ENSC 427: COMMUNICATION NETWORKS
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FINAL REPORT

PERFORMANCE ANALYSIS OF VOICE OVER LTE USING OPNET

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Abstract

LTE is a new standard for wireless data communications intended to significantly improve the previous 3G standard. Unlike 3G, LTE only supports packet switching, making LTE incompatible with 3G and 2G. This introduces a new challenge, because previously all voice calls relied on circuit switched networks. Voice over LTE (VoLTE) has emerged as the leading solution for delivering voice services. In this project, we simulated voice calls with varying amounts of network congestion, and analyzed the impact this had on packet loss, end-to-end delay, jitter, and mean opinion score. We found that only extreme amounts of congestion caused significant amounts of packet loss, that delay increases exponentially with congestion, that jitter is practically unaffected by congestion, and that mean opinion score suffers significantly with increased congestion.

1. Introduction

1.1 Background

LTE is known as the global standard for the fourth generation of mobile broadband or 4G. LTE plays a big part in the global economy, since it enables the user to experience superior stability, throughput, and latency for long range wireless communication. In 2004, LTE was first developed by 3GPP, and launched in late 2009. LTE is part of the GSM evolutionary path of mobile broadband, whose main objective is to provide an extremely high performance radio-access technology. It considers a full Internet Protocol (IP) network architecture that is designed to support voice in packet domain. While 3GPP focuses on HSPA+, LTE focuses on the next generation of LTE for the International Telecommunication Union's (ITU) IMT-Advanced requirement, even though they both were developed nearly simultaneously by 3GPP standards.

Since LTE relies entirely upon packet switched networks, it is inherently incompatible with its predecessor technologies 2G and 3G which use circuit switched networks for making voice calls. This incompatibility demanded a new solution for making voice calls over an LTE network. The obvious answer seemed to be Voice-over-IP (VoIP), which is an already well established and proven technology. However, VoIP has some serious problems. VoIP relies on the Internet to deliver packets, which means a high Quality-of-Service (QoS) cannot be guaranteed since the Internet operates on a "best effort" basis. This is clearly unsuitable to be the main means of voice communication for all modern cellular phones. This is where Voice over LTE (VoLTE) comes in.

VoLTE is LTE with specific profiles and media planes of voice services. It is the core network's standard architecture in IP era because it supports 4G LTE operators in order to offer voice, video and messaging services, and allows a drastic cost saving and recover inefficient spectrum for additional data capacity.

1.2 Objective

In our project, we will use OPNET 16.0 to implement mobile nodes on an LTE network and compare and analyze the performance as network congestion increases. We will use varying amounts of FTP traffic to simulate increased network congestion. To analyze the quality of services, we will examine several parameters: packet loss, end-to-end delay, jitter, and mean opinion score (MOS). The parameters are explained below.

Packet Loss

Packet loss is the rate of failure when packets do not arrive at their destination. Similarly to VoIP, LTE also has a packet loss rate, but it is intended to be maintained low enough to give sufficiently good quality.

End-to-End delay

Delay is the amount of time taken between when the packet is sent and when the packet is received. This is usually affected by network performance and the distance of the sending node to the receiving node.

Jitter

Jitter is the variation of arrival time between each packet, or the time difference between when the packet is supposed to be received compare to when it actually arrives.

MOS

MOS or Mean Opinion Score is the test of user's view of the quality of the telephony networks. It is graded of a scale of 1 to 5, from bad to excellent, based on the subjective measurement of the quality of the call.

Standards of Parameters

The International Telecommunication Union (ITU) is the specialized agency that is responsible around the information and communication. This agency made a standard for telecommunication that is called ITU-T or ITU Telecommunication, and the value for the average and ideal quality for the parameter, shown in Table 1.2.1. Table 1.2.2 shows the various rankings for mean opinion score.

Parameter	Average Quality	Ideal Quality
Jitter	<60 ms	<20 ms
End-to-End Delay	<150 ms	<50 ms
Packet Loss Rate	<5%	<1%

Table 1.2.1: ITU Standards for parameters [1]

Quality Scale	Mean Opinion Score (MOS)
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

Table 1.2.2: MOS Scale [1]

2. Implementation

2.1 Topologies and Configurations

Our project consists of two network topologies, shown in Figures 2.1.1 and 2.1.2. The first topology is used for the voice only configuration to get our baseline values for the parameters of interest. The second topology adds FTP nodes which send and receive files via FTP to add network congestion. We use three different file sizes for varying amounts of network congestion, giving us three more configurations. All configurations are summarized in Table 2.1.1. All simulations for each configuration are run for 30 minutes.

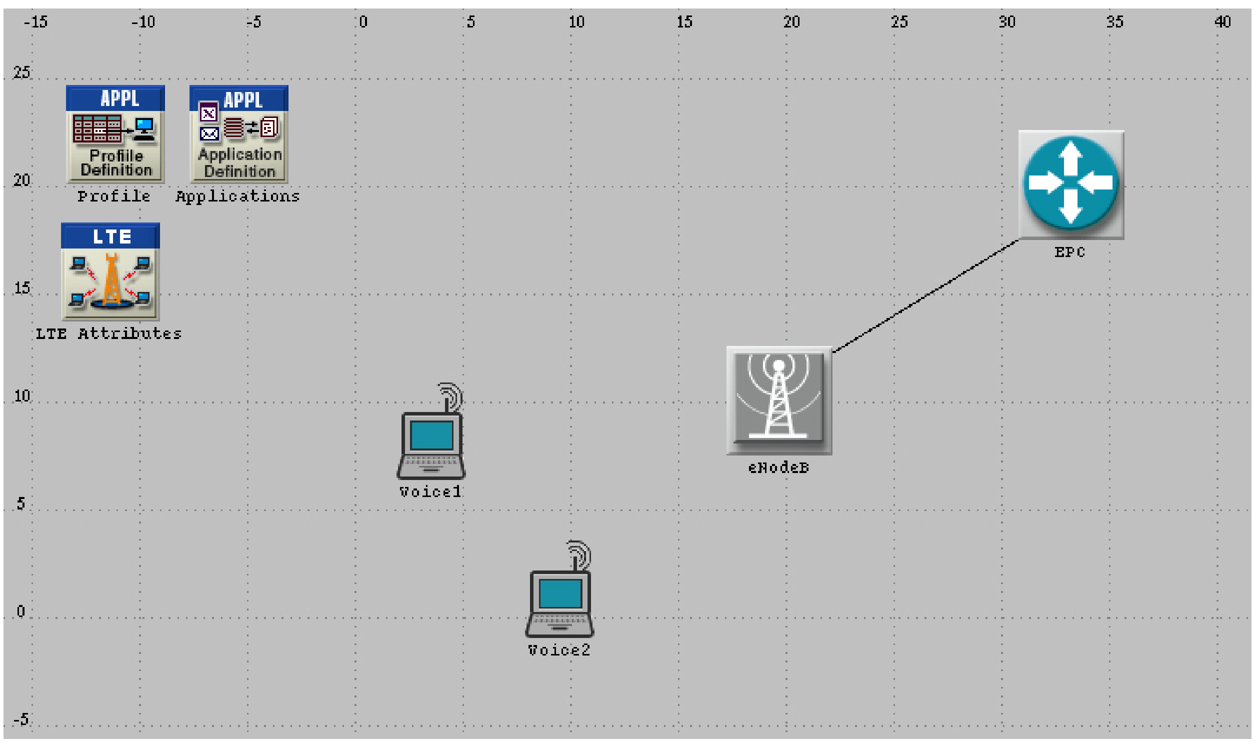

Figure 2.1.1: Topology used for voice only configuration

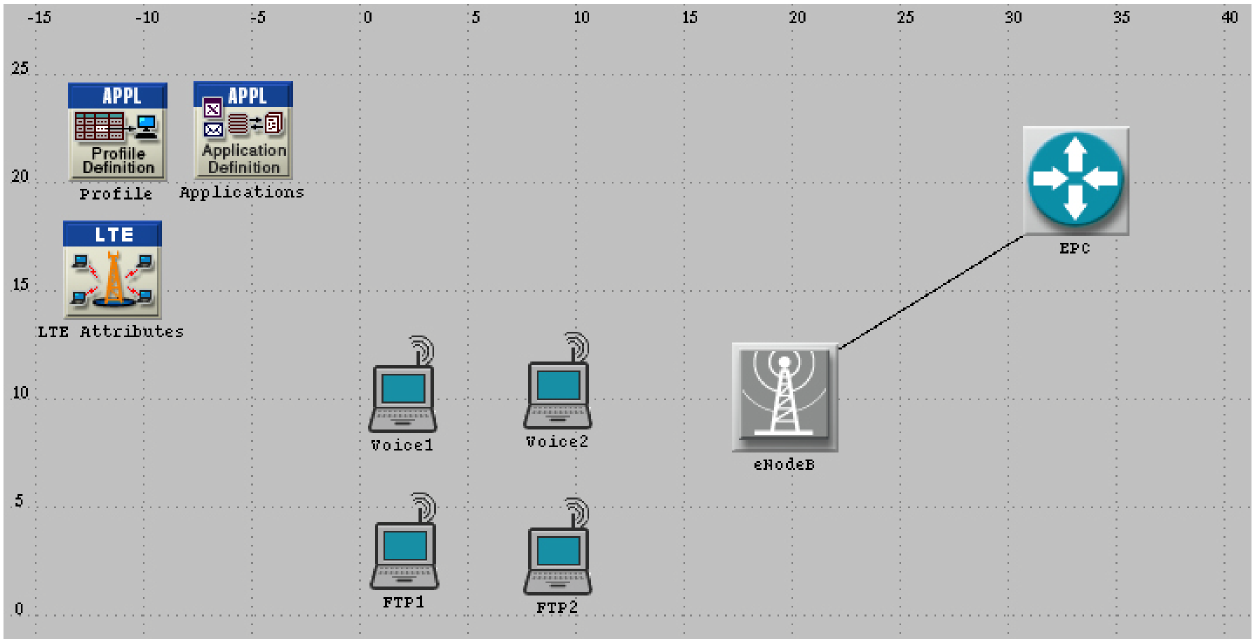

Figure 2.1.2: Topology used for voice and FTP configurations

Configuration	File Size (KB)
Voice Only	0
Voice and Light FTP	100
Voice and Heavy FTP	500
Voice and Massive FTP	1000

Table 2.1.1: Summary of all simulation configurations

2.2 Models

All of our models are from the lte_adv library which was part of our OPNET 16.0 installation. The specific models we used are lte_enodeb_atm4_ethernet4_slip4_adv for the eNodeB, lte_epc_atm8_ethernet8_slip8_adv for the EPC, ppp_adv for the link, and mobile workstation for the voice and FTP nodes. All model attributes were left as defaults, except for the mobile workstations which were set to support either the voice or FTP application. The settings for the voice and FTP applications and profiles are shown in Figures 2.2.1 through 2.2.3.

