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Voice Call Quality Using 802.11e On A Wireless Mesh Network

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***Abstract*—Wireless Local Area Networks (WLANs) provide an affordable solution for last mile network access. They also allow for extension of a network by configuring a Wireless Mesh Network (WMN) where it may otherwise be physically infeasible or cost prohibitive to do so. With the increasing use of real-time applications such as video conferencing and Voice over IP (VoIP), networks are stressed to guarantee QoS requirements for these applications. Examples of key requirements include bounded delay and packet loss ratios. Addressing this issue in WLANs, the IEEE 802.11e amendment was proposed to provide a QoS mechanism. However, the performance of 802.11e in meshed environments is yet to be studied.**

In this work, we study VoIP call quality in a meshed environment with provisions for QoS. We study the call quality and throughput of background traffic in an experimental WMN testbed in order to test how well the IEEE 802.11e QoS provisions support voice calls. Call quality is tested in different configurations and scenarios. We study the effect of the number of wireless hops on VoIP call quality. In addition, we investigate the number of VoIP calls that can be supported simultaneously for different numbers of wireless hops. We also study how fairly the network treats different calls in different configurations. Then, we look at how much effective bandwidth a VoIP call uses on the network. Finally, we examine the VoIP call quality of different calls when calls have different QoS parameters and study the effect that a busy central node has on traffic passing through it. We provide suggestions to improve call quality on a WMN and hint at possible future work.

I. INTRODUCTION

Wireless networks have revolutionized the telecom industry and have allowed for customer mobility on many different networks. Cellular, WLANs, and WiMAX are just a few types of wireless networks in use today. With the convergence of different communication methods to IP based networks, it is understood that different kinds of media have different requirements. That is, services such as E-mail, file transfer, and web browsing have different delay tolerances than, say, VoIP. While transfer speed is important to E-mail and file transfer services, both can tolerate delays if needed. Web browsing is becoming more interactive and thus cannot tolerate as much delay as E-mail or file transfer services, but can still tolerate some delay without difficulty. Video conferencing is an application with requirements that fit somewhere in between VoIP and file transfer. That is, it can adapt to variable network conditions by adjusting the video quality upwards or downwards while keeping the voice quality constant. While this is not ideal, it reduces the amount of bandwidth a video conference uses. When comparing VoIP and file transfers, VoIP cannot tolerate large amounts of delay or loss, and as such, is one of the most demanding services on an IP network.

Needless to say, VoIP is a useful service and is becoming widespread. It is projected that the percentage of VoIP lines being installed in the corporate sector will reach 44% by the end of 2008 [1]. Motivations for using VoIP include the cost savings and ease of deployment since the existing data network infrastructure can be used. Alternatives to VoIP in an office environment are costly proprietary PBX systems with exclusive wiring for each phone.

Due to the growing use of both WLANs and VoIP, VoIP over wireless networks is receiving wide interest. Using wireless VoIP can provide a user with limited mobility and the convenience of using a portable phone, while not being tied to a computer or desk phone. Some cellular telephones also have the ability to connect to WLANs, therefore providing a user with two network types in one device. This allows for VoIP use when in range of a WLAN, and cellular network use otherwise.

Providing wireless VoIP users with phone services over a wide area requires good WLAN coverage. This coverage can be provided and extended by use of WMNs. WMNs, unlike traditional wireless networks where each access point is connected to the wired network, allow all of the access points to be connected to the wired network through each other. A WMN is similar to the wired network in that routers forward packets from one node to the next. In a WMN, however, the packets are forwarded from access point to access point through the wireless medium until the destination is reached. The destination can be on another network, in which case the data travels over the mesh network to the wired part and beyond. It can also be on the WMN, in which case the data travels only over the WMN. Regardless, data must traverse the mesh network and the network must be able to handle a variety of services successfully.

Providing real-time services such as voice over a wireless network is difficult given that no QoS mechanism was provided in the original IEEE 802.11 [2] standard. The 802.11e amendment to the IEEE 802.11 [2] standard adds a QoS mechanism to 802.11 WLANs. 802.11e has been previously evaluated in a non-meshed environment, but has not been

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evaluated on a WMN testbed.

