Designing 802.11 WLANs for VoIP and Data

Lizzie Narváez†, Jesús Pérez††, Carlos García††† and Victor Chi†

†UADY Unidad Tizimín, ††ITESM Campus Cuernavaca, †††IIE, UADY Unidad Tizimín, México

Summary
VoIP transmission in a WLAN is becoming a necessity; nevertheless there is a very important problem, the fact that the typical access point (AP) distribution is not ideal to establish a proper communication in the WLAN, since they are deployed only for data transmission. This paper presents a procedure and its implementation in an experimental scenario for designing WLANs with VoIP support, using two of the most important current standards for WLANs: 802.11b and 802.11g. According to our results, design of WLAN for data and VoIP can be recommended in small places with standstill clients; results concerning to SNR, speed-distance between clients and APs vary depending on the APs' brand and the type of wireless adapters used in the clients. The expected amount of calls was not obtained, because of this amount was not according to the bandwidth utilization published by the codecs.

Key words:
WLAN, VoIP, 802.11b, 802.11g, standard.

1. Introduction

Since communications evolved from wired to wireless, companies and service providers have been witnesses of the new emerged possibilities. For example, implementing higher speed networks with lower cost including mobile services.

Currently, the most popular WLAN standard is the IEEE 802.11b, which can theoretically support data rates up to 11 Mb/s, however, this data rate is for optimal conditions [1]. When several users are working simultaneously, the real bandwidth is divided among the whole users. In order to obtain a good voice transmission on this standard using an IP-based telephone, there are several codecs that require less than 10 kb, for example [2,3]. In consequence the theoretically 11 Mb/s provided from this standard, it is enough for voice transmission [4] and, in principle, it could support more than 500 VoIP sessions [2], but the real problem is that only 5.5 Mb/s are the actual rate from those 11 Mb/s [5], we get this speed only while staying near the AP and when it is used only by one single node; in real life a network provides simultaneously services to many nodes and even for data transmission, therefore a better network design is needed to obtain a satisfactory communication. In agreement with [6], due to the large overhead involved in transmitting small packets, the achievable throughput for 802.11b is far less than its maximum of 11 Mb/s data rate that it currently supports. Hence, we can realize that most WLANs nowadays are not designed for voice support.

On the other hand, 802.11a and 802.11g networks have data rates up to 54 Mb/s and they are not designed to support voice transmission (because of the APs are not distributed in the most optimum way, so that communication can be established properly); they are used for data transmission, and a network only designed for data transmission is not ideal for voice transmission.

In this context, the main goal of this work is to set a WLAN design procedure to support VoIP and data, and its experimental implementation based on 802.11b and 802.11g standards with VoIP and data support, with the current technology.

Many studies have been made to evaluate the VoIP performance on the IEEE 802.11 standards; but as we can see in [7] they focused on presenting analytic results from simulations of the 802.11b, *a* and *g* standards, without setting a procedure to design it and for its experimental implementation. In [8] an analysis of the capacities and VoIP performance for WLAN is done. However as well as in [7], there is no either a procedure or an implementation. Results in [9] show several scenarios for VoIP implementations for corporate networks; however our research is not focused on corporate networks. In [4] some parameters for voice transmission are presented (using the 7920 Ideal hardphone), which are needed in our design procedure and we took them into consideration. On the other hand, the results in [10] show us a very important point, the coverage areas for 802.11b, *a* and *g*, which are essential for our procedure; however they do not carry out a VoIP transmission analysis.

2. Wireless Local Area Network 802.11

IEEE 802.11 is a communication standard developed by the IEEE in 1997 that became the first standard for WLAN [11]. Among the main variants of 802.11 we can mention 802.11a, *b*, *g* and a draft version called Pre-N of the 802.11n (the complete version would appear by the end of 2006 and it is expected to reach 500 Mb/s for real transmission data rate and it will have to work on 2.4 GHz and 5 GHz frequency bands) [12]. This research is bounded for the 802.11b and *g* standards.

