Voice over Wireless Local Area Network (WLAN) Performance Analysis

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Abstract

The modeling and simulation of communication networks is a powerful tool that helps us design new systems and analyze changes in existing networks. In this paper, we present a student project on the simulation of voice over 802.11 wireless local area networks (WLANs). The network model, performance analysis, simulation results and findings provide a good example of exposing students to new technologies through network simulation.

1 – Introduction

In the design of new systems, modeling and simulation allow us to verify new architectures before their actual implementation. In the analysis of existing networks, modeling and simulation allow us to identify bottlenecks and evaluate the impact of new users, applications or changes to the network infrastructure. Using Opnet™ as a software tool to simulate and model computer networks, our course on Communication Networks Modeling, Simulation and Testing, in the Telecommunications Engineering Technology program at Texas A&M, teaches our students to evaluate and identify limitations in network architectures and protocols.

Moreover, one of our goals is to teach them to integrate new network architectures and protocols that have not been used together very often. New and usually expensive equipment may not be available in our laboratories to test these new technologies. Thus, we are taking advantage of our communication networks modeling and simulation course to teach new technologies and protocols and test their integration.

As an example of this approach, this paper presents a course project that our junior students performed. The goal of this project was to evaluate voice over IP (VoIP) over 802.11 wireless local area network (WLANs). As discussed in [1], "both IP voice and 802.11 WLANs are new technologies, and so the base of practical experience in merging the two is small." Voice over IP applications are real-time applications with strict requirements on delays and packet losses. A question arises on how the WLAN protocols - at the physical and medium access control (MAC) layers - support the real-time voice applications. In more detail, what is the efficiency of these protocols to support a large number of simultaneous voice users?

First, we presented our students with several articles [1, 2, 3, 4, 5] that discussed the performance of 802.11 physical layer technologies (e.g., 802.11b and 802.11g) and VoIP traffic. Initially, the students were presented a couple of articles only. As they needed more information to do the project they found relevant articles and shared them among themselves. Among the things that interested them in those articles were that the maximum transmission rates of the 802.11 MAC protocol can be very low with respect to the nominal data rates (e.g., 6.06 Mbps maximum transmission rate, whereas the nominal rate is 11Mbps, based on [3]). This project helped them to understand some of the causes of this low performance.

We proposed this project to our students so that they could use the network simulator Opnet™ to verify the theoretical maximum number of calls for VoIP over 802.11 WLAN, similar to the experiments done in [2, 4, 5]. We evaluated two technologies:

- IEEE 802.11 b networks running at 1 Mbps, 2 Mbps, 5.5 Mbps, and 11 Mbps,
- IEEE 802.11 g networks running at 6 Mbps, 18 Mbps, 36 Mbps, and 54 Mbps,

with three types of audio codecs: G.711, G.729a, G.723.1 (all with silence suppression). Twenty-four scenarios were simulated in Opnet to come up with results supported by theory.

This paper consists of four sections. The first three sections show a compilation of the students' projects: first, they were asked to present an overview of WLANs and the MAC protocol; second, they performed a theoretical analysis of the estimated throughput of each WLAN at different data rates for different audio codecs; then, they presented their simulation model and results. After this compilation, we draw our conclusions in the last section.

2- Overview – 802.11 WLANs and Voice over IP

Wi-Fi is a standard developed by working group 11 of the IEEE LAN/MAN standards committee. The current technologies in use by this standard include 802.11a, 802.11b, 802.11g and most recently 802.11n. This standard covers the first two layers of the open systems interconnect (OSI) model.

IEEE 802.11 wireless LANs use the distributed coordination function (DCF) as the main access method (i.e., Layer 2 of the OSI model). DCF is a contention-based scheme that is based on the Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) MAC protocol. Under DCF mode, a channel must be idle for a certain time referred to as the DCF interframe space (DIFS) before a station can start transmitting in this channel. However, if the channel is busy, the station enters a random backoff procedure before it can transmit. With this scheme, there is no guarantee that time-sensitive data will be delivered in any minimum amount of time.