In this paper, we focus on the functionality of 802.11e in providing desired QoS for voice calls on a WMN. We will evaluate call quality in a variety of configurations and scenarios on a WMN testbed. The paper provides these main contributions:

- Identifying performance metrics for VoIP calls that will allows the comparison of call-to-call quality.
- Creating test scenarios to evaluate VoIP over a WMN, as well as scenarios that test different characteristics, such as: call quality per-hop, maximum number of calls supported, etc.
- Establish guidelines for configuring parameters of 802.11 WMNs to help increase call quality.

In order to measure voice quality, we will be recording data to calculate end-to-end delay, jitter and packet loss. We will use this data to calculate a MOS value that represents perceived voice quality. This will be calculated through the use of the ITU-T E-model [3], a method of objectively calculating perceived voice quality for use in network planning and evaluation. The E-model calculates perceived voice call quality and provides a way to calculate a MOS.

The rest of the paper is organized as follows. In Section 2 we survey some of the similar work. In Section 3, we provide some implementation details of our WMN testbed. In Section 4, we detail the results of the different test scenarios and summarize the results. In Section 5, we suggest ways of improving the performance of WMNs with respect to voice call quality. We also suggest future research directions in addressing voice call quality over WMNs.

II. Literature Survey

Performance of voice in 802.11 networks was studied in several papers. In [4], voice performance on a single hop wireless network was studied, however, in 2003, the 802.11e standard had not yet been ratified, so QoS could not be studied. In [5], throughput of UDP traffic and voice performance was studied. In this case, testing was also done on a single hop wireless network. In [6], experimental tuning of 802.11e parameters was conducted, but only on a single hop network. In [7], a simulation was performed to study the dynamic tuning of the CWmax parameter. In [8], performance in 802.11 WMNs was studied with respect to voice, but no QoS implementation was used.

In [4], a testbed network was configured to test 802.11 single hop wireless network performance. Anjum, Dutta, et al measured the performance and capacity of the standard 802.11 DCF mode to deliver high quality voice service with and without background traffic.

In [5], performance of UDP traffic and VoIP was studied over a single hop 802.11b network, from the perspective of the number of simultaneous calls that a single access point can handle. Garg and Kappes discuss that due to the small payload size of the voice packets, the 802.11 network becomes inefficient and cannot support many calls. The study found that only six calls could be supported on an access point when the G.711 CODEC was used with a 10 ms sampling time. This equates to 100 packets per second. We expect to be able to handle more simultaneous calls with the more normal 20 ms sampling time.

In [6], experimental tuning of the CWmin parameter is tested in a single hop configuration with respect to voice calls. Davis and Narbutt test numerous CWmin parameter settings and determine that adjusting the CWmin of the background traffic has a positive effect on the voice call quality.

In [7], dynamic tuning of the CWmax parameter was carried out in a simulation. Hanley and Murphy showed that an adaptive scheme in a simulation environment could increase performance on a single hop network. Results show that the adaptive scheme that dynamically adjusts the CWmax parameter can indeed increase performance and can consistently provide a higher VoIP call capacity.

In [8], an experimental mesh network testbed was configured to test the performance of VoIP. Niculescu et al tested several configurations, including a standard single radio setup and multiple radio setup. They also tested packet aggregation. The standard single radio configuration tested the number of calls that can be supported at one time.

The above papers do not all cover mesh networks, nor do they cover the three combined features that we cover in our research. These are, WMNs, VoIP, and QoS. We are testing how well the current QoS mechanism in 802.11 works with respect to VoIP, background traffic, and WMNs. These, to our knowledge, have not been previously combined. The experimental value of the other works applies to specific configurations only. None of the above papers study WMNs, VoIP, and 802.11e QoS simultaneously, so it is difficult to use their experimental results to compare with ours.

In addition to the above mentioned papers, there are several others that deal with modification to the 802.11 standard, and as a result, are not implemented on any commercially available 802.11 routers.