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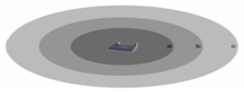
2.1 802.11b standard

In 1999 a modification to the original IEEE 802.11 standard was ratified, which was designated as IEEE 802.11b. Nowadays, this standard is at the top of the wireless networks success; this specification has data rates that range from 2 to 11 Mb/s, it works in the 2.4 GHz frequency band usually called Industrial Scientific and Medical (ISM), its modulation technique is Direct Sequence Spread Spectrum (DSSS) with the codification system of Complementary Code Keying (CCK) and it has 11 available channels in America, from which some of them are recommended to avoid overlapping: channel 1, 6 and 11. As we can see in figure 1 the approximated coverage is shown (without obstructions) of an 802.11b AP [10].

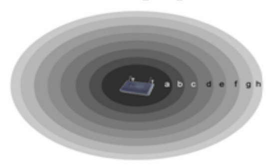


a	11 Mb/s DSSS 0 – 48 mts
b	5.5 Mb/s DSSS 48 – 67 mts
c	2 Mb/s DSSS 67 – 82 mts

Fig 1. Theoretical data rates of an 802.11b AP.

2.2 802.11g standard

This standard appeared in 2003 and as the 802.11a it has data rates up to 54 Mb/s, but just like 802.11b, it also operates in the same 2.4 GHz portion of the RF spectrum (ISM). It uses OFDM modulation for data transmission. 802.11g devices can also work at speeds up to 11 Mb/s, so that 802.11b and *g* can coexist under the same network; and its compatibility with standard b makes it more attractive. Equal to 802.11b, it counts with 11 channels and 3 non overlapping channels. [10, 1] In figure 2 ranges of coverage and theoretical data rates are shown without obstructions. [10]



a	54 Mb/s OFDM 0 - 27 mts
b	48 Mb/s OFDM 27 – 29 mts
c	36 Mb/s OFDM 29 – 30 mts
d	24 Mb/s OFDM 30 – 42 mts
e	18 Mb/s OFDM 42 – 54 mts
f	12 Mb/s OFDM 54 – 64 mts
g	9 Mb/s OFDM 64 – 76 mts
h	6 Mb/s OFDM 76 – 91 mts

Fig 2. Theoretical data rates of an 802.11g AP.

3. Voice over IP

Voice over IP can be defined as a set of applications that allow live voice transmission over the Internet using TCP/IP protocols. The VoIP standard was defined in 1996 by the ITU [13].

It can be said that independently of the protocol used for VoIP transmission; there must be at least three fundamental elements in the structure: the client (who makes the call by means of softphone or hardphone), the server (who coordinates the calls) and the gateway (the binding with the traditional public switched telephone network who acts in a transparent way to the user) [13, 14].

Protocols are the language that different VoIP devices use for their connection; being SIP and H.323 the most used at the moment [15].

Another important aspect is the codec. A codec is an algorithm (program) that is used to encode and decode a voice conversation over IP. The most commonly used codecs are: g711a, g711u, g726, g728, g729, iLBC, gsm and speex; which vary mainly in bandwidth use by each call and the quality they offered [16].

Experiments [17] with VoIP in IEEE 802.11b networks showed that the effective available bandwidth in the wireless network is reduced by ongoing VoIP connections.

4. Design Procedure of WLAN for VoIP

Even though in state-of-the-art appears 802.11n like the best candidate to support VoIP because of its high data rates and distances, the procedure was not based on this standard because it is not finished yet and there is only a draft version called pre-N. On the other hand, manufacturers do not guarantee compatibility with the final standard even between their same products. Finally the cost of acquiring equipment based on this technology is relatively high, compared with the currently popular ones. From these reasons we decided to work with 802.11b and 802.11g standards.