IEEE 802.11 WLANs also support a contention-free scheme called point coordination function (PCF) mode. During the PCF mode transmission time, the wireless access point (AP) has full control of all the transmissions by polling the users one by one. If a user has data to transmit, there is no risk of collisions. The transmission sequence for the wireless scenario modeled in this project is based on the model shown in Figure 1. The PCF interframe space (PIFS) interval is used when a station transmits data. The short interframe space (SIFS) is used for sending acknowledgements for an existing dialogue.

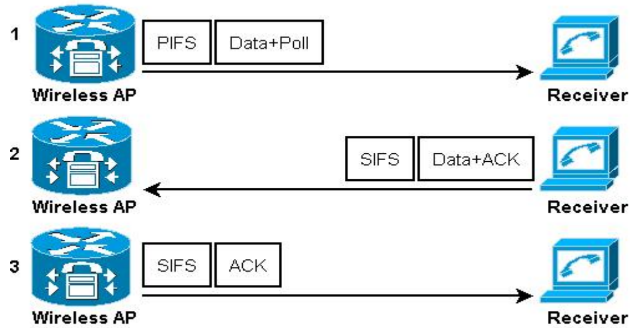

Figure 1 - Content-free transmission model (PCF mode)

We propose to simulate the PCF mode. The work of Chen *et al.* [5] shows that the PCF mode supports better voice over IP applications with silence suppression (i.e., variable bit rate) than the contention-based DCF mode.

For 802.11b and 802.11g, the PIFS interval has different values, because of different characteristics of the physical layer. Although both 802.11b and 802.11g operate in the 2.4GHz frequency range and employ direct sequence spread spectrum, 802.11g extends 802.11b to higher data rates. Thus, it has smaller time slots and consequently smaller PIFS interval. Table 1 summarizes some of the physical layer parameters of both protocols.

Table 1 - 802.11 physical layer parameters

Physical Layer Parameters	**802.11b**	**802.11g**
SIFS (Short Interframe Space)	10 μs	10 μs
Time Slot	20 μs	9 μs
PIFS = SIFS + Time Slot	30 μs	19 μs
Frame Preamble	192 μs	20 μs
Frame Extension	NA	6 μs
Symbol Size	8 bits	216 bits

Considering VoIP applications, the 802.11 frame is composed of the audio codec sample and all the control information from the different layers of the communication networks (Figure 2). The layered architecture of communication networks adds overhead to the application, such as the Real Time Transport Protocol (RTP) header (12 bytes), User Datagram Protocol (UDP) header (8 bytes), IP header (20 bytes), and 802.11 MAC header and checksum (36 bytes).

We are using the User Datagram Protocol (UDP) instead of the Transport Control Protocol (TCP) because we are transferring real-time data. Most of all real-time multimedia applications employ UDP. When UDP receives the voice data sample it sends it out right away to the network without any buffering, acknowledgements, or retransmissions. The on-time delivery of packets is crucial in interactive applications such as voice over IP.

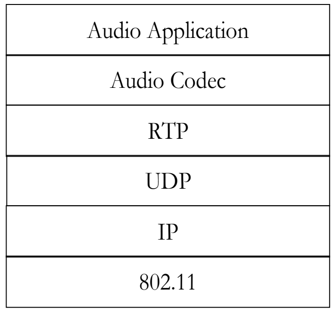

Figure 2 - Protocol architecture

Codec Comparisons

There are three codecs used in this project: G.711, G.729a, and G.723.1. The codec data size varies depending on the codec and sampling rate. Based on [6], we have assumed the rates and data sizes shown in Table 2, with one codec sample per 802.11 frame. All of these elements of each codec will have an impact on the throughput and maximum amount of calls.

Table 2 - Codec parameters

Codec	**Bandwidth (kbps)**	**Sample time (ms)**	**Data size (bytes)**
G.711	64	20	160
G.723.1	5.3	30	20
G.729A	8	20	20

In typical voice over IP calls there is silence on either end of the conversation at any given time. During this time data from the silenced end does not need to be transmitted. Thus, a technique called silence suppression is used. Silence suppression saves network bandwidth by not transmitting silenced data, but still keeps up with the synchronization on the conversation. For this project, a silence suppression rate of 40 percent (meaning that 40 percent of the data on either end is not sent) is assumed.