Two schemes discussed in [9] deal with packet multiplexing and header compression. Packet multiplexing is the process of analyzing packets at each node and merging packets together temporarily. Only packets that are travelling in the same direction and on the same link can be multiplexed. Header compression is a process that takes place at each node and compresses the IP, UDP and RTP header down in size, therefore increasing the wireless network payload efficiency of the transmission.

The proposed schemes in [9] demonstrate through simulation that multiplexing can improve the VoIP capacity, but must be implemented at the access point and requires significantly more processing power than most access points currently have. Header compression also provides a slight performance increase, but also requires more processing power. Both schemes, due to the extra processing, increase delay, but are certainly an option worth exploring on high end access points. In [9], it is stated that 802.11e can provide a solution to the QoS problem, but tuning the parameters and evaluating the results is still an outstanding issue that needs

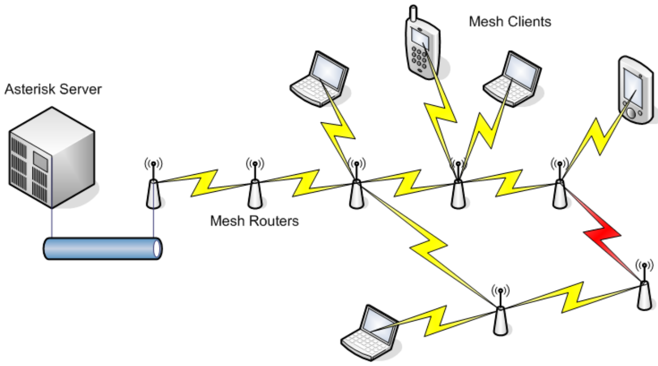

Fig. 1. The generic mesh network configuration

to be investigated.

Gauging call quality from WLAN resource usage is discussed in [10]. Davis and Narbutt [10] shared that establishing a new call on a wireless network near capacity when calls are already active can dramatically decrease call quality. They proposed an application that passively monitors wireless local area network resource usage and can provide the information to a call admission control system. The purpose of the passive monitoring approach proposed is to allow a call to be established if there are enough available resources, while not allowing a call if the available resources are not sufficient.

III. Implementation

In our experimental WMN testbed, the mesh routers used are single radio 802.11 routers. With the DD-WRT [11] software, these routers are versatile in what can be done with them because of their Linux [12] software base. They can be configured as clients, access points, and network bridges, and have many other configuration options. The implementation of 802.11e and the ability to control many parameters of the routers allows us to test various experimental scenarios.

The generic configuration of our WMN testbed is a forced linear configuration, with each router forwarding packets on to the next in a linear pattern. This allows for a defined, specific test configuration that will not change during an experiment. This contrasts to a WMN with dynamic routing, where the next hop is dynamically chosen based on link quality or other factors. This configuration is typical of traffic in a mesh network, with traffic flowing across the network from a computer to a gateway node. The traffic may traverse only one hop, or it may traverse multiple hops. The number of hops that traffic will traverse depends on how far away the wireless station is from the gateway node and the configuration of the mesh network. The generic mesh network configuration is shown in Fig. 1.

Several 802.11 parameters were configured to specific values in order to obtain consistent results. These options include the transmit rate of the nodes, CTS protection mode, frame burst, transmit power and encryption settings, among others. A discussion of the reasoning for the choices is offered below.

The transmit rate of an 802.11 device is normally controlled automatically based on channel conditions. Preliminary testing showed that automatic rate setting, with a maximum of 54 Mbps, resulted in wildly unpredictable performance and extreme error spikes. A fixed rate setting of 11 Mbps provided a low error rate and consistent performance, while not being too slow to prohibit a voice call at a hop count of five on the mesh network.

The RTS and CTS mechanisms help to improve performance in 802.11 wireless networks. CTS protection mode prevents 802.11b clients from being blocked in an 802.11b/g environment where there are many 802.11g clients competing for access to the medium. To offer this protection to 802.11b clients, overall performance is decreased. Since the experimental testbed does not contain a large number of both 802.11b and 802.11g clients, this option has been disabled on the mesh network routers.