The procedure for 802.11b and 802.11g standards is as follows:

1. *Determination of physical coverage volume.* It is necessary to have a plan of the places where wireless coverage is needed, both spaces with obstacles, as well as areas without construction. It is needed to calculate the available total volume. Total volume can be separated in two parts: outdoor and indoor volume.
2. *Verification of Signal Noise Ratio (SNR) from the AP.* When VoIP is transmitted through a WLAN the SNR received in each client should be at least 25 dB or more [4] without receiving a signal from another AP that works in the same channel and is greater than 10dB. It is important to determine the SNR that is received by the hardware that we are going to use and carry out the corresponding measurements in order to define the right distances to place the APs. Literature recommends an overlapping of 30% of the radios theoretical values (figure 1 and 2) in order to have a complete volume coverage; but it is advisable to verify it.

3. *Determination and verification of the speed-distance relation.* As mentioned in [4] voice transmission would be properly established if:
   - 802.11b standard has the suggested data rates of 11 Mb/s; that implies a radio coverage of up to 48 mt of distance from the center of the AP (table 1).

Table 1. Theoretical data rates of 802.11b for VoIP.

Data Rate in Mb/s	Distance in mts
11	0 – 48

   - 802.11g standard has at least data rates above 18Mb/s; it means that it has a radio coverage of up to 54 mts of distance from the center of the AP (table 2).

Table 2. Theoretical data rates of 802.11g for VoIP.

Data Rate in Mb/s	Distance in mts
54	0 - 27
48	27 - 29
36	29 – 30
24	30 – 42
18	42 – 54

It is important that the clients are located at these distances while transmitting in order to get the required speed.

4. *Determination of the number of access points.* It is important to determine the volume of coverage and take into consideration the SNR (step 2), and the relation speed-distance (step 3). Dividing the total volume between the diameter of coverage of each AP (after attenuation considerations) we will have an approximate number of APs.
5. *Considering signal attenuation by obstacles.* When needed, you must consider obstacles, but the principle to determine the APs amount it is the same that the one in step 4 (table 3 and 4) [18].

Table 3. Loss in dB for different kinds of material (TO - Type of Obstacle)

TO	Loss	TO	Loss
Open space	0 dB	Thick walls	15 – 20 dB
Window (tinted nonmetallic)	3 dB	Very thick walls	20 – 25 dB
Window (tinted metallic)	5-8dB	Thick ground/ceiling	15 – 20 dB
Thin walls	5 - 8 dB	Very thick ground/ceiling	20 – 25 dB

Table 4. Obstacles and attenuation of signal

Kind of Obstacle	Attenuation
Wood, plastic, synthetic materials, crystal	Low
Bricks , leaves	Average
Ceramic, water, paper, cement, bulletproof crystal	High
Metal	Very high

6. *Determine the possible power losses.* According to Cisco's studies the rule for increasing and decreasing power (equation 1) is as follows:
   - Outdoor (without obstructions), an increase of 6dB will double the distance and a decrease of 6dB the distance range will be cut in half.
   - Indoor (with obstructions), an increase of 9dB will approximately double the distance and a decrease of 9dB the range will approximately be cut in half.

$$dB = 10 \log_{10} \left(\frac{Power\_of\_A}{Power\_of\_B} \right) \quad (1)$$

Measure of the gain or loss in dB

7. *Determine the channels distribution*. Having in mind that only three non overlapping channels, 1, 6 and 11 are available, they must be used without assigning the same channel in two contiguous cells. This is because they can produce interference each other (figure 3).

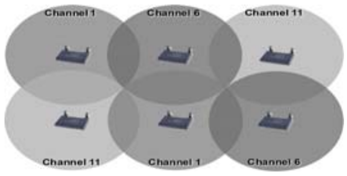


Fig 3. Distribution of frequency channels

8. *Transmission powers must be identical in the transmitting and the receiving equipments.* Emitter and receiver must transmit at the same power; this point applies mainly when hardphones are used [4].
9. *Guarantee a 45% channel usage from an access point.* The index of channel usage for data from an AP must be smaller than 45% [4], to guarantee a good quality of voice; this value was obtained from experimental studies carried out at Cisco's laboratories, leaving a 55% available in order to guarantee that the calls will have an acceptable quality. This can be obtained applying QoS priorization in the AP to obtain specific bandwidth or creating VLANs.
10. *Verify the real network performance.* Additionally it is recommendable to make tests to verify the real network performance using a specific software.