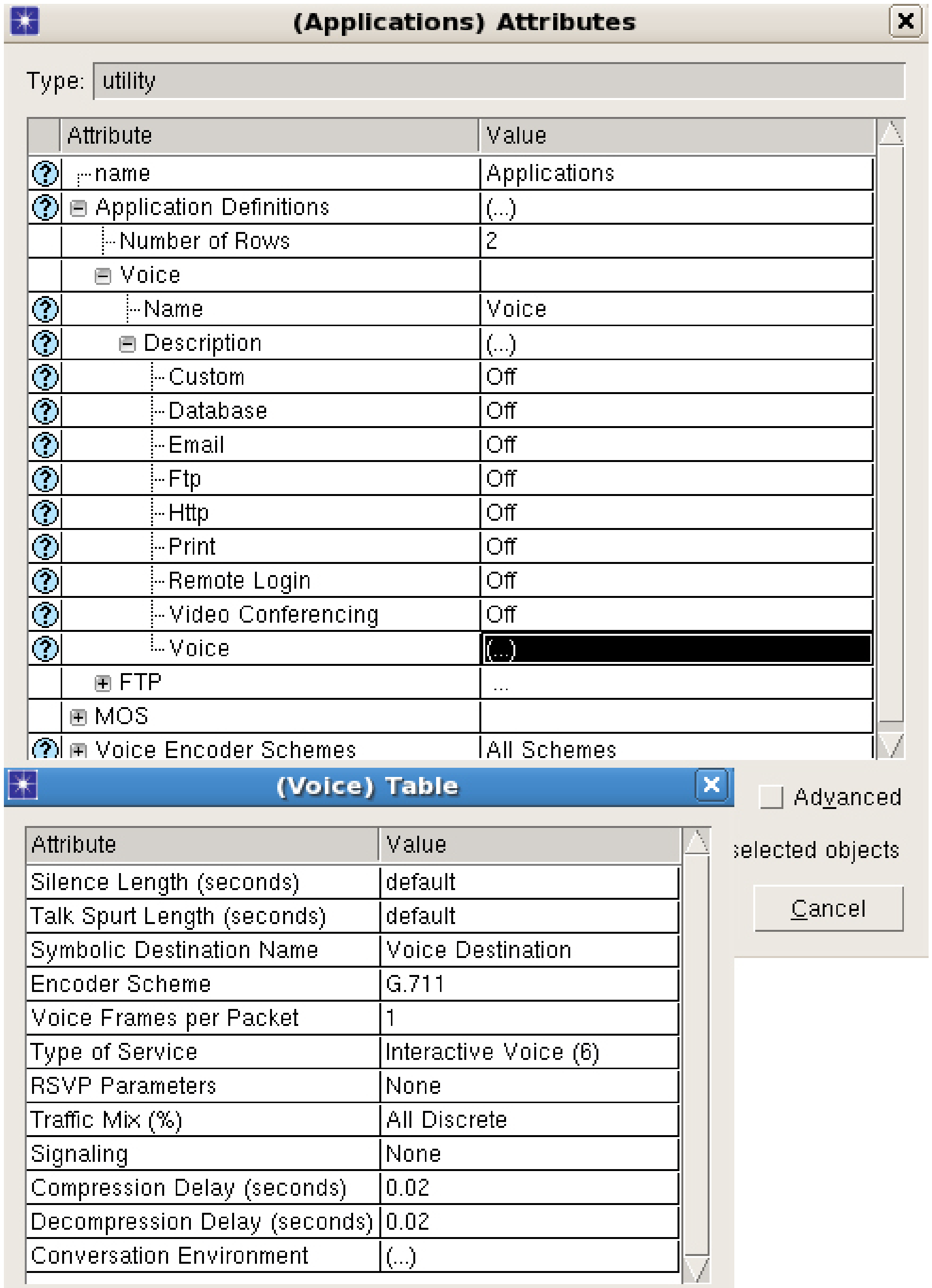

Figure 2.2.1: Settings used for voice application

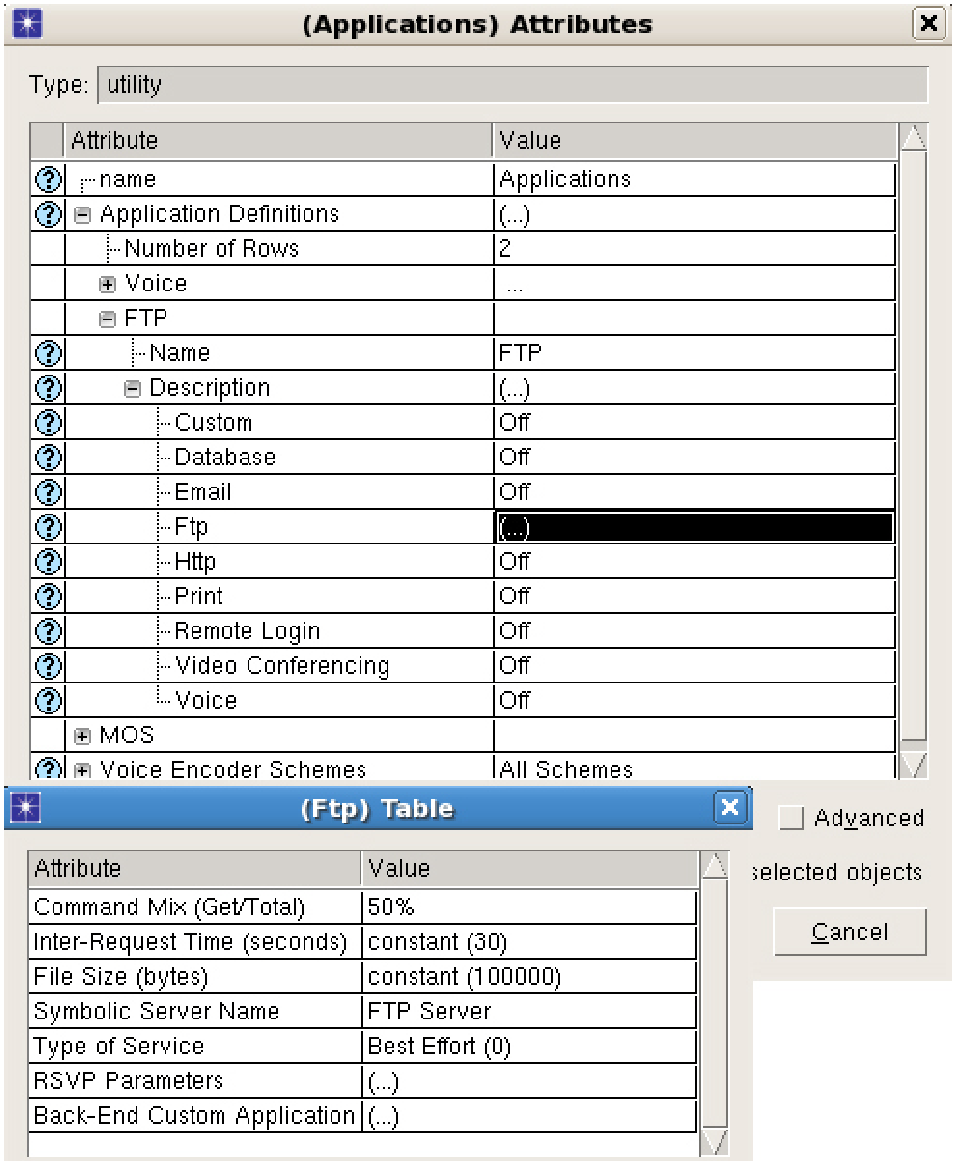

Figure 2.2.2: Settings used for FTP application, not that the File Size field is changed depending on the configuration

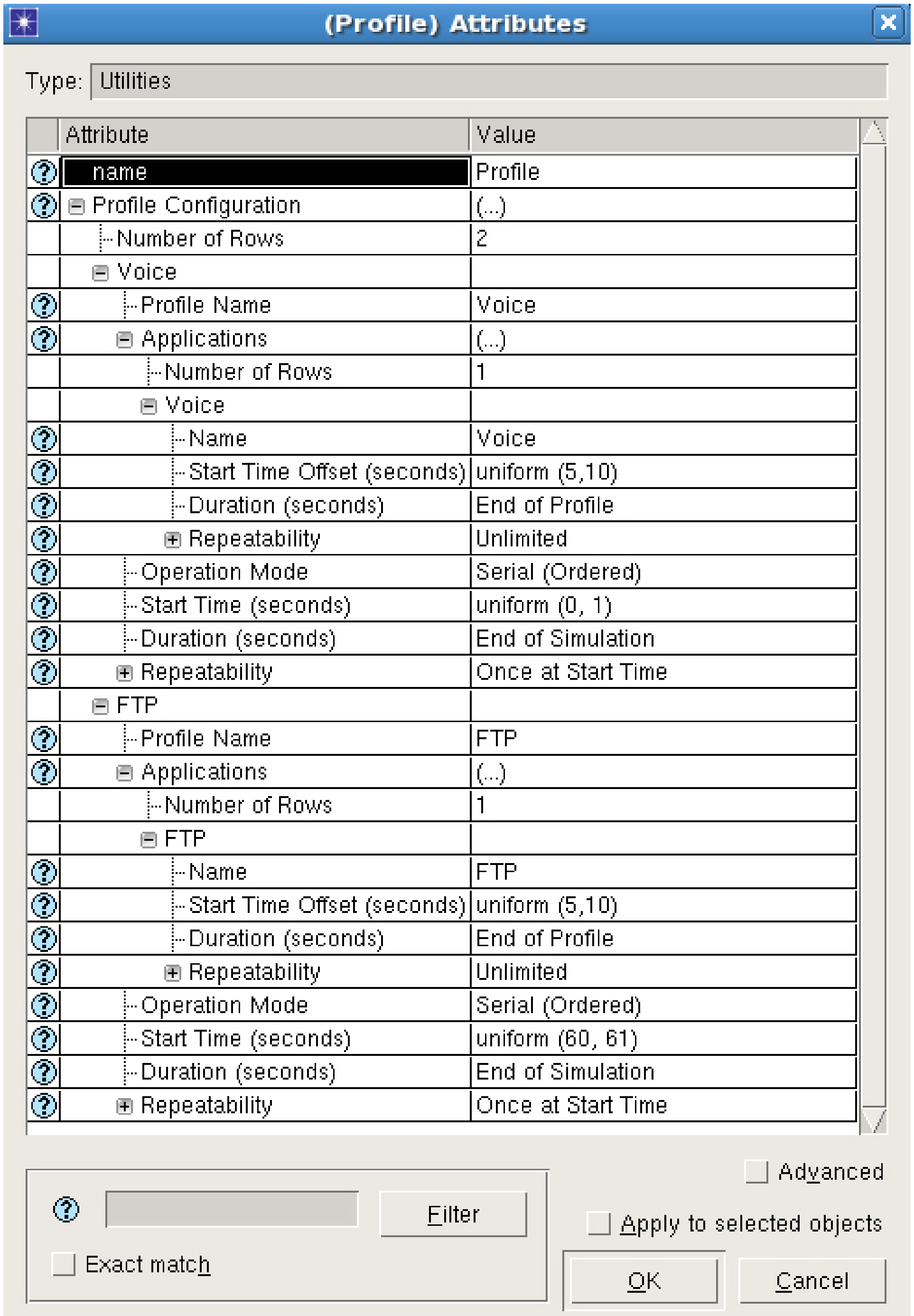

Figure 2.2.3: Settings used for profile configuration

3. Simulation Results and Discussion

3.1 Voice Only Configuration

Figure 3.1.1 shows the network load for the voice only configuration. Although this figure does not contain any results of interest, it mainly serves to give some additional information about how the simulation was run and verify that the simulation was working correctly. A point of interest is how the load at the eNodeB is the sum of the loads of the two voice nodes, as should be expected.

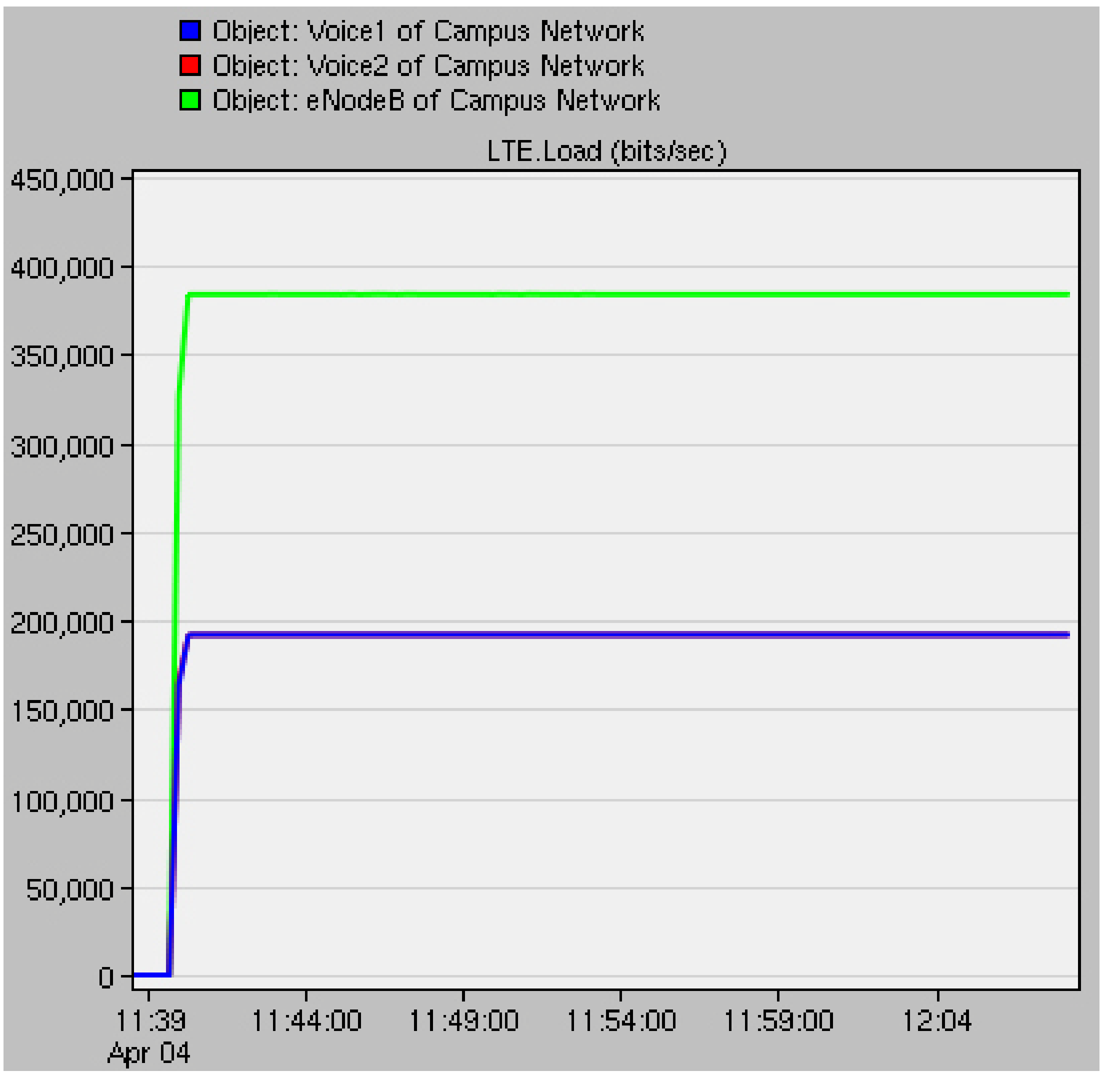

Figure 3.1.1: LTE network load for voice only configuration

Figure 3.1.2 shows the sent and received voice traffic between the two voice nodes. From this figure we can see that the sent and received traffic are equal, so there is no packet loss, which meets ideal quality standards.