3 – Delay Analysis

Before testing the networks, it was necessary to determine the theoretical number of calls over the different wireless networks, at different speeds. First, we performed a delay analysis of the transmission sequences in a VoIP over WLAN environment using PCF mode (Figure 1). Considering one voice call as two UDP flows, we have used one flow as data (i.e., the 802.11 encapsulated voice frame), and the other flow as the 802.11 ACK + data. For simplicity reasons, we left out the polling sequences associated with the PCF mode.

Considering the different PCF inter-frame space (PIFS) and physical layer parameters for the WLAN physical layer technologies considered (Table 1), we estimated the actual throughput of the WLANs. This can give us an idea of how many simultaneous voice calls it could handle. The throughput for each speed is not equal to that of the advertised speed of the protocol. Codec data, preambles, and headers proved to be critically important to the final results of analytical and simulated testing.

The throughput values were obtained by estimating the total time to send data and then dividing the codec sample size by this time. Table 3 gives a sample of the time calculations for 802.11b and G.711 codec, and Figure 3 shows the throughput values (as a percentage of the nominal data rates) for 802.11b and 802.11g WLANs.

Table 3 - Calculating total time to send data (two flows)

G.711	802.11b				802.11g			
	1 Mbps	2 Mbps	5.5 Mbps	11 Mbps	6 Mbps	18 Mbps	36 Mbps	54 Mbps
PIFS (µs)	30.0	30.0	30.0	30.0	19.0	19.0	19.0	19.0
Preamble (µs)	192.0	192.0	192.0	192.0	20.0	20.0	20.0	20.0
802.11 data (µs)	1888.0	944.0	377.6	171.6	324.0	108.0	54.0	34.7
Extension (µs)	0.0	0.0	0.0	0.0	6.0	6.0	6.0	6.0
SIFS (µs)	10.0	10.0	10.0	10.0	10.0	10.0	10.0	10.0
Preamble (µs)	192.0	192.0	192.0	192.0	20.0	20.0	20.0	20.0
802.11 ack+ data (µs)	1888.0	944.0	377.6	171.6	324.0	108.0	54.0	34.7
Extension (µs)	0.0	0.0	0.0	0.0	6.0	6.0	6.0	6.0
Total (us)	4200.0	2312.0	1179.2	767.3	729.0	297.0	189.0	150.4