11. *Determine the number of clients for voice and data* that will be connected to the WLAN in a specified period of time, for this, we can use the formula 2 (this is only an initial approach, because reality could be different from theory):

$$Number\_Calls = \frac{Correc\_Fac\left(RB - RB\_Data\_Used\right)}{Codec} \qquad (2)$$

where:

Number_Calls	*Number of Calls.*
Correc_Fac	*Correction factor of real network performance.*
RB	*Real bandwidth usage.*
RB_Data_Used	*Real bandwidth used for data transmission.*
Codec	*Bandwidth used by the codec to establish a call.*

Previous studies have demonstrated that using a WLAN only for VoIP, without data transmission, the network can support up to 7 concurrently calls using a g711 codec [17], up to 8 calls using a g729 codec [4] and up to 12 calls with a gsm codec [4].

12. *Carry out tests to avoid interference from other networks.* It is important to consider if there are WLANs near our WLAN, where APs channels from our neighbors do not interfere with ours.
13. *Adjustments to the coverage area to avoid interferences.* An AP can cover certain distance depending on the standard we use:
    - 802.11b standard can cover up to 82 mts of distance, considering that only the first 48 mts are usable for voice, the other 34 mts are not usable, therefore cellular area must be fit only to the 48 mts from the AP to avoid interferences.
    - 802.11g standard can cover up to 91 mts of distance, considering that only the first 54 mts are usable for voice, the other 37 mts are not usable, therefore cellular area must be fit only to the 54 mts from the AP to avoid interferences.

Graphs of the main design for outdoors using the 802.11b and *g* standards and considering only 6 APs are illustrated in figures 4 and 5. For indoors all the considerations mentioned in previous steps must be considered. We have a total coverage area of 259.2 mts of length and 177.6 mts wide with 802.11b standard and 291.6 mts of length and 199.8 mts wide with 802.11g.

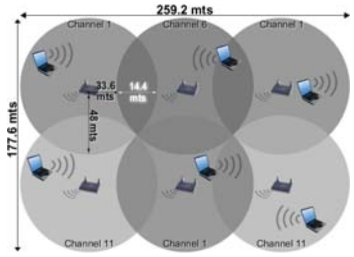


Fig 4. Initial design for the 802.11b standard.

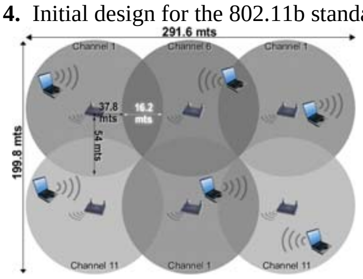


Fig 5. Initial design for the 802.11g standard.

5. Implementation

In this section we present the hardware and software used in the implementation, and the description step by step of the proposed procedure.

The hardware used was the following: two AP Cisco model *AIR-AP1231G-A-K9* 802.11g compatible with 802.11b, two AP 3Com model *OfficeConnect Wireless 11g*, an AP Proxim model Orinoco AP-2000, a switch 3Com® OfficeConnect® with 5 ports, 11 laptops with their corresponding 802.11b and 802.11g adapters.

The software used was the following: Voice Server Asterisk (to set-up the VoIP calls), eyeBeam and X-Lite of Xten (softphones), SIP (VoIP protocol), g711u and gsm (VoIP codecs), Apache Server (to transfer files), Net Meter (to measure bandwidth), Network Stumbler (to obtain the SNR network values), UDPFlood (to flood the network with traffic) and Explorer (to configure the AP).

Next we will describe the implementation step by step:

1. *Determine the physical coverage volume.*
   In our case we want to cover an outdoor area of approximately 180 square mts, so we need 4 APs to carry out this implementation. It was performed outdoors without obstacles and the area to cover was according to the consideration mentioned above about a SNR of 25dB or higher, without receiving any signal from other AP in the same channel being higher than 10dB. For this reason we place the APs with an overlapping area of 30% after doing the proper measurements to get the right ranges in order to obtain the speed needed. Figures 6 and 7 show the covered area.