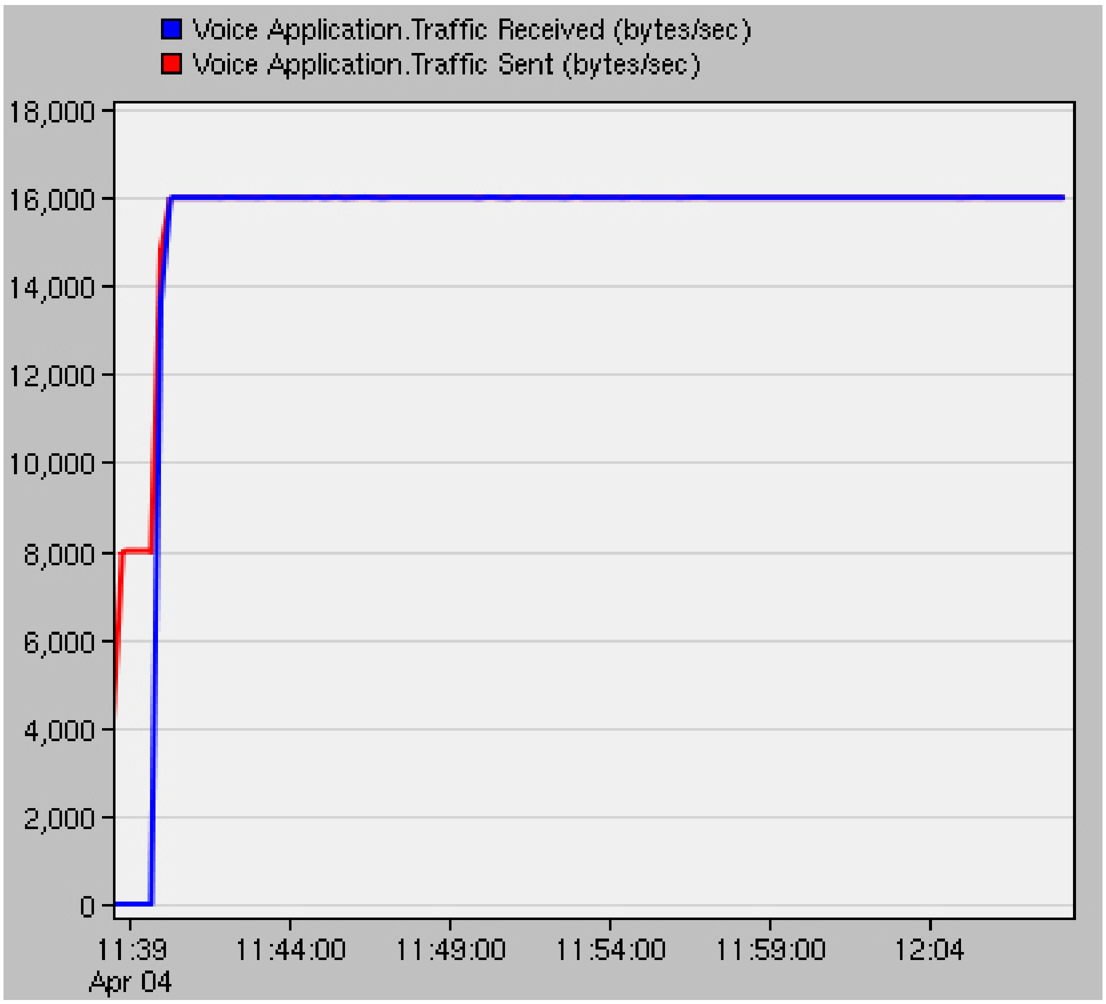

Figure 3.1.2: Sent and received voice traffic for voice only configuration

Figure 3.1.3 shows the end-to-end delay for each of the voice nodes. Although the delays are different, they are both very similar with steady state values of 77 ms for the Voice1 node, and 75 ms for the Voice2 node. Both of these values fall outside of ideal quality, but are within the average quality range.

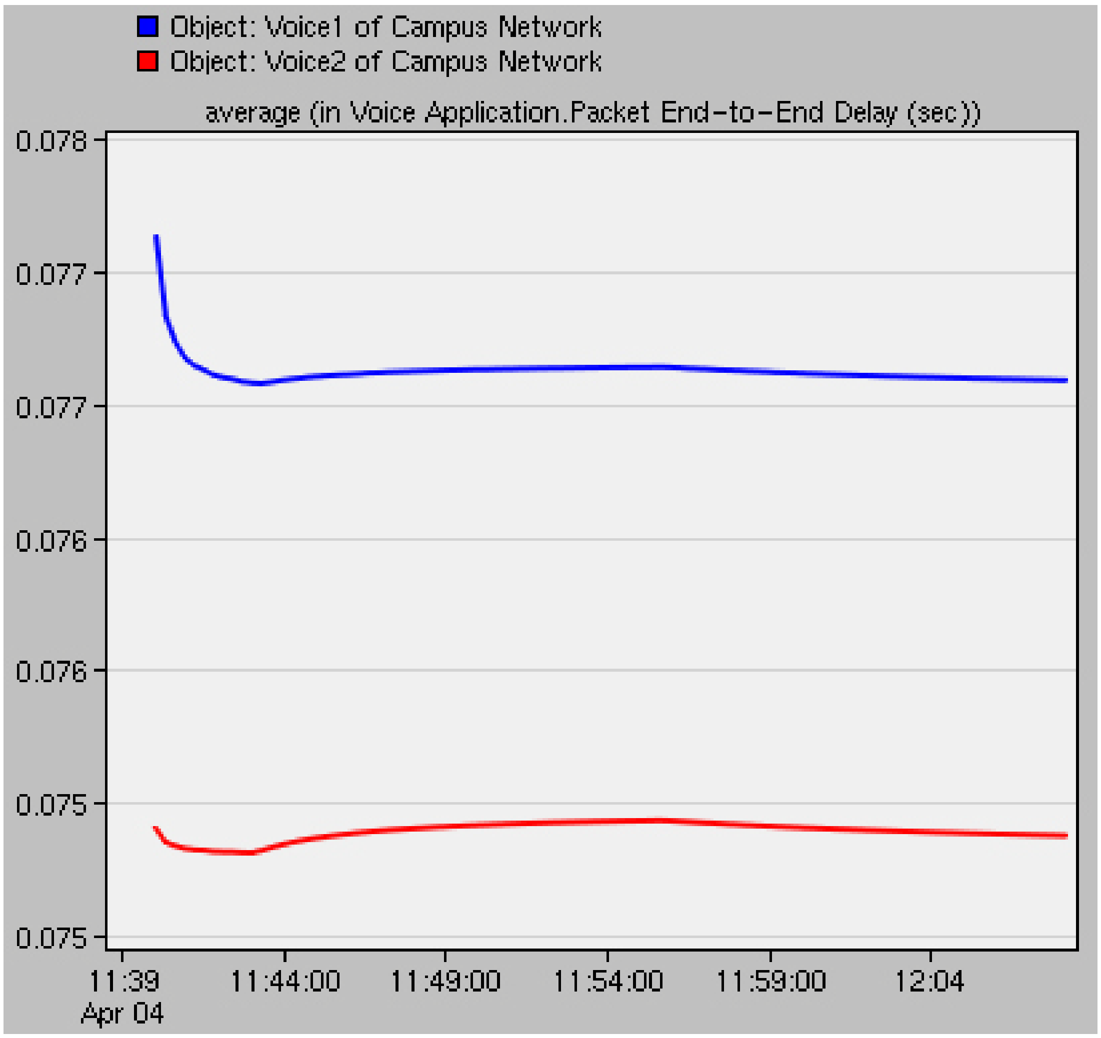

Figure 3.1.3: End-to-end delay for each voice node in voice only configuration

Figure 3.1.4 shows the jitter for each of the voice nodes. As shown in the figure, once the simulation reaches a steady state, the jitter for both nodes is approximately 0, which meets the standards for ideal quality.

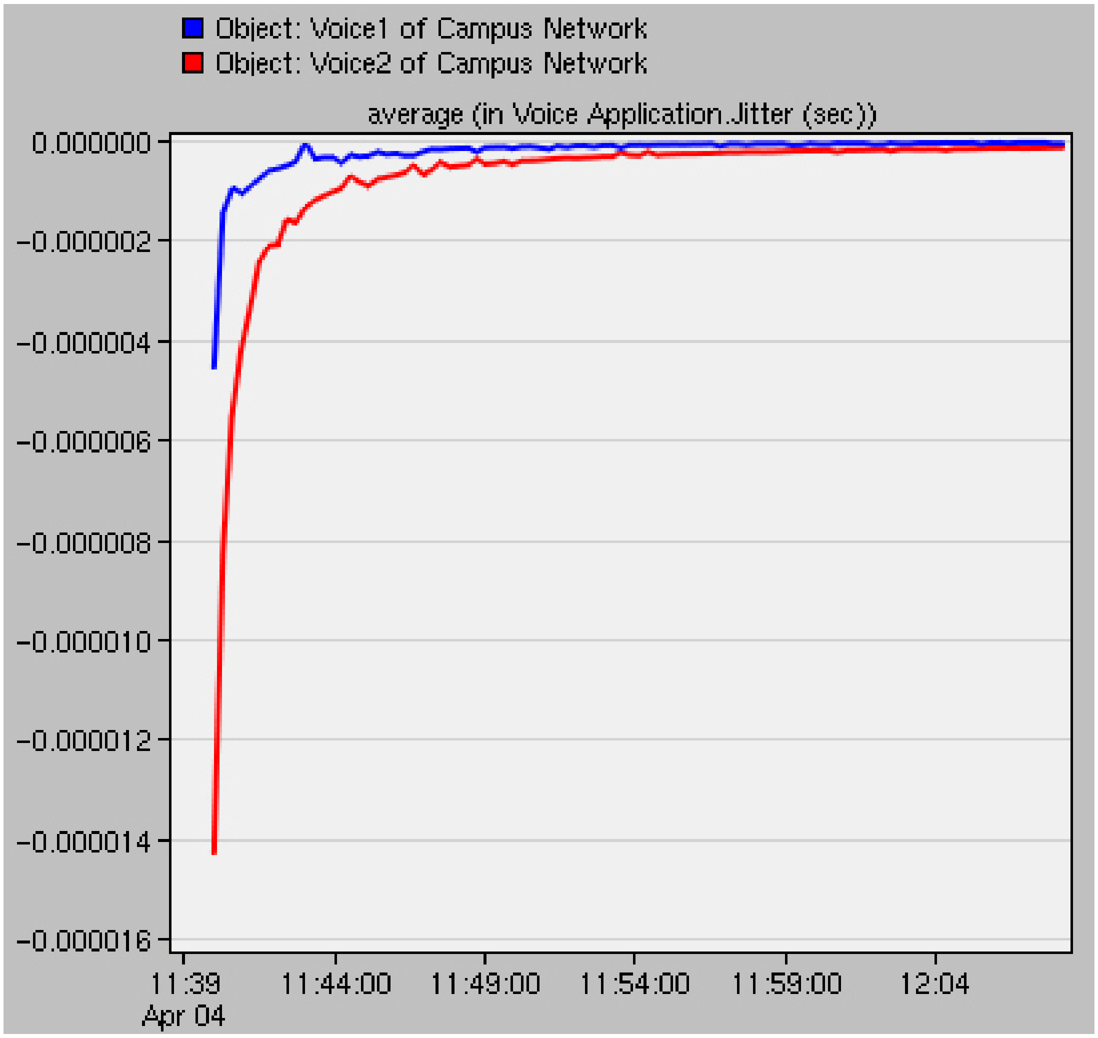

Figure 3.1.4: Jitter of each voice node in the voice only configuration

Figure 3.1.5 shows the mean opinion score for each of the voice nodes. The Voice2 node has a steady state MOS of about 3.599 whereas the Voice1 node has a steady state MOS of about 3.588. This result makes sense, as Figure 3.1.3 shows the Voice1 node having a larger end-to-end delay than the Voice2 node, and Figure 3.1.4 shows both nodes having the same jitter. We would therefore expect the Voice2 node to have a slightly higher mean opinion score, which is exactly what we see from this simulation.