Frame burst is an option which offers increased performance when there are very few clients. The way that it works is by transmitting more packets than normal, therefore tying up the medium for a longer period of time. This makes other stations wait longer, which explains why it only works well if there are a very small number of clients. In a mesh network, where routers are retransmitting packets and there are also clients accessing the network, frame bursting decreases overall performance. As a result, frame bursting has been disabled on the mesh network routers.

The transmit power setting on mesh routers adjusts the amount that the signal should be boosted before being transmitted. On the mesh routers in our testbed, it is not a useful value to change. It may, however, be useful in a case where a mesh network is only a backhaul network. The problem with increasing the transmit power when there are mobile clients is that while the increase may permit a mobile client to better hear a node, it does not mean that the node can hear the mobile client any better. This is because the transmit power of mobile clients, such as a laptop, PDA, or smart phone, is not usually adjustable by the user. In our case, mesh clients can directly associate with any of the mesh nodes, so the transmit power setting was kept at the default of 28 mW. This default was found to work well and provide good signal strength and integrity between mesh nodes.

Encryption is not used in the mesh network because of the additional overhead incurred by using encryption. We are not concerned with the ability of encryption or security for this work. The nodes are all running in clear, unencrypted mode to provide the best possible performance. With encryption enabled, we would expect to see a slight decrease in performance, both in total throughput and in call quality. Find below a table outlining the details of the chosen settings.

A. Mean Opinion Score

To measure call performance, a value known as the MOS is used. This is an industry standard number, detailed in ITU-T recommendation P.800 [13] that represents perceived call quality and ranges from 1 to 5. In large scale testing, this

TABLE I
DD-WRT MESH ROUTER SETTINGS

Setting Name	Value
Basic Settings / Wireless Mode	Client Bridge
Adv. Settings / Basic Rate	Default
Adv. Settings / Transmission Fixed Rate	11 Mbps
Adv. Settings / CTS Protection Mode	Disable
Adv. Settings / Frame Burst	Disable
Adv. Settings / WMM Support	Enable
Adv. Settings / No-Acknowledgment	Disable

value is actually calculated using provided scores from many people listening to a call. The ITU introduced a method of calculating MOS using a formula called the E-model.

In voice communications, and used primarily for VoIP and voice data that is compressed, the MOS is a representation of the quality of human speech. To determine MOS for a specific configuration, a number of listeners rate the quality of test sentences read by both male and female speakers. Each sentence is given a rating, from 1 to 5, 1 being the worst, and 5 being the best. The MOS of a specific configuration is the arithmetic mean of the individual MOS values as recorded by the listeners. Typical MOS values of a cell phone call are in the upper-3 range while a land phone line ranks in the mid-4 range.

In our research, we are looking for a MOS value of 3.1 or greater. Below 3.1 is considered unacceptable quality and any call that has a MOS of less than 3.1 is not counted as a successful call [3].

MOS is a very useful means of measuring voice quality, as it allows for easy comparison of voice call quality from one test to the next. The drawback with MOS, however, is the cost of hiring many people to listen to sentences and rate them.

The E-model is a recommendation published by the ITU-T, given recommendation identifier G.107 [3]. It is a transmission rating model that gives an estimated call quality based on network factors. The most recent version of the E-model was approved by the ITU-T in the year 2005 [3].

IV. PERFORMANCE EVALUATION

In this section, we test many different scenarios and comparing the performance of the calls in each scenario. The types of tests we execute are as follows. We test the effect of increasing number of hops and background traffic on voice call quality. Following this, we test the effect on call quality as more calls are added to the network at different hop counts. This allows us to determine the maximum number of supported calls at different hop counts. Then, we test the fairness exercised by the network on calls. After this, we test the amount of bandwidth that is available in other areas of the network when calls are active. This allows us to test what effect making calls in a certain area of the network has on other areas. Then, we test call quality of simultaneous calls with different QoS parameters.