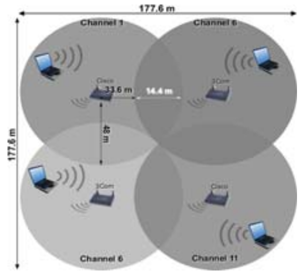


Fig 6. Coverage area for 802.11b standard.

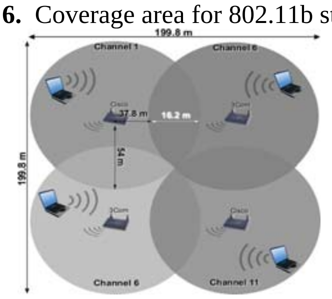


Fig 7. Coverage area for 802.11g standard.

2. *Verify the Signal Noise Ratio (SNR) of the access points.* We use the Network Stumbler to measure the SNR in suitable distances; up to 48 mts for the standard *b* and 54 mts for *g*, using 3 AP brands (Tables 5 and 6).

Table 5. Values obtained with 802.11b standard.

Distance of measurement	**SNR in dB**		
	Cisco	**3Com**	**Proxim**
0mts	70 to 65	65 to 60	53 to 50
10mts	45 to 38	43 to 39	30 to 28
15mts	45 to 30	43 to 39	30 to 27
20mts	39 to 35	35 to 33	15 to 10
30mts	25 to 24	25 to 24	16 to 8
40mts	25 to 24	23 to 22	12 to 8
48mts	23 to 22	20 to 17	Less than 12
More than 48mts	Less than 22	Less than 17	

Table 6. Values obtained with the standard 802.11g.

Distance of measurement	**SNR en dB**		
	Cisco	**3Com**	**Proxim**
0mts	49 to 44	54 to 36	52 to 50
10mts	36 to 35	36 to 35	35 to 32
20mts	35 to 30	28 to 24	32 to 24
27mts	28 to 27	28 to 24	30
42mts	26 to 17	23 to 22	25 or under wthsignal loss.
54mts	25 to 17	22 to 7	
More than 54mts	Less than 17	Less than 7	

A pattern among the results from the different brands does not exist. We can notice that for 802.11b it is desirable that the client could be connected to a distance of up to 40 mts or less with the Cisco AP, 30 mts or less with the 3Com AP and 15 mts or less with de Proxim AP. These distances are those where we obtained a SNR of 25dB. For the 802.11g the client could be up to 54 mts or less with the Cisco AP, up to 27 mts with the 3Com AP, and up to 20 mts with the Proxim AP. During the determination of the SNR from the 3 brands, it was determined the distance where another AP working at the same channel could be place. We concluded that an overlapping of 30% along the APs coverage radius is needed: For the 802.11b, the radius must have an overlapping of 14.4 mts (with 11 Mb/s) and for the 802.11g, the radios must have an overlaping of 16.2 mts (with 18 Mb/s).

3. *Determination and verification of the speed-distance relation.* Network Stumbler was used for this purpose, collating its results with the Windows indicator, in order to determine the speeds that were obtained at the distances the standards propose (Tables 7 and 8).

Table 7. Speed–Distance Relation: 802.11b (DM-Distance of Measurement, SR-Speed Reached)

DM	**SR (Cisco AP)**	**SR (3Com AP)**	**SR (Proxim AP)**
0mts a 20mts	11 Mb/s	11 Mb/s	11Mb/s
30mts			11Mb/s to 6Mb/s
40mts			6 Mb/s
48mts			2 Mb/s to 1Mb/s
More than 48mts	Even less than 1Mb/s	6 to 1 Mb/s, or signal loss	signal loss

Table 8. Speed–Distance Relation: 802.11g (DM-Distance of Measurement, SR-Speed Reached)