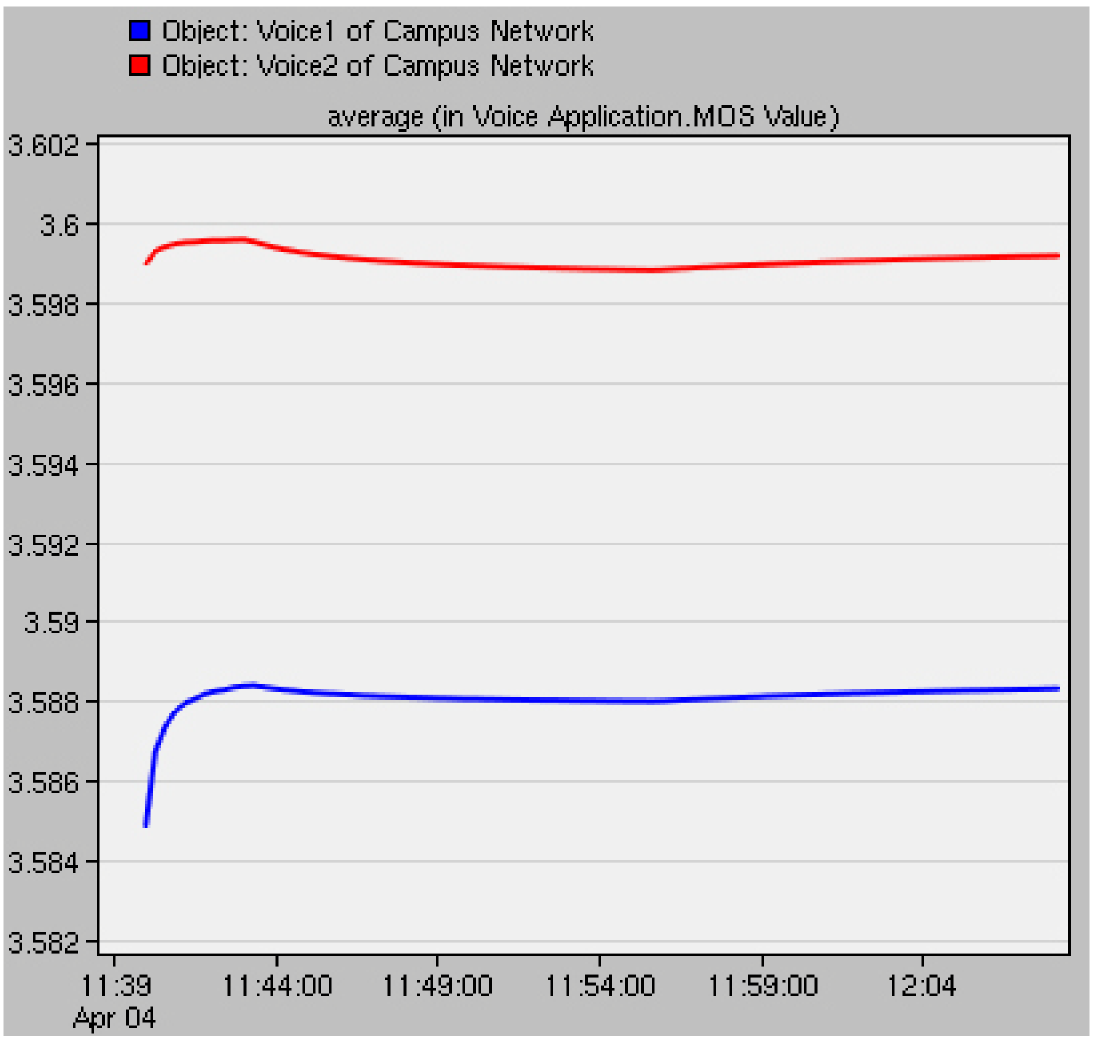

Figure 3.1.5: MOS of each voice node in the voice only configuration

3.2 Voice and Light FTP Configuration

Figure 3.2.1 shows the FTP traffic over the LTE network. Each peak in the figure corresponds to one file transfer of 100 KB. Understanding the traffic pattern in this figure is essential for understanding some of the remaining results in this section.

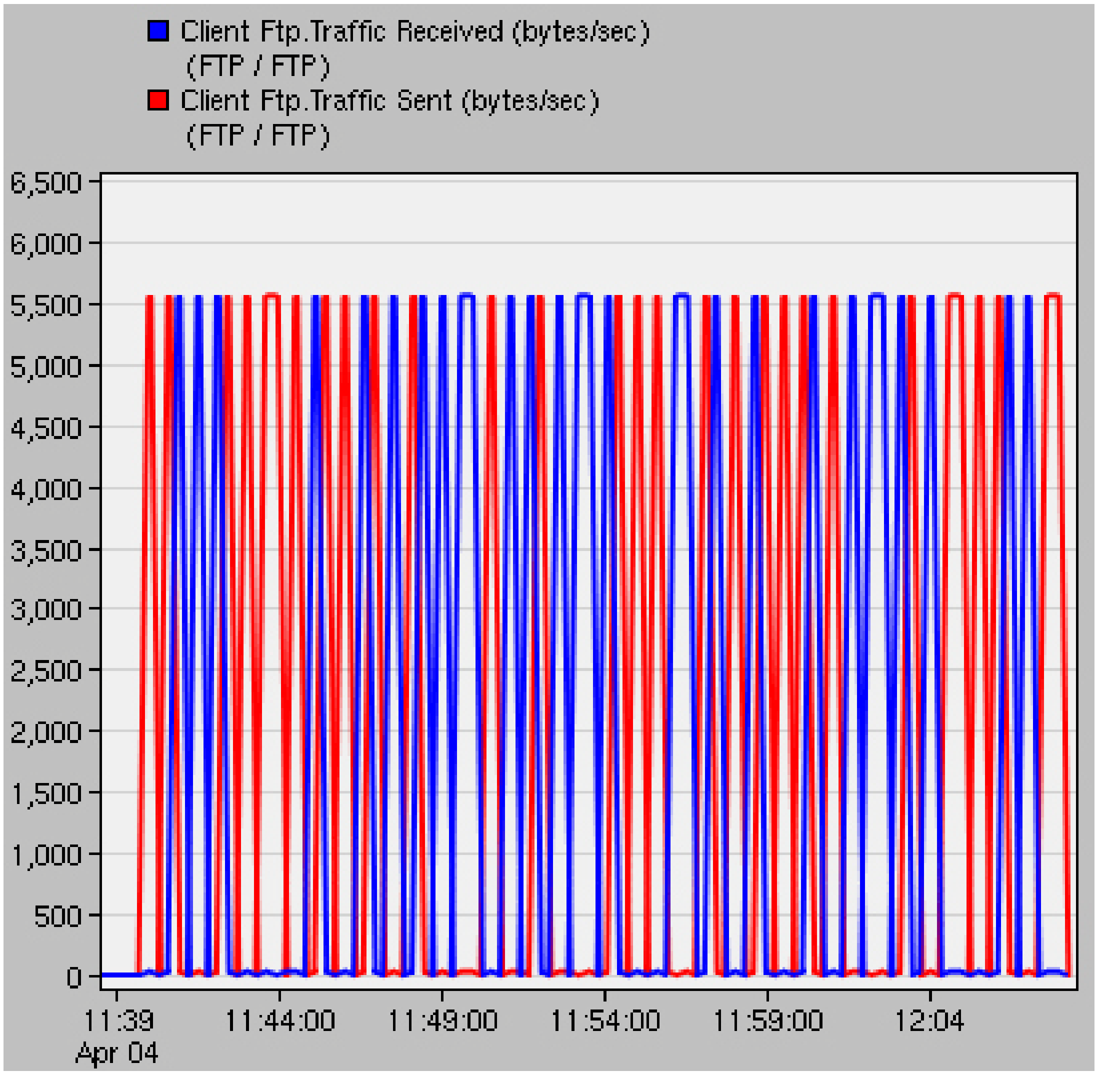

Figure 3.2.1: Sent and received FTP traffic for the voice and light FTP configuration

Figure 3.2.2 shows the total LTE network load at the eNodeB for the voice and light FTP configuration overlaid over top of the network load for the voice only configuration. We can see the baseline load is the same as from the voice only configuration, with additional load caused from the FTP file transfers.

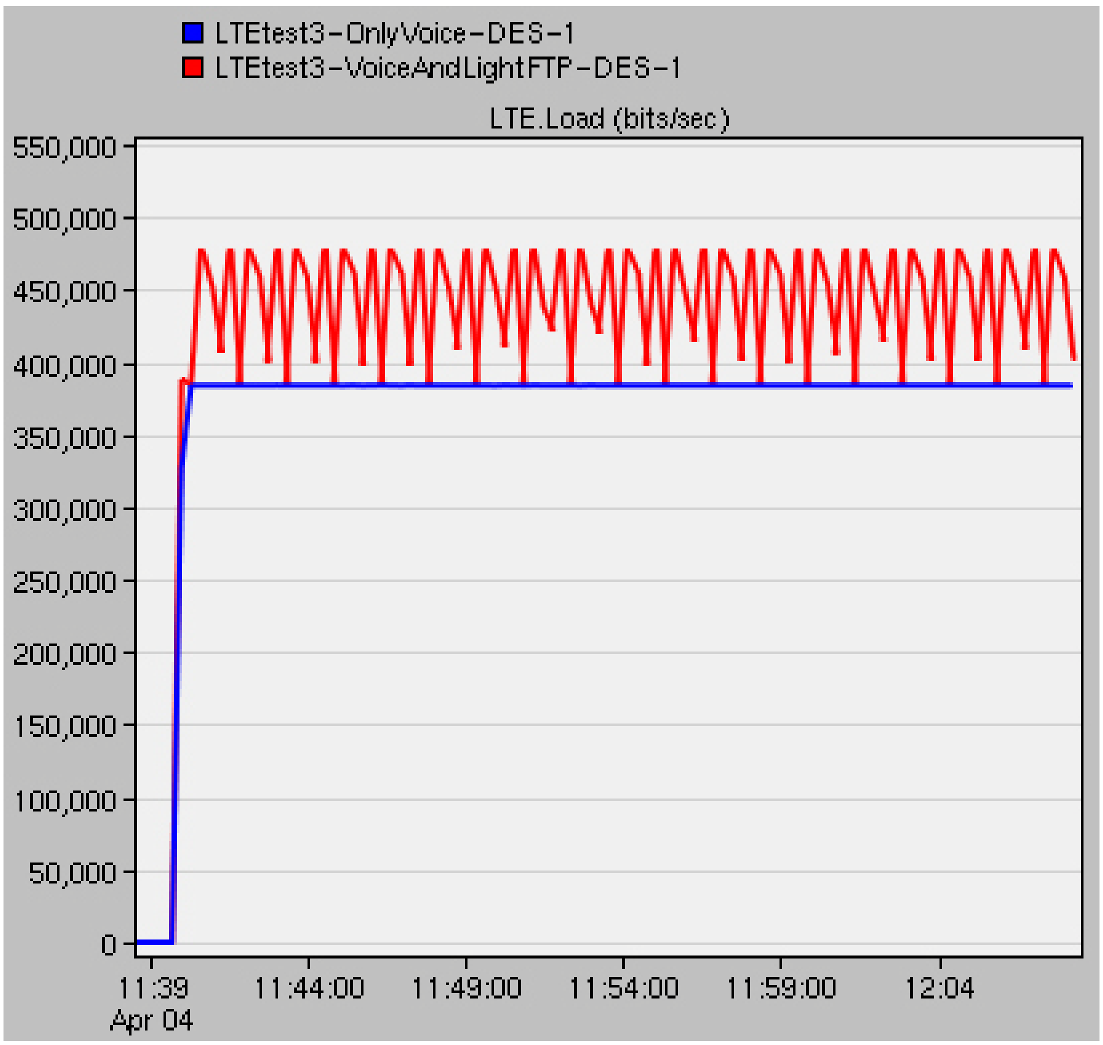

Figure 3.2.2: LTE network load for the voice only and voice and light FTP configurations

Figure 3.2.3 shows the sent and received voice traffic between the two voice nodes. From this figure we can see that the sent and received traffic are no longer equal. The sent traffic is at 15,536 bytes/sec, and the received traffic is at 15,162 bytes/sec. This corresponds to a packet loss of 2.4%, which is average quality. We would expect to see some packet loss as network congestion increases.