A. *Effect of Number of Hops and Background Traffic on VoIP*

The n-Hops performance evaluation is meant to determine several important factors via a large set of tests. The objective is to determine test running values of the EDCA parameters CWmax and AIFS for the voice traffic class. To do this, we obtain call qualities, in the form of a MOS value, obtained from packet loss, jitter, and end-to-end delay in many different cases. The motivation of the n-Hops performance evaluation is to demonstrate the effects of increasing the number of hops in the network and modifying the EDCA parameters. In addition, the tests are performed with and without background traffic in order to show the effect of additional traffic on voice calls in different cases.

TABLE II
AVERAGE CALL QUALITIES WITH DIFFERENT CWMAX SETTINGS

CWmax	Average MOS
No 802.11e	3.81
7 ms	3.72
15 ms	3.88
31 ms	3.96
63 ms	4.04
127 ms	3.87
255 ms	3.96
511 ms	3.96
1023 ms	3.96

A CWmax setting of 63 ms nets the best average MOS result. Using this setting, with various AIFS settings, the results are as follows:

TABLE III
AVERAGE CALL QUALITY COMPARISON OF 63 MS CWMAX WITH AND WITHOUT AIFS ADJUSTMENT

CWmax, AIFS	Average MOS
7 ms (default), 3 ms	3.72
63 ms, 3 ms	4.04
63 ms, 15 ms	4.14

Using a CWmax of 63 ms for the voice traffic class and an AIFS of 15 ms for the background traffic class results in better performance than the defaults. These parameters are used in all of the next tests. Fig. 2 shows the MOS of both 1 and 2 calls, with and without background traffic.

With our chosen settings of 63 ms for the CWmax and 15 ms for the AIFS, we have a 0% drop from the maximum MOS of 4.41 in call quality for a single call ranging from 1 to 5 hops. The MOS does not drop at all. When this is extended to include background traffic, we see a 2% drop in call quality from one to two hops. We see a further 2% drop in call quality from two to three hops, a 4% drop in call quality from three to four hops, and a 4% drop in call quality from four to five hops. This is a total drop in call quality of 12% from one hop to five hops.

When the number of calls is increased to two with no background traffic, the change from one hop to two hops and

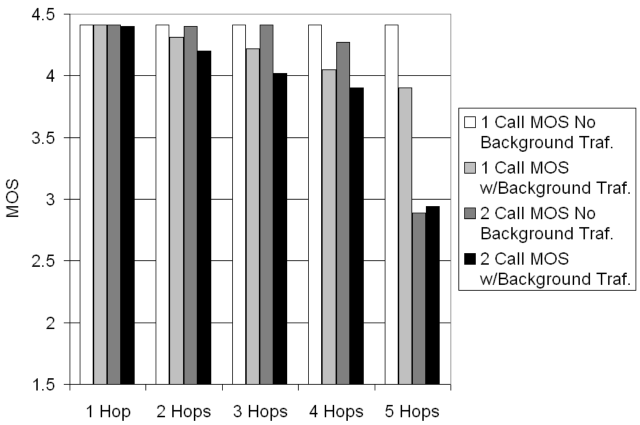

Fig. 2. MOS of 1 and 2 calls, with and without background traffic, CWmax 63 ms, AIFS 15 ms

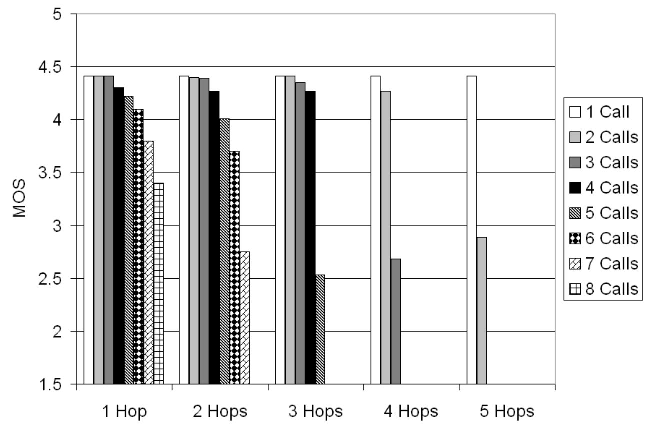

Fig. 3. Average MOS of voice calls at each hop count with 1 to 8 simultaneous calls