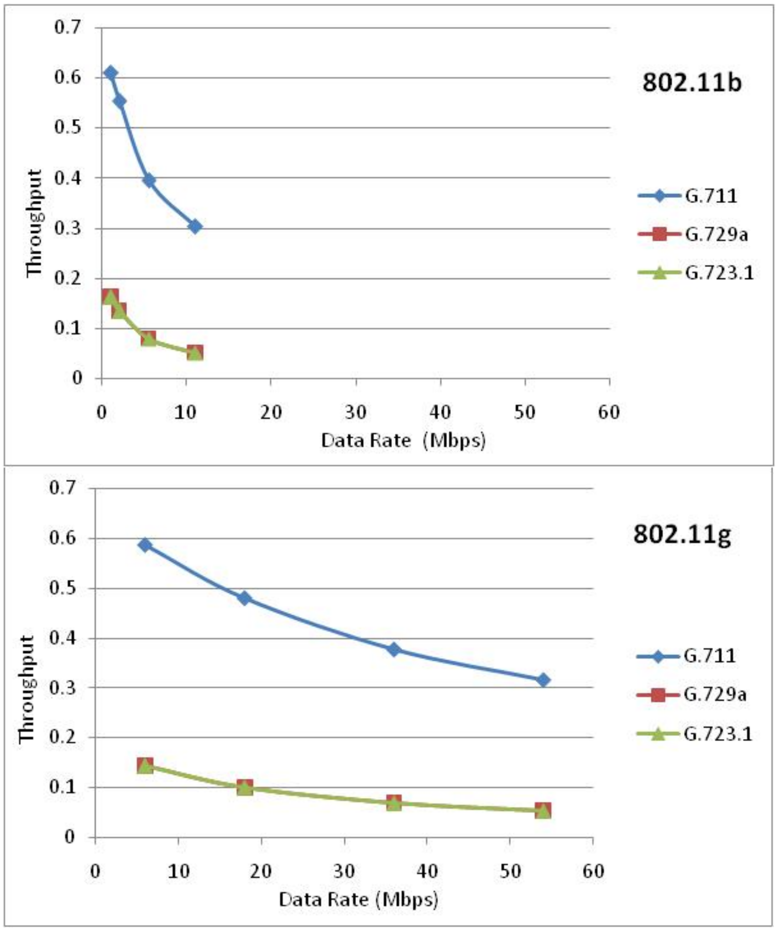

Figure 3 - Estimated throughput for VoIP over WLAN

We can see in Figure 3 that the smaller the audio sample size (e.g., for G.729a and G.723.1), the smaller is the throughput (i.e., the overhead is proportionally larger). Note that G.729a and G.723.1 have the same sample size (20 bytes), so their throughput values coincide. Furthermore, for each WLAN technology, the throughput decreases as the data rate increases. The reason is that even though the 802.11 encapsulated voice frames are sent at faster transmission rates, the physical layer overhead characteristic of each WLAN technology (e.g., preambles, inter-frame spaces) is fixed (i.e., more bits could be sent during those times). To the throughput calculations this translates into more overhead and less efficient transmission.

4 – Network Simulation Model and Results

Simulation Model

In this class, the Opnet Modeler version 11.5 Educational Version is used. The configuration of the model networked for all situations is shown in Figure 4. It consists of 20 wired workstations and 20 wireless laptops. The wired workstations are connected to a standard LAN router (using Ethernet ports), which is the linking device between the workstation and a wireless access point or WLAN router. The WLAN router and wireless laptops will be matched in wireless 802.11b or 802.11g protocol, operating channel, and transmission speed. Various transmission speeds will be used in each scenario. The configuration of each scenario was done automatically by changing the programmed parameters, as shown in Figure 5.