DM	SR (Cisco AP)	SR (3Com AP)	SR (Proxim AP)
0mts	54 Mb/s	54 Mb/s to 48 Mb/s	48 Mb/s to 36 Mb/s
10mts	54 Mb/s to 36 Mb/s	36 Mb/s	36 Mb/s to 18 Mb/s
20mts	36 Mb/s to 24 Mb/s	24 Mb/s	18 Mb/s to 12 Mb/s
27mts	24 Mb/s to 18 Mb/s	12 Mb/s	12 Mb/s to 6 Mb/s
29mts	18 Mb/s	12 Mb/s to 6 Mb/s	signal loss
30mts	18 Mb/s to 12 Mb/s	12 Mb/s to 6 Mb/s	
42mts 54mts	12 Mb/s 12 Mb/s to 2 Mb/s	6 Mb/s to 1 Mb/s	

4. *Determine the number of access points.* We used 4, previously mentioned.
5. *Consider the attenuation of the signal by obstacles.* The experiment was outdoors with a loss of 0 dB by obstacles; the free space loss of the signal is always considered.
6. *Determine the possible power losses.* The verification of power was done with the Network Stumbler in all the cases during the tests for the determination of the SNR.
7. *Determine the channels distribution*. We had 4 APs and there was only one repeated channel. We did not observe any interference.
8. *Transmission powers must be identical in the transmitting and the receiving equipments.* Only softphones were used; the powers used by adapters were the same.
9. *Guarantee a 45% of channel usage from an access point*. The used 3Com AP did not support QoS functions and VLANs [19], so tests were needed to fit the traffic to an amount less than 45%, injecting traffic to the network (to simulate the 45% of data usage and later to test the network with traffic and without traffic). We used an Apache web server and clients so that we first proceed transferring without traffic and afterwards with traffic, using UDPFlood to simulate the QoS policies (Tables 9 and 10).

Table 9. Values of 45% usage of an AP, 802.11b standard (RAT-average transference rate)

Manufacturer	Without traffic	With traffic
Cisco	370.56 Kb/s	277.51 Kb/s
3Com	362.44 Kb/s	273.11 Kb/s
Proxim	It was not possible to be determined.	

Table 10. Values of 45% usage of an AP, 802.11g standard (RAT-average transference rate)

Manufacturer	Without traffic	With traffic
Cisco	460.03 Kb/s	319.31 Kb/s
3Com	468.25 Kb/s	355.44 Kb/s
Proxim	It was not possible to be determined.	

10. *Verify the real network performance.* In order to be able to determine the number of calls, the real bandwidth was determined in each one of both standards, by means of files transference between clients and the file server. Several transferences were carried out in this point for both standards, locating the clients to different distances from the AP (0 to 48 mts for 802.11b and 0 to 54 mts for 802.11g) (Tables 11 and 12).

Table 11. File transference results for 802.11b standard.

RAT Without traffic			RAT With traffic		
Cisco	3Com	Proxim	Cisco	3Com	Proxim
370.56	362.44	254.21	277.51	273.11	139.73
RAT = average transference rate in Kb/s					

Table 12. File transference results for 802.11g standard.

RAT Without traffic			RAT With traffic		
Cisco	3Com	Proxim	Cisco	3Com	Proxim
460.03	468.25	410.8	319.31	355.44	142.28
RAT = average transference rate in Kb/s					

11. *Determine the number of clients for voice and data.* In this section two important tests have been carried out. First, we determined the number of clients for voice and data that could be connected to the WLAN at any time either with traffic or not, and in the second test we determined whether the clients can roam among APs while calling without losing the connection. The voice calls were carried out using first a gsm codec (13 Kb/s bandwidth) and afterwards a g711u codec (64 Kb/s bandwidth) [20]. With both codecs the calls were carried out first without network traffic and afterwards the traffic was injected, maintaining a 45% channel usage from an AP; using the value previously found. Calls flow in this test occurred between clients located in APs with the same brand. Table 13 presents the collected data, evaluated to a distance of 20m.

Table 13. Calls results.