two to three hops is not noticable, with a 0% drop. The change from three hops to four hops shows a 3% drop in call quality. When this is extended to five hops, the drop in call quality from four to five hops is 32%. This suggests that two calls at five hops is not sustainable on the mesh network testbed. This is a total drop in call quality of 34% from one hop to five hops versus a total of 3% from one to four hops.

When background traffic is included, the drops between hops are more significant. A 5% drop is seen between one and two hops, a 4% drop between two and three hops, a 3% drop between three and four hops, and a 25% drop between four and five hops. This is a total drop in call quality of 33% from one to five hops versus a total of 11% from one to four hops. Again, two calls at five hops are not sustainable.

B. Maximum Number of Supported Calls

The objective of the maximum supported calls test is to determine the maximum number of voice calls that can be supported at different numbers of wireless hops. The motivation of this test is to be able to define numbers that represent the maximum number of voice calls supported depending on how many wireless links are traversed in the test bed WMN. These numbers are useful for planning and deploying an actual WMN.

In gathering the results for this test, we considered the number of calls that can be supported at each hop. At one hop, eight simultaneous calls can be supported. At two hops, the number of supportable calls drops to six, a 25% drop. When the network is extended to three hops, four simultaneous calls can be supported, a further 33% drop. At four hops, two simultaneous calls can be supported, and at five hops only one simultaneous call can be supported. The fifth hop data supports the data from the first test showing a large drop in quality when two calls are active at five hops. They both show that only one call can be supported with the minimum quality level (MOS 3.1) maintained. A graphical representation of this data can be found in Fig. 3.

C. Fairness of Calls at Different Hop Counts

The objective of testing fairness between voice calls is to determine if the network maintains fairness between calls with the addition of the 802.11e QoS mechanism.

For example, two similar or identical calls should exhibit the same performance on the network in ideal conditions. The testing is done with the QoS settings of CWmax and AIFS as defined earlier. The configuration tested are chosen to show if the fairness is affected when calls begin to compete with one another for access to the wireless medium.

The fairness index used is the Min-Max fairness index [14]. This index is sensitive to degradation in call quality. A result of 1 means that the network treats the calls fairly, while less than 1 means less than fair allocation of resources. Fairness is measured between the lowest MOS value of a test set, and the highest MOS value of a test set. To calculate the Min-Max fairness index, the below formula is used:

$$Index = \frac{x_{min}}{x_{max}} \tag{1}$$

where x_{min} is the lowest MOS value of a test set

and x_{max} is the highest MOS value of a test set

This test examines the data recorded in the maximum supported calls test to see how fair simultaneous identical calls are. Calls overlap completely and in each case are exactly the same number of hops and traverse the same mesh nodes. What we expect to see in this test is that the calls are not perfectly fair, but do not have hugely different MOS values. Table IV shows individual MOS numbers for each active call, as well as the average for each hop count. For the single hop case, eight calls were active with an average MOS of 3.4 and Min-Max fairness index of 0.95. The two hop case has an average MOS of 3.7 with a Min-Max fairness index of 0.93. For both the three hop and four hop cases, the average MOS is 4.27 and the Min-Max fairness index is 0.98 and 0.99 respectively.

The five hop case, given that there was only one call active, does not have a Min-Max fairness index. The individual MOS numbers for each call are shown in the same column, above the average MOS.