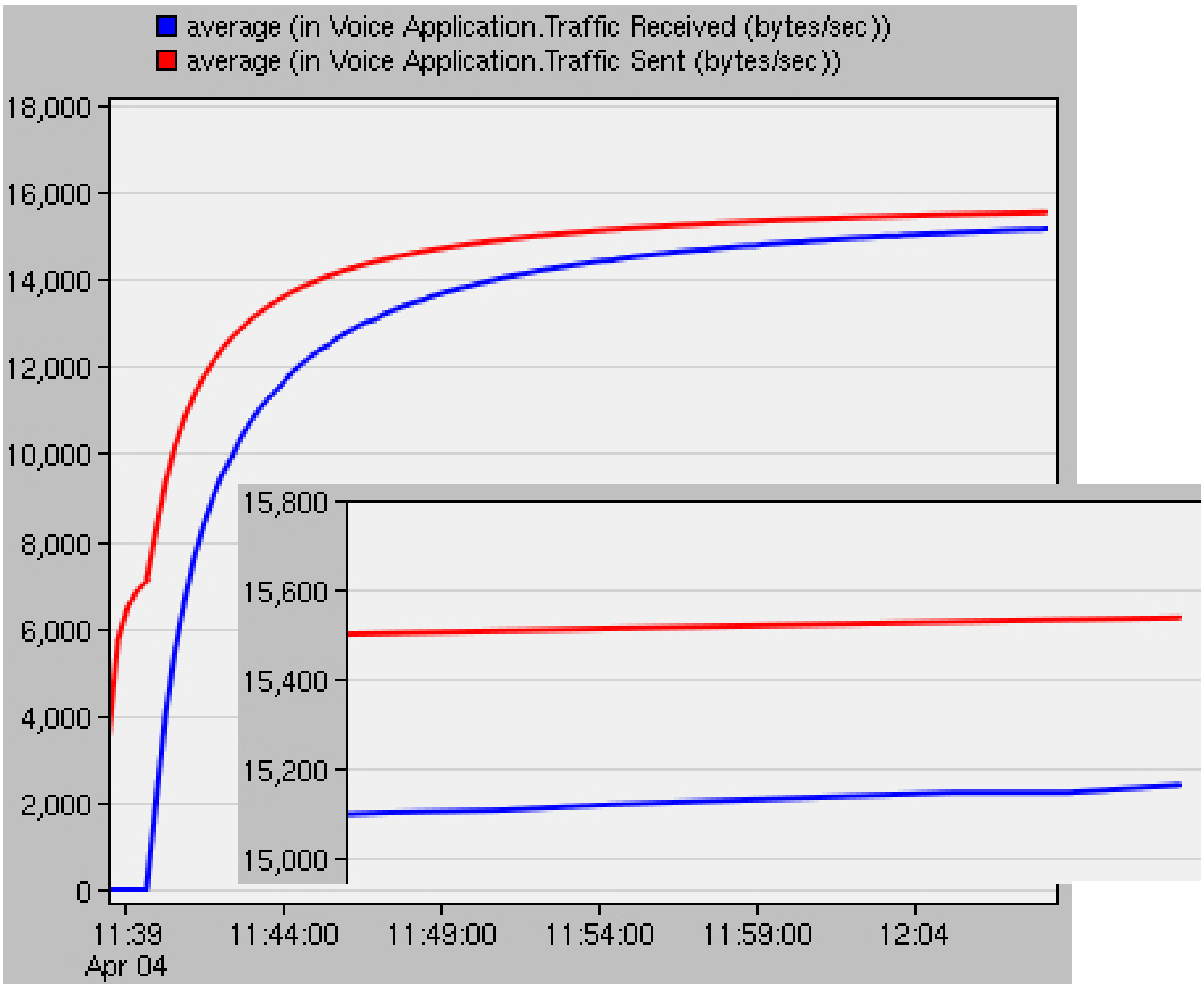

Figure 3.2.3: Sent and received voice traffic for the voice and light FTP configuration

Figure 3.2.4 compares the end-to-end delay for the voice only and voice and light FTP configurations for each of the two voice nodes. We can see that the additional FTP traffic has caused an increase in the end-to-end delay, from about 77 ms to 165 ms at steady state. We would expect such a delay increase due to the increased network congestion. This longer delay causes the voice application to no longer meet the ITU average quality standard of 150 ms.

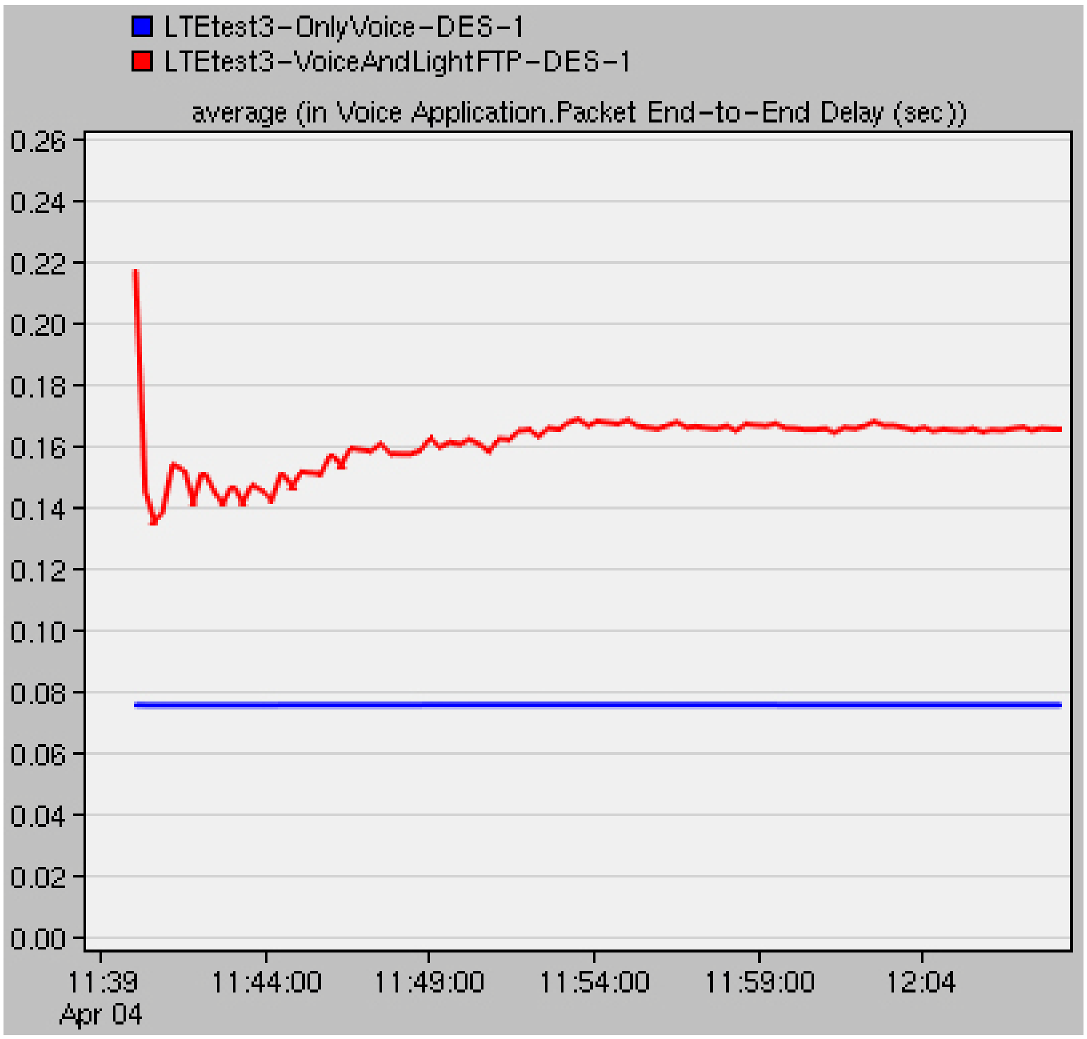

Figure 3.2.4: Comparison of end-to-end delay for voice only and voice and light FTP configurations

Figure 3.2.5 compares the jitter for the voice only and voice and light FTP configurations. We can see the jitter for the light FTP configuration has some spikes which correspond to the FTP file transfers, ending with a steady state value of 0.01 ms. This jitter is quite negligible, and still meets the ideal quality standards.

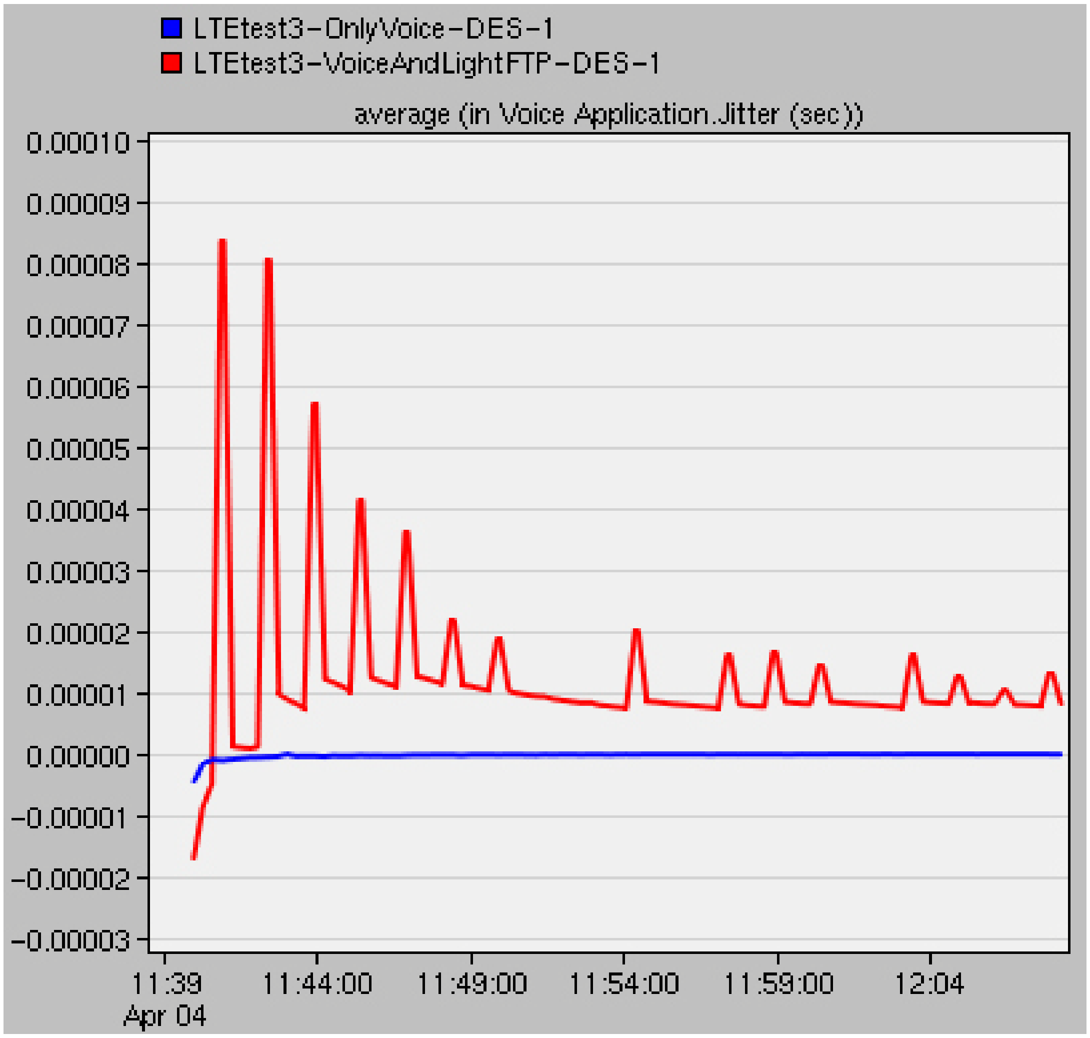

Figure 3.2.5: Comparison of jitter for voice only and voice and light FTP configurations

Figure 3.2.6 compares the mean opinion scores for the voice only and voice and light FTP configurations. We can see that the MOS has fallen to a steady state value of about 3.25. This result makes sense as we would expect a lower score due to the increased delay and packet loss.

