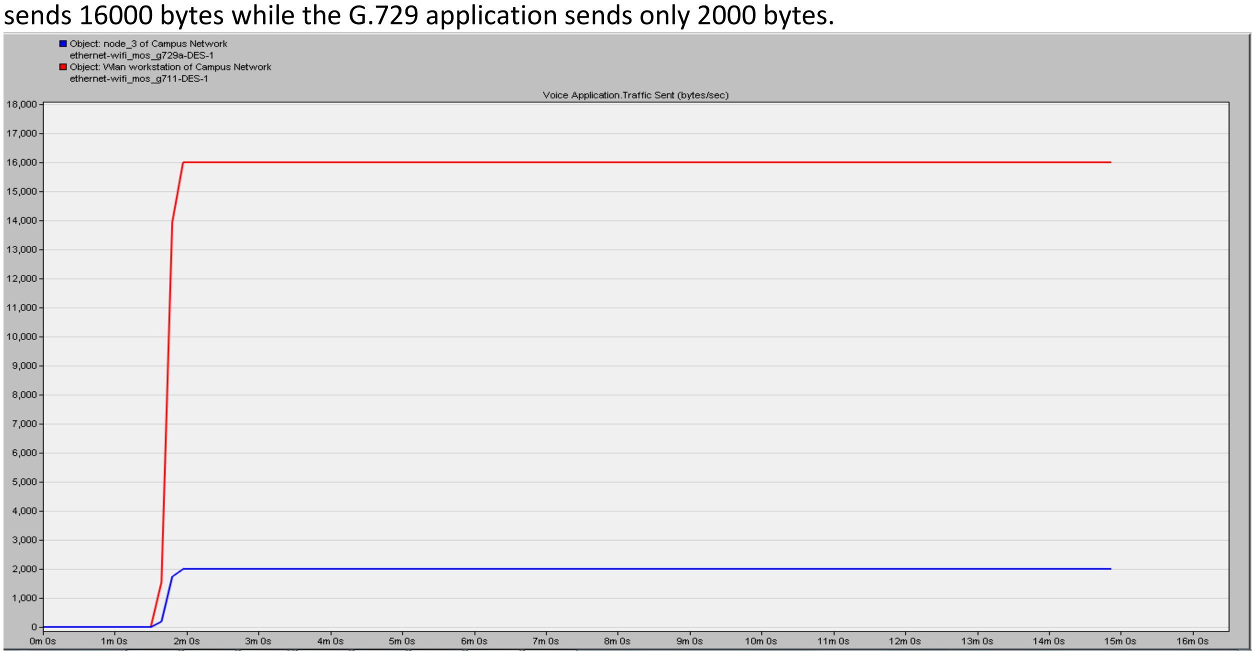


Figure 31

Extra: Using TCP vs UDP for Voip over Wifi

By default, voice applications on Riverbed are defaulted to using UDP as the transport protocol as we have been using in all our scenarios. To change the transport protocol to TCP we go the application attribute of the WLAN workstation in our topology shown in figure 32.

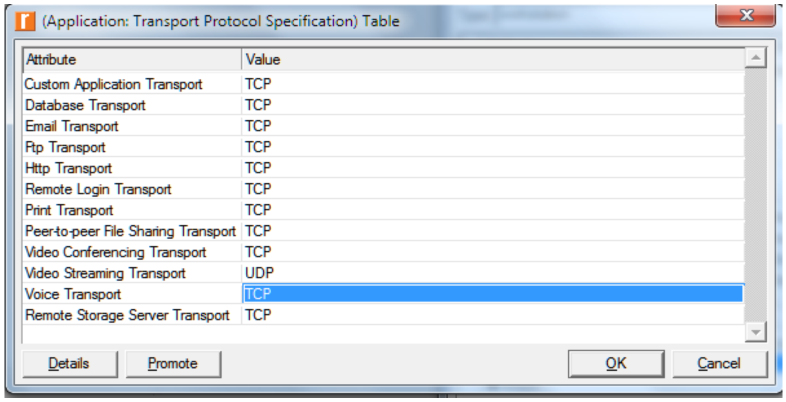


Figure 32

TCP in Voip applications is not recommended because Voip communications does not need a perfect transport layer protocol and common bit errors or packet loss only slightly impacts audio quality. Since most algorithms that TCP utilizes for congestion control such as slow start, congestion avoidance, fast retransmit and fast recovery are all triggered when there is even only small packet loss, TCP will introduce delay in order to retransmit lost segments. Not only will this cause delay which cannot be afforded in a real-time service such as Voip but the retransmitted segments will also congest the network and cause lots of jitter which as we have learned, will affect MOS voice quality. As a result, UDP is better because while it does not have congestion control or error checking, it can keep a voice stream real-time and does not delay the Voip session for seconds trying to retransmit lost packets.