TABLE IV
RESULTS OF FAIRNESS OF MAXIMUM SUPPORTED CALLS TEST

Call	1 Hop	2 Hops	3 Hops	4 Hops	5 Hops
1	3.33	3.55	4.31	4.29	4.41
2	3.37	3.72	4.24	4.25	—
3	3.51	3.80	4.28	—	—
4	3.40	3.66	4.24	—	—
5	3.43	3.72	—	—	—
6	3.38	3.75	—	—	—
7	3.40	—	—	—	—
8	3.39	—	—	—	—
Average	3.4	3.7	4.27	4.27	4.41
Fairness	0.95	0.93	0.98	0.99	—

The results show that while the individual call MOS numbers do vary slightly from call to call, no one call stands out as having noticeably better quality or noticeably worse quality. The network treats the calls fairly, as can be seen in the bottom row of Table IV. Since the QoS parameters are equal for all VoIP traffic, call quality varies only marginally from call to call even with many active simultaneous calls.

D. Available Bandwidth When Calls Are Active

We now test how much of a bandwidth decrease calls cause depending on how far away from other types of traffic they are. The objective of this test is to examine how much of an effect voice calls have on other traffic when the voice calls and other traffic are spaced out, nearby, and adjacent. The following tests all have a voice call that must traverse three wireless hops, followed by one final test with a call that must traverse four wireless hops.

We did this by separating calls and background traffic first by two hops, then by one hop, then by having them adjacent to one another. Following this, we extended the call length by one hop to test if the additional hop of the call outside the interference range of the background traffic would effect the background traffic at all.

In the first part of this test, with a call spaced from the background traffic by two hops, the MOS of the call was 4.41, and the background traffic 3081 kbps. When the space was decreased to one hop, the MOS of the call was 4.24 and the background traffic 2885 kbps. This is a drop of 4% in the call quality and 6% in the speed of the background traffic. When the space between the call and the background traffic was decreased to zero hops, that is, adjacent to one another, the MOS of the call dropped to 4.14 and the background traffic 2507. This represents a further drop of 2% for the call quality and 13% for the background traffic. When stacked, this represents a total drop of 6% for the call quality and 19% for the background traffic when going from no interference to adjacent and interfering.

The final part of this test extended the length of the call by one hop. The extra hop on the call did not affect the background traffic throughput at all because the extra hop was outside of the hearing range of the background traffic.

E. Simultaneous Calls with Different QoS Parameters

This section examines call quality when some calls have QoS enabled while others do not. The calls are placed in different QoS queues by changing the DSCP flag in both X-Lite, the software SIP phone, and the Asterisk server. The non-QoS enabled calls are placed in the background traffic queue, while the QoS enabled calls are placed in the voice queue.

With three calls that overlap exactly, the call quality varied significantly depending on which calls had QoS enabled and which did not. For example, when Call 1 was QoS-enabled while Call 2 and Call 3 were not, they experienced a call quality 6% lower than Call 1.

With the second part of this test, we tested call quality of two calls with spacing between them. When both calls are active simultaneously, they do not effect one another, and as a result, the MOS is high for both calls at 4.41 in the case where one call is QoS-enabled and the other is not. When background traffic is added, both call qualities decrease. The call that is not QoS-enabled has a call quality MOS that is 7% lower than the QoS-enabled call.

With the third part of this test, we tested call quality of two adjacent calls, one that was QoS-enabled, and one that was not. When both calls are active and there is no background traffic, call quality MOS of the not QoS-enabled call is 1% lower than the QoS-enabled call. With background traffic added, this number increases to 9%.

F. Summary

Recall the n-Hops test, where we tested voice call quality with varying numbers of hops from one to five on the WMN. We looked at one call and two calls, both with and without background traffic, while adjusting the CWmax parameter for the voice traffic and the AIFS parameter for the background traffic. In our testing, a CWmax of 63 ms and AIFS of 15 ms gave us the highest call quality of all the parameter value combinations we tried.

In the maximum calls test, we tested how many calls can be supported simultaneously at different hop counts. Our results are summarized below.

TABLE V
MAXIMUM NUMBER OF SUPPORTED CALLS

Hop Count	1	2	3	4	5
Maximum Num. of Calls	8	6	4	2	1

In the fairness test, we studied how well the network maintains fairness between different calls. Calls were maintained fair by the network when expected, showing that the 802.11e QoS mechanism does treat calls fairly when able to.