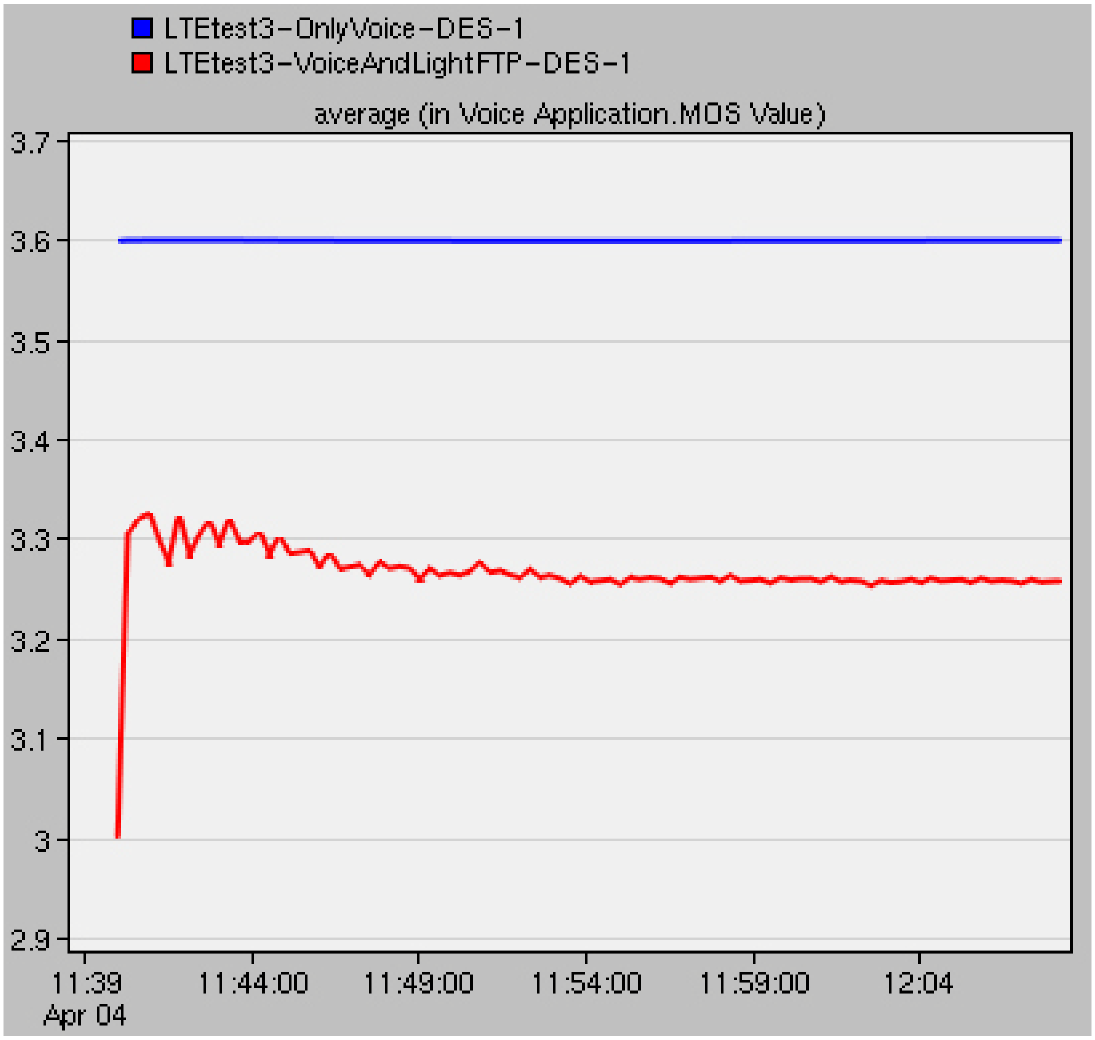

Figure 3.2.6: Comparison of MOS for voice only and voice and light FTP configurations

3.3 Voice and Heavy FTP Configuration

Figure 3.3.1 shows the sent and received voice traffic between the two voice nodes. From this figure we can see that there is an even greater difference between the sent and received traffic as was the case in the light FTP configuration. The sent traffic is still the same at 15,536 bytes/sec, but the received traffic is now down to 15,077 bytes/sec. This corresponds to a packet loss of 2.95%, which still meets average quality standards. These results make sense, as we have slightly more packet loss than the light FTP configuration, but we also have increased network congestion.

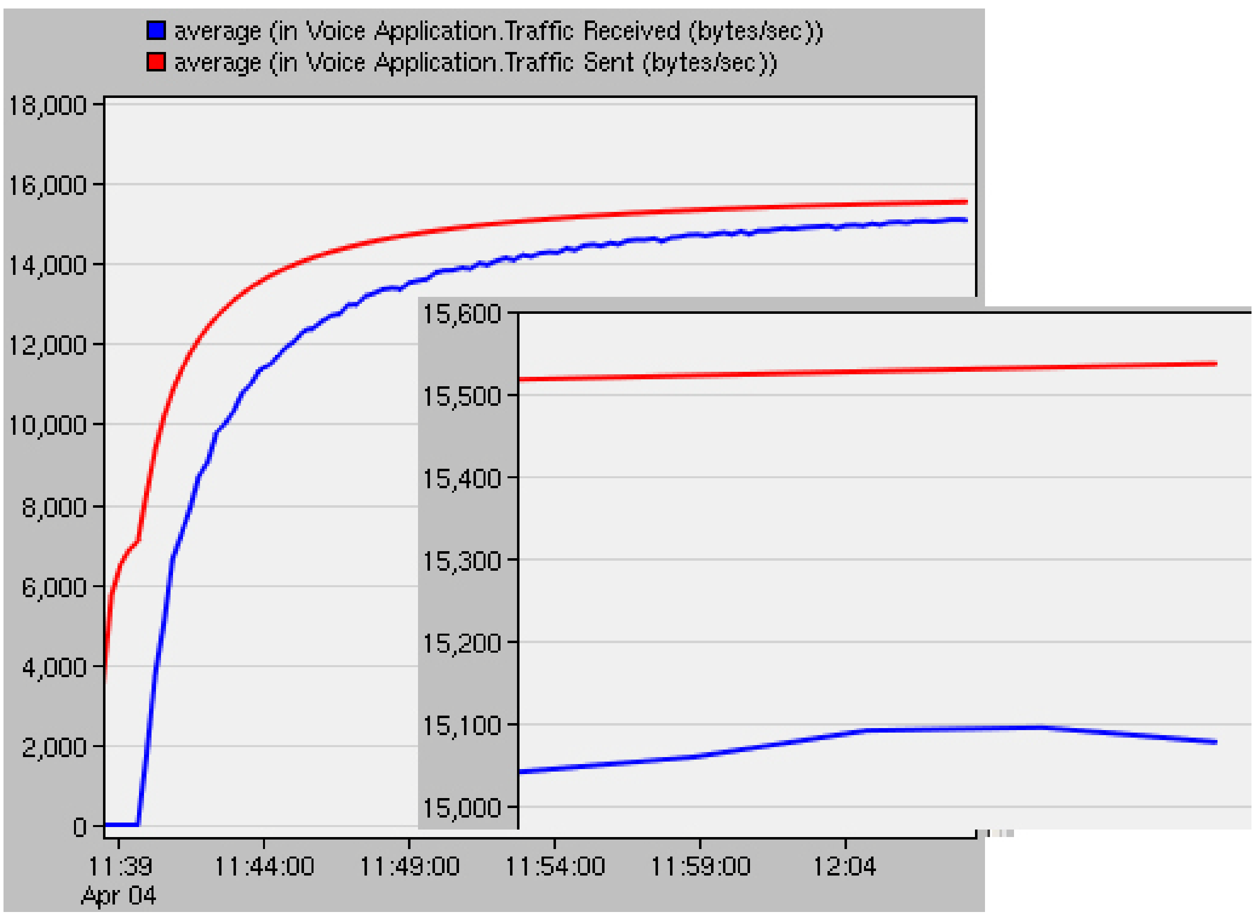

Figure 3.3.1: Sent and received voice traffic for the voice and heavy FTP configuration

Figure 3.3.2 compares the end-to-end delay for the voice only and voice and heavy FTP configurations for each of the two voice nodes. We can see that the increased FTP traffic has caused an even larger increase in the end-to-end delay, with a steady state value of about 2.5 seconds. We would again expect to see a delay increase due to the increased network congestion. The FTP file size was increased five-fold to 500 KB, but the end-to-end delay increased from 165 ms to 2500 ms, which is about a 15-fold increase. This suggests that there may be an exponential relationship between congestion and end-to-end delay.

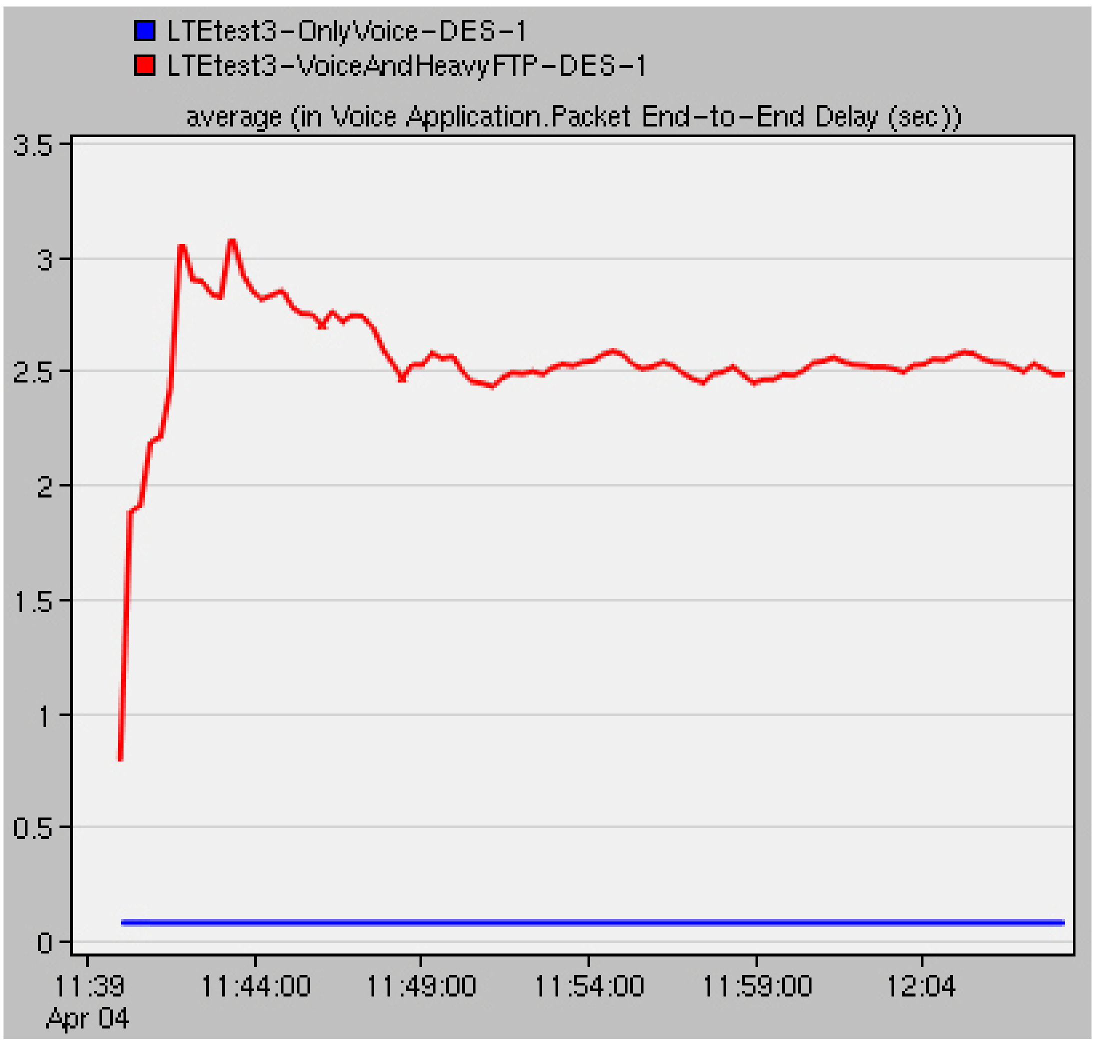

Figure 3.3.2: Comparison of end-to-end delay for voice only and voice and heavy FTP configurations

Figure 3.3.3 compares the jitter for the voice only and voice and heavy FTP configurations. The steady state jitter value has increased from the light FTP configuration, with a new value of about 0.3 ms. This jitter is still negligible, meeting ideal quality standards. These results suggest that network congestion does not have a significant effect on jitter.