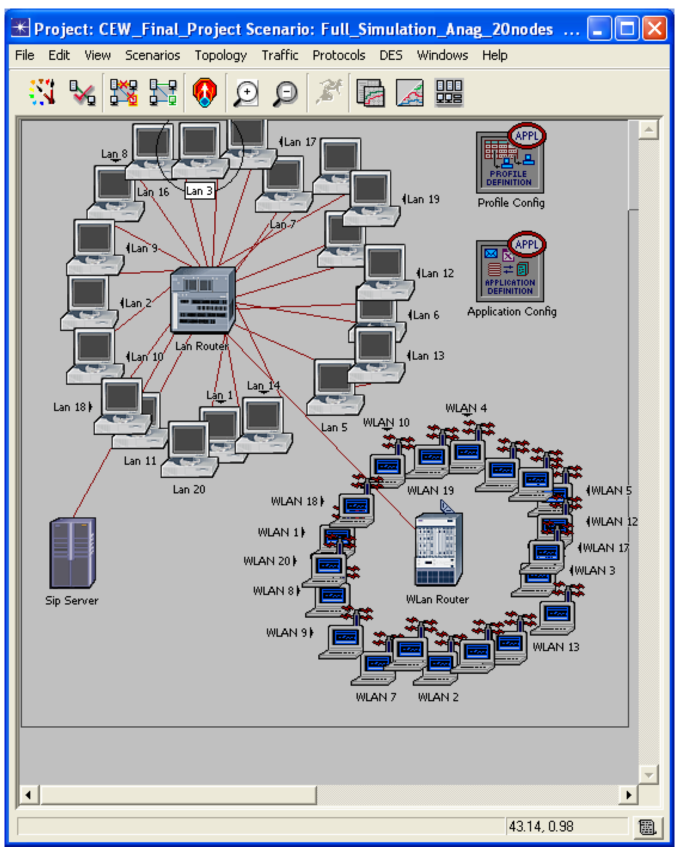

Figure 4 – Network layout

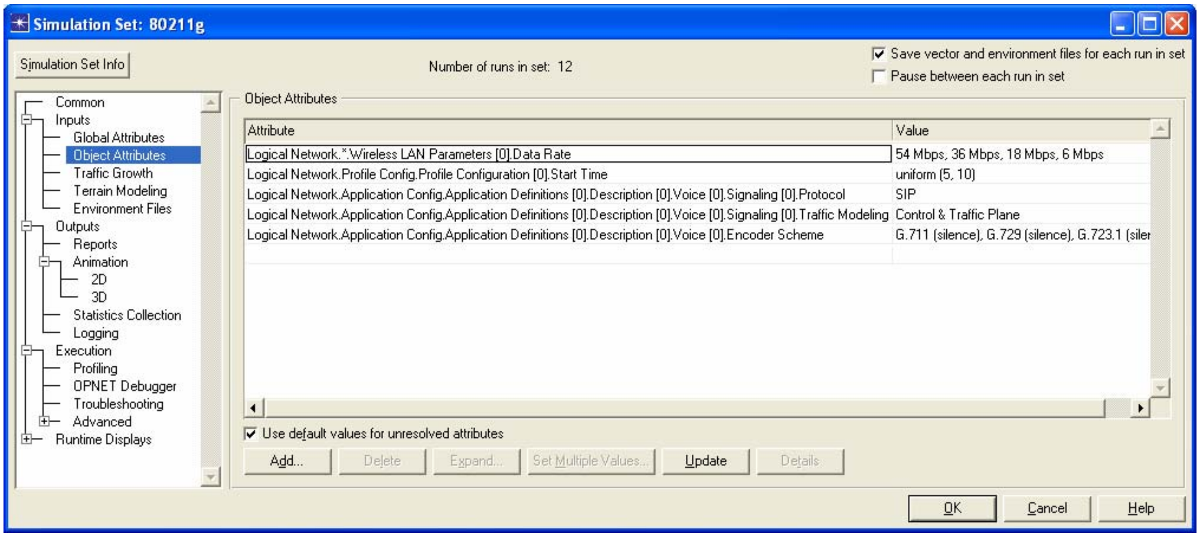

Figure 5 – Automatic scenario configuration

VoIP calls occur between a wired node and a wireless node. A silence suppression rate of 40 percent was assumed. To simulate a large number of calls and at the same time have a scalable model, in our model a node can run simultaneous VoIP calls. They are initiated one by one within exponential intervals with an average interval of 60 seconds. The calls take place until the end of the simulation. To keep track of the number of active calls in the network, we use the Session Initiation Protocol (SIP) server connected to the LAN router. This server acts as a proxy and all the call signaling to initiate each new call goes through this server.

The maximum number of calls was determined by analyzing the end-to-end packet delay and percentage of packet losses with each additional call. The criteria for the maximum number of calls were that the end-to-end packet delay should be less than 200 msec, and no packet losses. When one of these conditions fails, we register the time it occurs, and the number of calls at that time at the SIP server. For example, Figure 6 shows the case of G.711 codec, and 802.11b at 11 Mbps. At approximately 24 sec, there are packet losses (e.g., WLAN access point's buffer overflows). The delay is approximately 115 msec, and starts increasing shortly after. The number of active calls at the SIP server at this time is 32 calls.

This number of calls is not necessarily the maximum number of calls that can be placed in the network, but it is the maximum number of calls that the network can sustain without any noticeable impairment. Though more calls can be placed on the network past this point, it is assumed that beyond this marker the Mean Opinion Scores (MOS) will begin to deteriorate.

The performance of VoIP calls over the PCF mode depends on the duration of the contention-free period (which runs PCF mode) and the duration of the contention period (which runs DCF mode). Based on the results presented in [5], we considered that the contention period cannot exceed 20 msec, or queues will build up at the nodes since the inter-arrival of VoIP packets that we used is 20 msec for G.711 and G.729a codecs. We used a contention free period of 50 msec and a contention period of 20 msec.