In the available bandwidth test, we studied the effect of varying numbers of calls on available bandwidth in adjacent portions of the network. We first separated calls by two hops, then one hop, and then having them adjacent to one another. Results showed that background traffic throughput dropped by a total of 19% when the call was moved from two hops distance from the background traffic to adjacent to the background traffic.

In the different QoS settings test, we looked at call quality when different calls have different QoS settings. Calls placed in the voice traffic class had higher call quality than calls that were placed in the background traffic class.

The experiments reveal that the 802.11e QoS mechanism can support the desired QoS for voice calls. When compared with call quality when 802.11e QoS is disabled, the call quality is improved on average, in all cases except for the one where the CWmax of the voice traffic class is set to 7 ms. In this case, the low CWmax setting has a large effect on decreasing the call quality for the four and five hop tests. In addition, the experiments revealed that the default EDCA parameters for the 802.11e QoS mechanism are not suitable on a multi-hop mesh network. Modified parameter settings have been provided and used in the testing. With these settings, increased call quality is observed over multiple hops. It is important to note, however, that voice over an 802.11 mesh network is not really practical to deploy except in a very small user base because the number of calls that can be supported is limited.

When the new parameter settings are used in the single hop case, the voice call quality is not adversely affected by a large amount, but, the CWmax setting for voice should be kept at the minimum possible value for any given hop count. The CWmax and AIFS setting we used were chosen by testing all different combinations of both parameters.

V. Conclusion

In recent years, VoIP has been increasing in popularity. Convergence of communications onto IP networks is happening everywhere, with video, voice, and other realtime services being sent over the same networks as other data, such as web and E-mail traffic. Different services have different delay tolerances and requirements to work properly. The IEEE 802.11e amendment to the IEEE 802.11 standard adds support for QoS to better support multiple services concurrently.

In this paper, we focused on the ability of 802.11e to meet the QoS requirements for voice calls on a WMN. We gave background on the 802.11 protocol, WMNs, VoIP, SIP, and discussed some other VoIP performance works. We discussed our implementation, which includes an experimental WMN testbed used for testing call quality. Following this, we discussed the way we record and process the data. We then evaluated call quality in a variety of configurations and different scenarios on our WMN testbed. We identified performance metrics to use for studying the quality of VoIP calls. We detailed the individual test scenarios and presented the results and conducted a series of tests to examine if the 802.11e QoS mechanism provides any improvement over the original 802.11 standard.

Voice quality was measured using recorded data about end-to-end delay, jitter, and packet loss. A comparable measure was calculated by using the E-model.

The 802.11e QoS mechanism does help improve call quality, however the default settings cause problems with mesh networks. To improve voice call quality with a similar configuration as our wireless mesh network testbed, we recommend the following:

- Configuring the AIFS parameter for the background and best effort traffic classes to a higher than default value. Doing this desensitizes voice traffic from background traffic without impacting background traffic throughput. In our research, we used the value of 15 ms. If it was set higher, background traffic throughput was noticeably impacted.
- Configuring the CWmax parameter for the voice traffic class to a higher value than the default one. Doing this decreases the contention on the wireless medium over multiple hops and helps increase the voice call quality at longer hop counts without causing excessive delays. In our research, we used the value of 63 ms. When the CWmax value was set higher, further improvement in call quality or background traffic throughput was not seen.
- Configuring a fixed transmission rate of at least 11 Mbps so that MAC-level errors do not automatically drop the rate too low.
- Ensuring 802.11e is enabled between mesh routers by using a wireless sniffer. Configuring mesh routers with DD-WRT in Client Bridge mode accomplishes this.

Using these guidelines will help to improve call quality on WMNs. Fairness will be maintained on the network when comparing similar types of calls. In addition, background traffic throughput will not be severely impacted.

In reality, VoIP over 802.11 mesh networks is not very practical, especially in extended hop cases. Actual deployment to serve a large number of users will require provisions to accomdate a higher number of calls that can be supported simultaneously. In order to improve the voice call quality more, further study is needed.

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