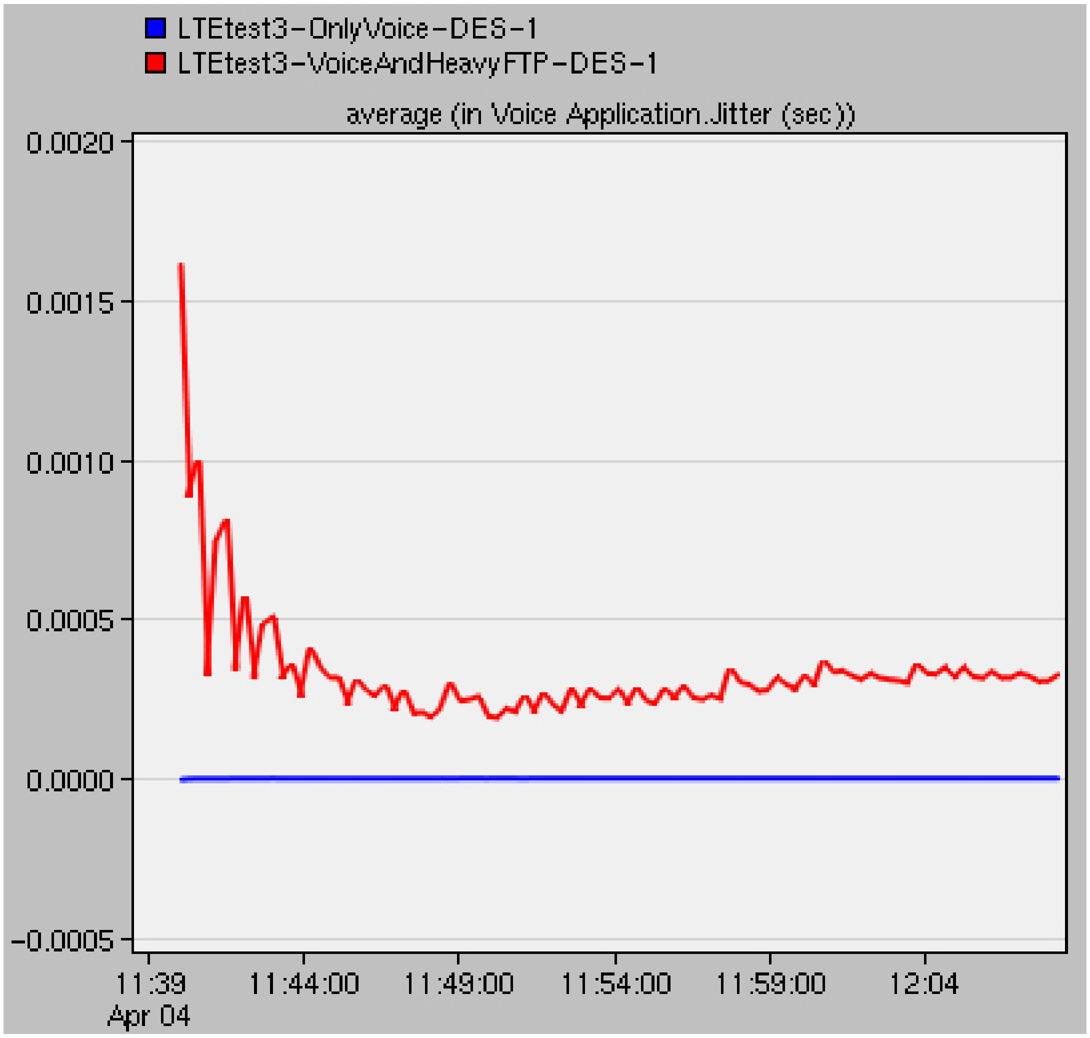

Figure 3.3.3: Comparison of jitter for voice only and voice and heavy FTP configurations

Figure 3.3.4 compares the mean opinion scores for the voice only and voice and heavy FTP configurations. We can see that the MOS has decreased even more than before to a new steady state value of just below 1.5. This result makes sense as there was a very large increase in end-to-end delay, so we would expect a large decrease in the MOS.

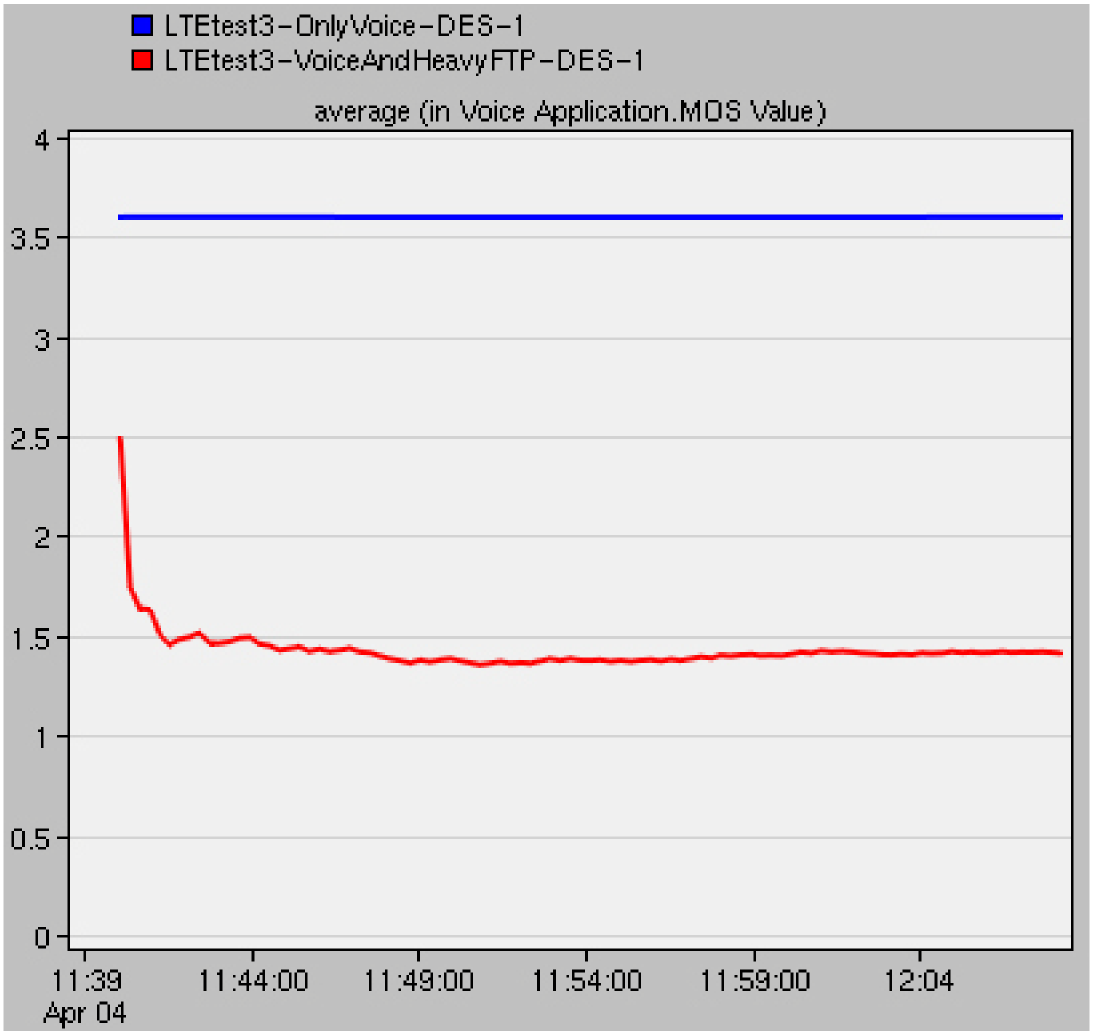

Figure 3.3.4: Comparison of MOS for voice only and voice and heavy FTP configurations

3.4 Voice and Massive FTP Configuration

Figure 3.4.1 shows the sent and received voice traffic between the two voice nodes. From this figure we can see that there is now a large difference between the sent and received traffic. The sent traffic is still 15,536 bytes/sec, but the received traffic is now significantly down to 10,579 bytes/sec. This corresponds to a packet loss of 31.9%, which is way below average quality standards of 5%. This result continues the trend of increased packet loss with increased network congestion.

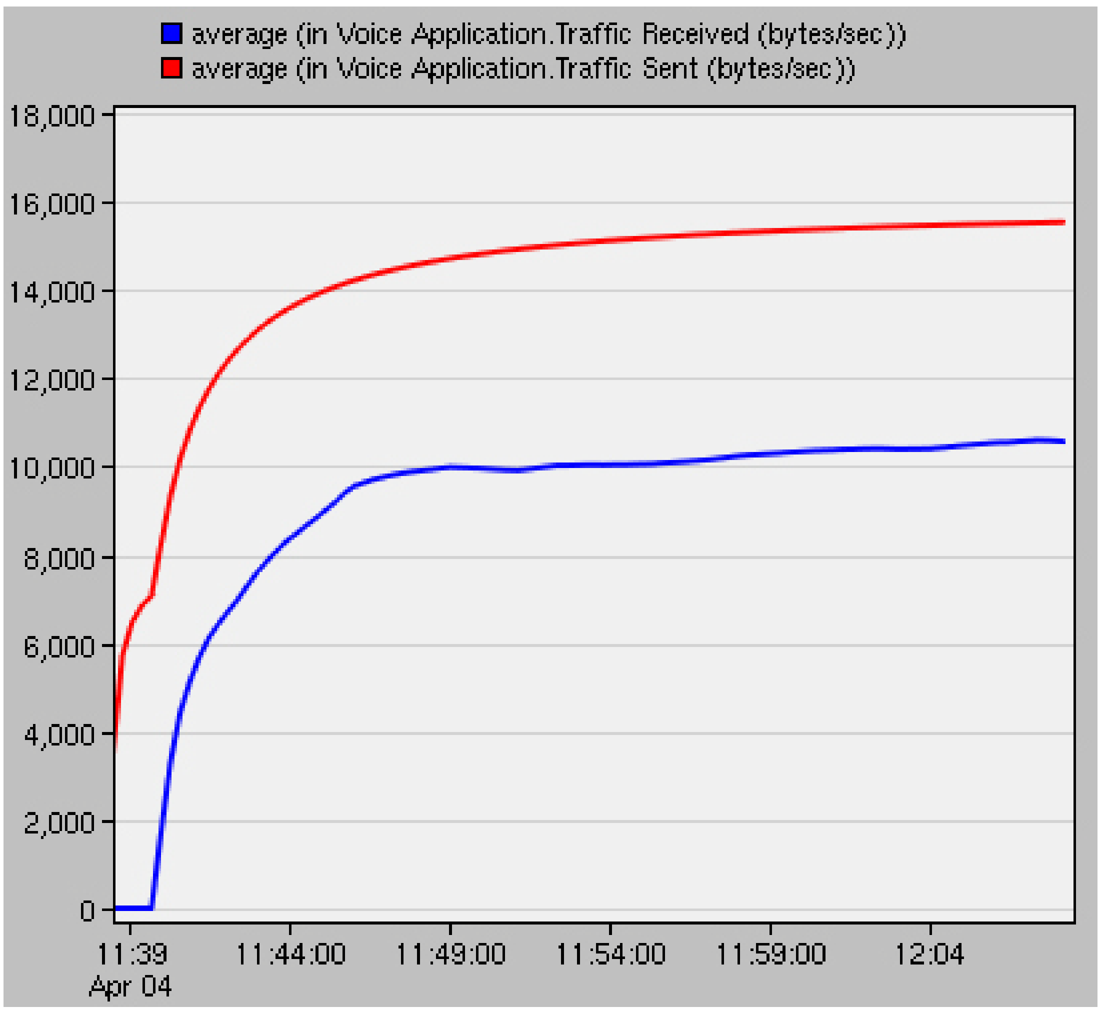

Figure 3.4.1: Sent and received voice traffic for the voice and massive FTP configuration

Figure 3.4.2 compares the end-to-end delay for the voice only and voice and massive FTP configurations for each of the two voice nodes. We can see that the increased FTP traffic has again caused an even larger increase in the end-to-end delay, with a steady state value of just over 10 seconds. This result continues the trend of exponentially increasing end-to-end delay with increasing network congestion.