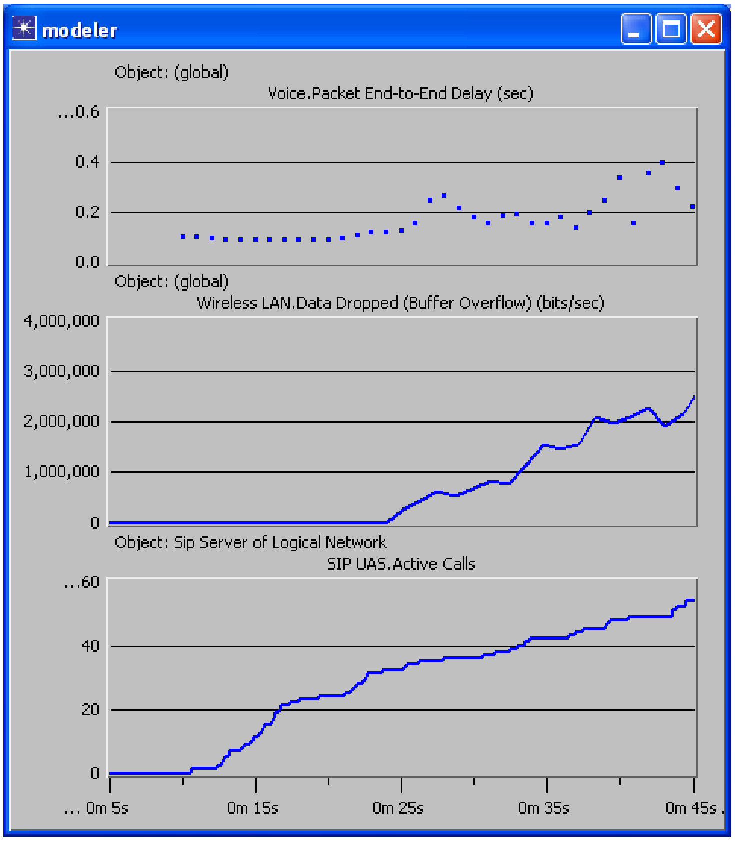

Figure 6 - Criteria for determining the maximum number of calls: delays and packet losses. This case represents 802.11b WLANs, at 11 Mbps, with applications using G.711 codecs.

Simulation Results

Table 4 illustrates the results for the maximum number of calls obtained in each scenario. G.723.1a codec has the best performance. Having a larger interval between samples (30 msec as opposed to 20 msec) proved to be more efficient for voice applications running over 802.11 WLANs. However, we noticed that the end-to-end delay measured for the maximum number of calls for G.723.1a codecs was slightly higher than for the other codecs. For instance, for 802.11g WLANs running at 54 Mbps, the average end-to-end delay was 112 msec for G.723.1a codecs, as opposed to an average of 88 msec for G.711 and G.729a codecs.

In order to compare our analytical results from Section 3 with the results obtained through simulations, we have plotted both types of results in Figure 7, for the 802.11g experiments. Although the analytical values were approximated values, they provided us with some guidelines on the maximum number of calls per 802.11 technology and data rate. Note that we have made some assumptions on our model (e.g., by using only 20 wireless stations with multiple active calls per station) and we also have made simplifications in our delay analysis. Moreover, the enabling of silence suppression in the codecs showed to be something difficult to model and to predict in the simulator's operation.

Table 4 - Simulated call results

Codec	802.11b				802.11g			
	1 Mbps	2 Mbps	5.5 Mbps	11 Mbps	6 Mbps	18 Mbps	36 Mbps	54 Mbps
G.711	6	8	21	32	32	75	111	149
G.729a	12	24	34	44	70	132	171	188
G.723.1	19	36	49	65	102	204	256	298