As a demonstration, we place the WLAN workstation in our usual figure 1 topology at 240 meters where we found out in our results that there should be no voice packet loss (minimal MOS network loss rate) with UDP as the default transport protocol. However when we used TCP as the transport protocol, the jitter to due congestion from retransmissions became increased rapidly as the simulation went on shown in figure 33. In fact the jitter value of 0.21 seconds is much higher than the suggested jitter. As a consequence, the MOS value declined quite fast due to the rising jitter whereas the UDP MOS value was stable shown in figure 34. The Riverbed simulation actually aborted due to reaching the retransmission limit shown in figure 35.

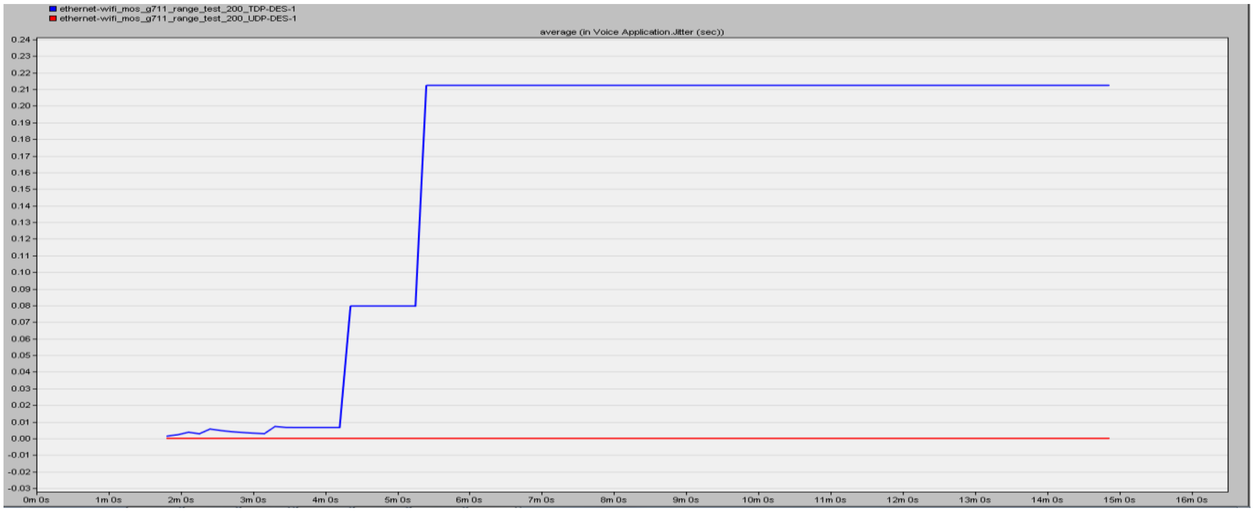


Figure 33

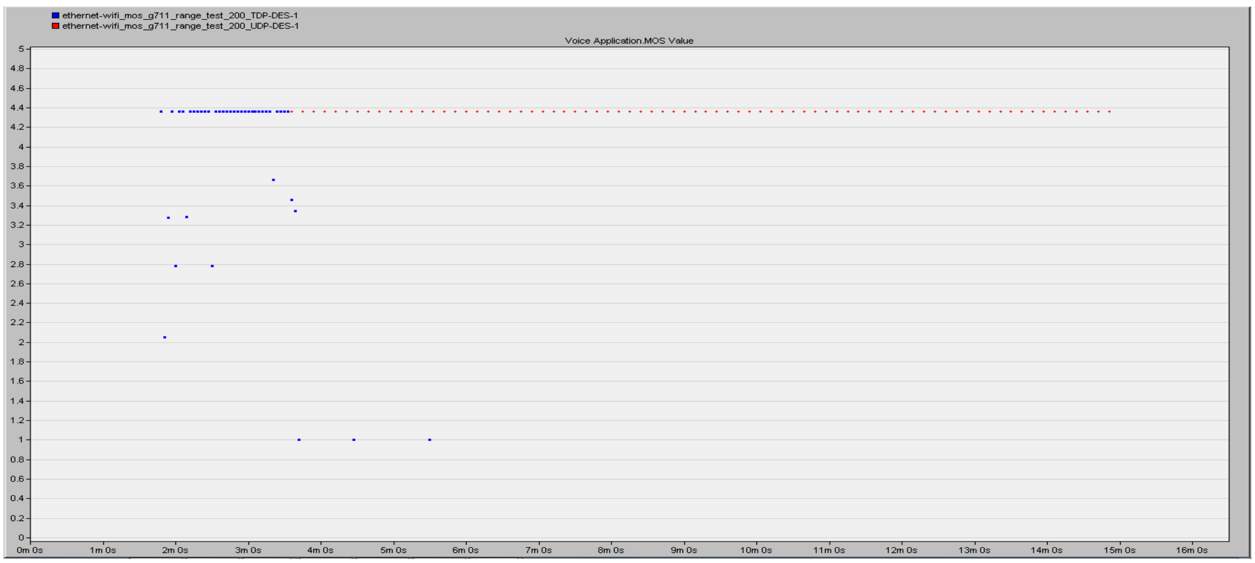


Figure 34

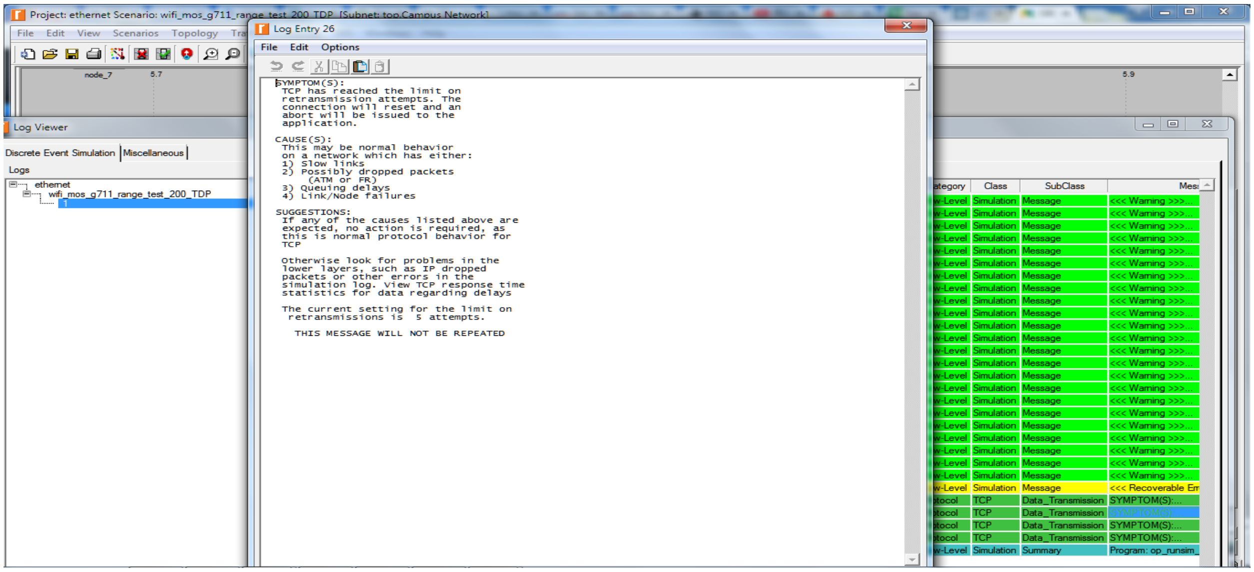


Figure 35

5.Conclusion

From our simulations we have learned that the 3 aspects jitter,delay, and packet loss are the major factors that affect the voice quality of Voip and those 3 aspects are also interrelated or go hand-in-hand. For example, packet loss will cause jitter since the discarded packets will cause a gap in the data stream and change time between the packets arriving at the receiver (jitter). Jitter also can cause packets to be dropped when jitter causes packets to be out of sequence. Delay due to being far away can cause packet loss due to network factors at far distances (MOS Network loss rate). From our 5 chosen distances for many of our scenarios, the client stations at the distances farther than our critical distance which we estimate is at 265m will receive bad signal quality and the MOS values at those distances will exponentially get worse. In other words, MOS values in VOIP over Wifi are only good if the receiver is within range of the router and receives good signal quality. The MOS score for voice quality also depends on the voice codec used. If bandwidth is abundant, G.711 is the best choice as the MOS score under ideal conditions is much higher and more importantly, G.711 is much more tolerant to jitter increase or packet loss.

6. References

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