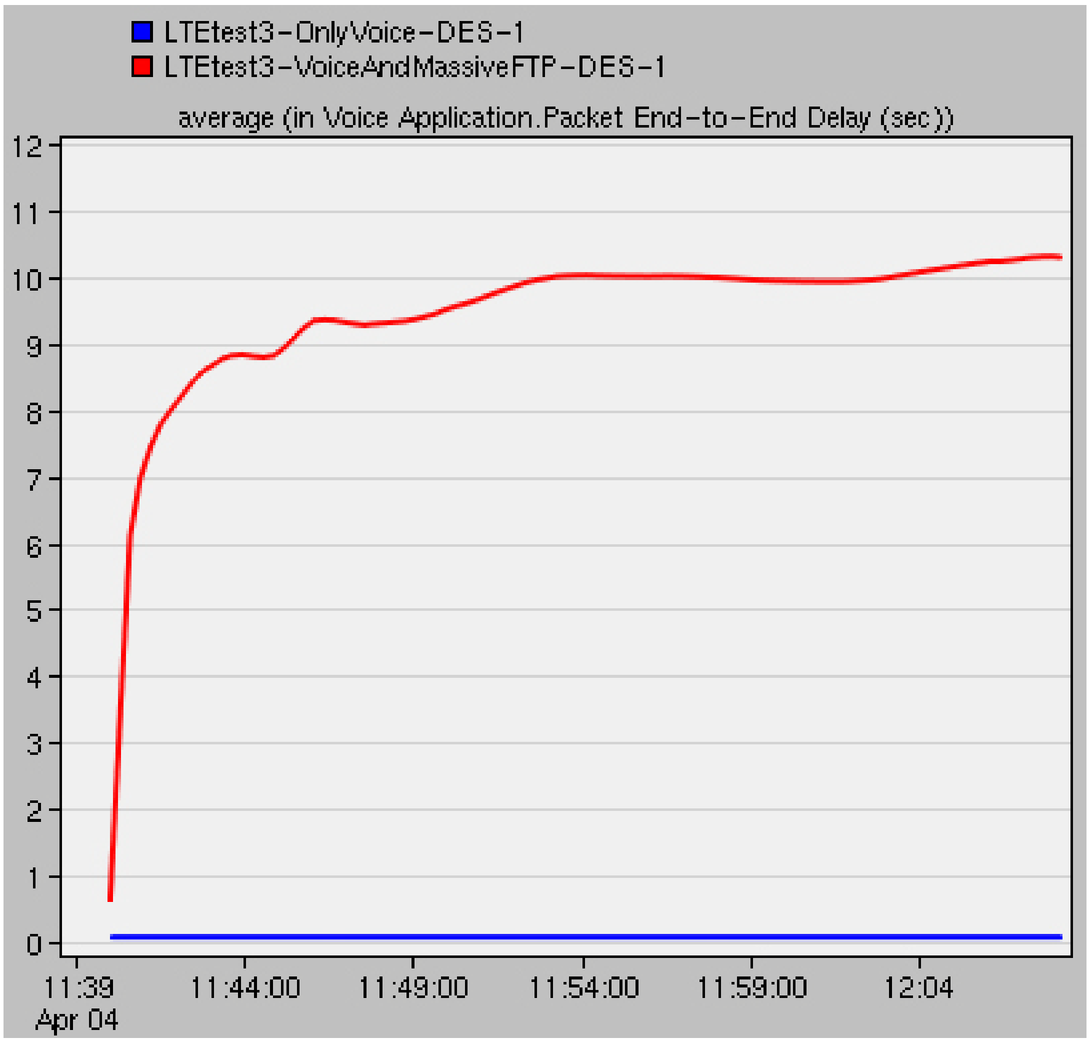

Figure 3.4.2: Comparison of end-to-end delay for voice only and voice and massive FTP configurations

Figure 3.4.3 compares the jitter for the voice only and voice and massive FTP configurations. Even with an incredibly large amount of network congestion, the steady state value of the jitter remains negligible. From this result, we conclude that network congestion has no significant effect on jitter.

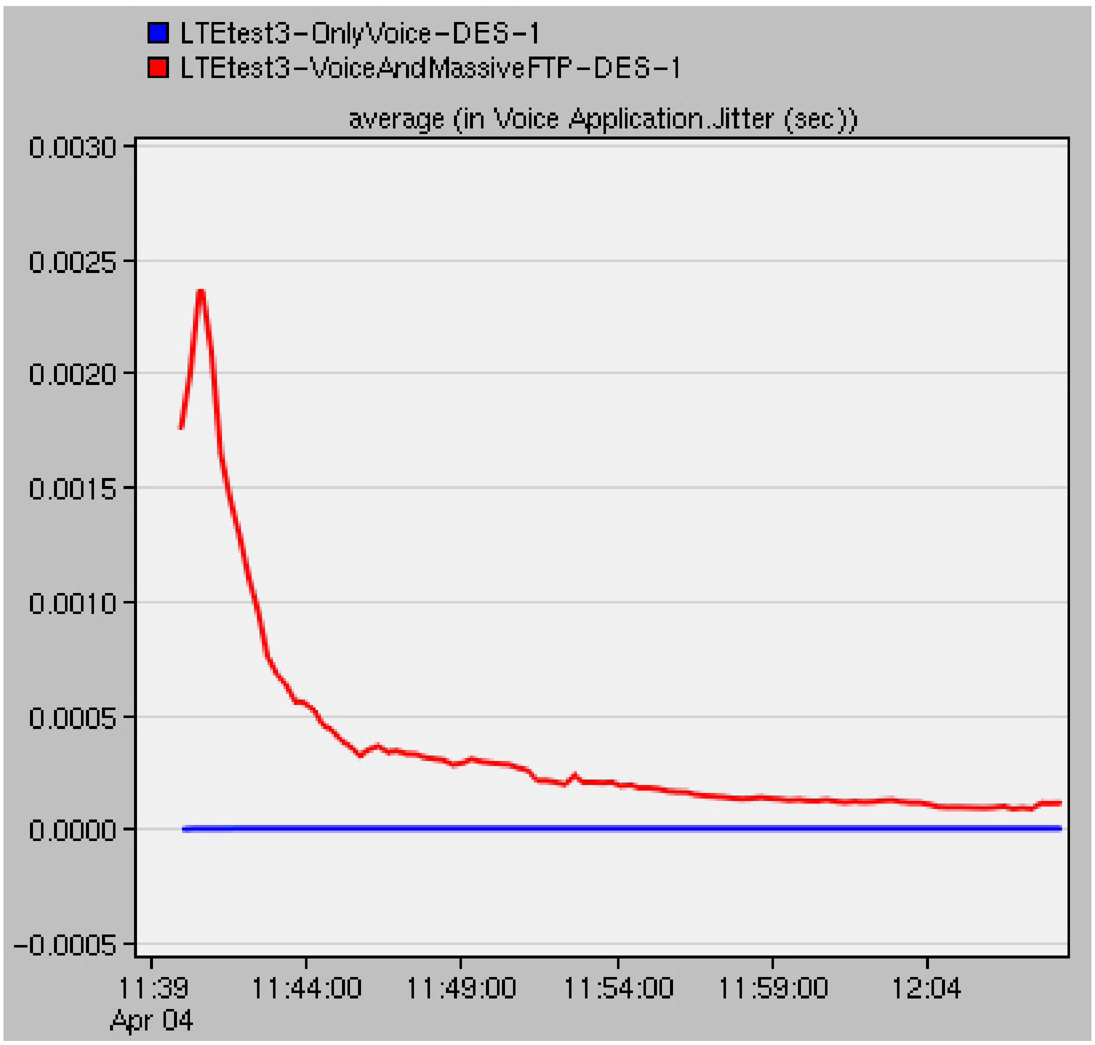

Figure 3.4.3: Comparison of jitter for voice only and voice and massive FTP configurations

Figure 3.4.4 compares the mean opinion scores for the voice only and voice and massive FTP configurations. Due to the incredibly long end-to-end delay of over 10 seconds, the MOS score for the voice and massive FTP configuration is the lowest possible value of 1, indicating bad quality.

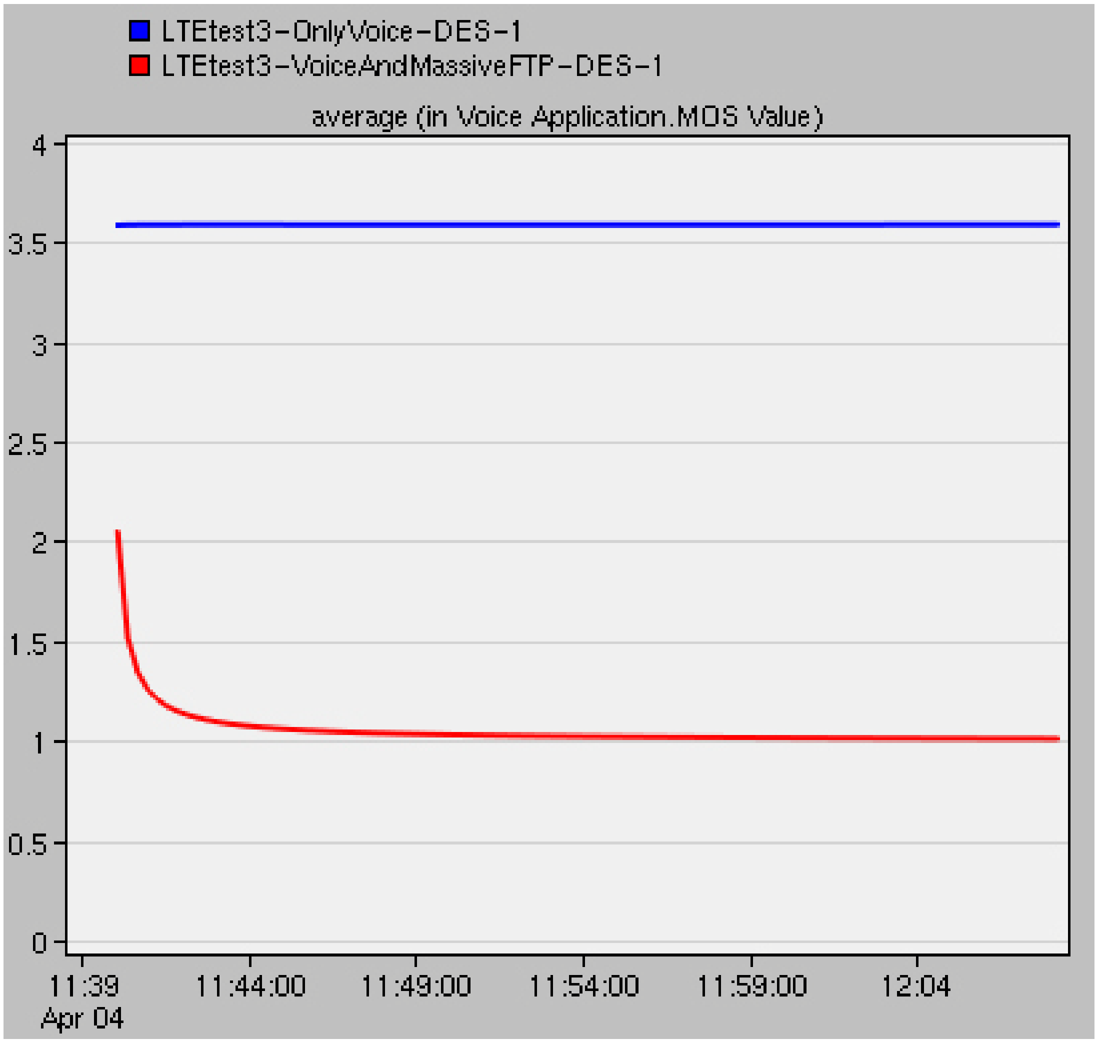

Figure 3.4.4: Comparison of MOS for voice only and voice and massive FTP configurations

3.5 Challenges

The main challenge in this project was getting the LTE network to work properly. OPNET does provide some documentation with regards to LTE networks, but it is largely incomplete and does not provide much detail with regards to setting up an LTE network. There were some example LTE networks provided with OPNET, but we could not get these networks to run simulations without errors, despite receiving help.

Originally, we wanted to investigate the effect of moving nodes on voice call performance, particularly with regards to how moving at progressively increased speeds would affect performance. However, we could not figure out how to make this work properly with an LTE network. We correctly set our mobile nodes to travel along a trajectory, but the results were always the same regardless of the trajectory used or no trajectory at all.

3.6 Future Work

Future work would involve investigating how voice node movement impacts performance, as well as how switching between base stations while moving would affect performance.

4. Conclusion

From our simulations, we found that even with a high amount of network congestion, packet loss during voice calls was still within average quality standards. After imposing an extreme amount of congestion, however, the packet loss finally became significant, with a loss rate of 31.9%.

We found that end-to-end delay increases exponentially with increasing network congestion. A five-fold increase in congestion led to a 15-fold increase in delay, and a ten-fold increase in congestion led to a 60-fold increase in delay.

Surprisingly, based on our examination of jitter, we found that network congestion has no significant effect on jitter. Even under extreme network loads, the jitter was still negligible.

The mean opinion score basically just takes the other three parameters into account; as they increase, the MOS decrease. Since end-to-end delay increases exponentially with congestion, the delay tends to dominate the results of the MOS score. We found that under zero to low amounts of congestion, the voice application still received fair to good scores.

5. References

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