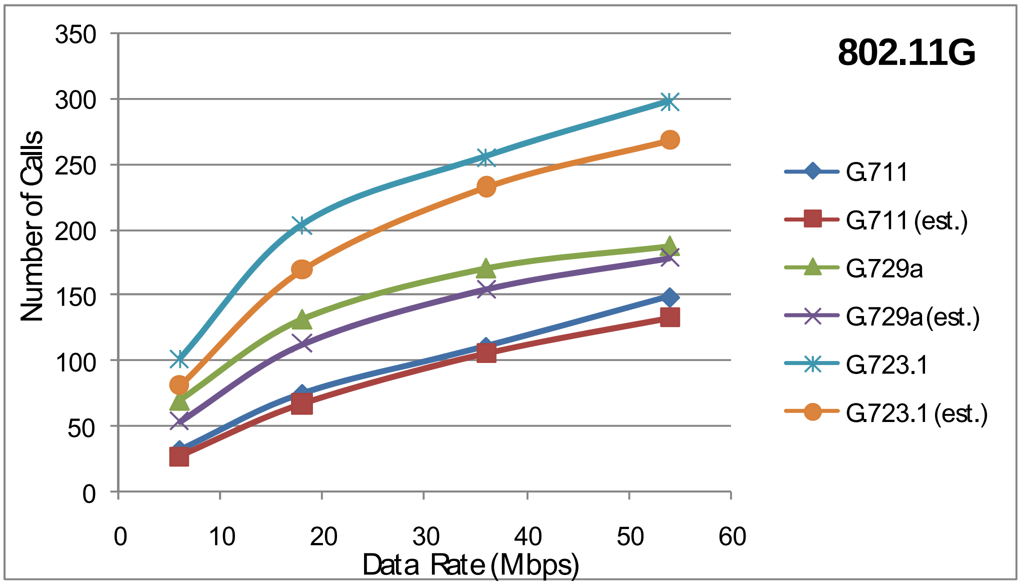

Figure 7 – Simulated vs. estimated results for different codecs over 802.11g WLANs.

5 – Conclusions

Throughout the project, the fact that our students questioned the impact that individual packet sizes have on wireless networks and the efficiency of 802.11 MAC protocol to transfer VoIP codec samples showed to be a great learning experience for them. As an example, one of our students wrote:

“Many different technologies and codecs were covered over the course of this assignment as well as the use of Opnet in a way never attempted throughout the course of lab. Coming away from this lab, a comfort with VoIP technology as well as comfort with the OSI model as a whole can be gained.”

In addition, they could see that ideal calculations and simulations are not exactly the same. This difference would likely be even greater if tests with real equipment were performed. They understood that factors like the wireless access point saturation and buffer size were not addressed in our simple delay analysis calculations, nor were details of the MAC operation such as the duration of the contention-free period and contention period. However, with the help of a network simulation tool, they have experienced how these factors can impact VoIP over WLAN systems.

6 – Acknowledgements

We would like to thank all our students that took the Communication Networks Modeling, Simulation and Testing class during Fall 2006: Jesse Bruce, Lucas Folegatti, Robert Hegedus, Chris Magnussen, Jason McConnell, Ryan Schroederr, Justin Vierra, John Vaughan, and Charles Watkins. Charles Watkins' model is shown in this paper, and Robert Hegedus' full report initiated the idea of publishing this work.

References

[1] "Can WLAN Switches Support Voice?" Business Communications Review, pp. 42-47, October 2006.

[2] Gast, M. S., "802.11 Wireless LANs – The Definitive Guide," Chapter 25 – 802.11 Performance Tuning, 2nd Edition, O'Reilly Media.

[3] E. Ferro and F. Potorti, "Bluetooth and Wi-Fi Wireless Protocols, A Survey and A Comparison", IEEE Wireless Communications, Vol.12, No.1, pp.12-26, February 2005.

[4] G.-H. Hwang and D.-H.Cho, "Voice Capacity in IEEE 802.11 Wireless LANs," IEEE Electronics Letters, September 2004, Vol. 40, no. 18, September 2004.

[5] D. Chen, S. Gang, M. Kappes and K. Trivedi, "Supporting VBR VoIP Traffic in IEEE 802.11 WLAN in PCF Mode," Opnetwork Conference 2002.

[6] "VoIP Bandwidth Calculation", White Paper, Newport Networks, 2005, http://www.newport-networks.com/cust-docs/52-VoIP-Bandwidth.pdf.

Biography

Ana Goulart is an assistant professor in the Telecommunications Engineering Technology program in the department of Engineering Technology and Industrial Distribution at Texas A&M. She received her doctoral degree in Electrical and Computer Engineering at Georgia Tech in 2005. Her research interests are on computer networks and wireless communications, with focus on real-time applications related to telemedicine and remote monitoring.

Charles Watkins is a former student in the Telecommunications Engineering Technology program. Charles is an expert on computer programming and soon became quite familiar with the Opnet simulator.

Robert Hegedus is a former student in Telecommunications Engineering Technology program, who was actively engaged on student organizations such as the IEEE Communications Society. He received his degree in May 2007 and is currently working for Lockard & White on the design of wireless transmission systems.