Standard	Codec	Number of calls	State of the network
802.11b	Gsm	6e, 2g, 1lq	wt
		5e, 1vg, 2g	with
	g711u	7e, 1vg	wt
		6e, 2vg	with
802.11g	Gsm	7vg, 2lq	wt
		6vg, 3lq	with
	g711u	8e	Wt
		8e	with

e = excellent, vg = very good, g = good,
lq = low quality, wt= Without traffic,
with= With traffic

In the second test carried out for this point, calls were set up while the client was located aside an AP and moving towards another AP. It was not able to maintain the call since in all the cases it was cut and later when arriving to another AP it had to be set up again. Therefore we concluded that the roaming can not be established even at layer-2 level, in other words, hand-off is not achieved at least with the equipment used.

12. *Carry out tests to avoid interference from other networks.* This fact was verified with the values obtained from the Network Stumbler.
13. *Adjustments to the coverage area to avoid interferences.* During the test we obtained the proper values of SNR, so we did not need to make any adjustment.

6. Results

The experiments were carried out with integrated and external adapters in the computers, the internal adapters showed better SNR values than the external ones.

The distance-speed relation, as it was appreciated in the procedure implementation, does not match with the theory reports (being clear about existing differences). SNR begins to degrade in a short distance from the AP, and in order to obtain excellent quality calls we needed a SNR of 25 dB o higher. Also some differences exist among manufacturers. It was not obtained any excellent quality call for the 802.11g standard with the gsm codec. g711u codec presented a better MOS (the most popular measurement of clarity according to ITU-P.800 specification) [21] than gsm in all the tests carried out. Even though more calls were obtained with gsm codec than with g711u, this is not significant considering that gsm consumes less than a quarter of bandwidth than g711u and the number of set-up calls were less than we hoped specially for the gsm codec. It was observed that strong wind causes radio signal loss; we lost the connectivity of some calls during strong wind gusts in both AP brands. The calls set up between clients connected to the same AP, but with close clients presented quality problems because of the interferences that they caused among one and other. The Cisco AP had better coverage and presented higher speeds than 3Com. The calls with excellent quality were obtained with a SNR above 25dB. We did not find significant difference between results from the 802.11b in comparison to the 802.11g. We did not observed significant differences between the 2 standards. We were expecting a better performance especially with the 802.11g. In previous tests performed in a WLAN designed only for data without using our suggested procedure, we got a lower number of calls set up because of the bandwidth consumed by data and because some clients were not able to get the proper speed to set up a call. Also a bigger number of disconnections were found while clients move to the outer areas of the cells. In this way, we have observed that following our procedure, we improve the network performance in order to support voice and data as much as possible.

7. Conclusion

The result from this research work was the creation of a design procedure (pattern) and distribution of APs for WLANs with VoIP and data support, considering a WLAN 802.11.

This procedure is useful to design a WLAN that support both voice and data obtaining the maximum performance. Furthermore the implementation is not complex and it has the advantage that it can be scalable to other standards.

An experimental scenario was implemented to prove its performance, for each one of the two standards and the following topics were evaluated: speed–distance relation, SNR, actual network performance, amount of calls that can be maintained simultaneously.

In conclusion, the obtained results are different from the ones reported from the manufacturers; substantial differences between the several evaluated brands were found.

As a result from this experimental research work we can conclude and suggest that: Designing a wireless network with 801.11b or *g* standards for VoIP and data as an extension of the wired one, it can only be feasible in small places (hot spots) with standstill clients, in short distances and with a low number of calls. The results concerning to SNR and speed-distance for indoor would be lower than the values we obtained in our experiment for outdoor because of the obstructions, which would cause less calls or would demand more APs. There was not the expected amount of calls according to the required bandwidth that codecs published, so it is concluded that an additional overload exists while calls are set up. Clients roaming is not achieved between 2 different APs, we did not manage to keep a set-up call while roaming between 2 APs. The proposed procedure was designed very close to the theory; nevertheless the implementation showed us that we obtained very different values. So, we demonstrated that not only differences with the theoretical values exist but also with different APs brands and type of wireless adapters.

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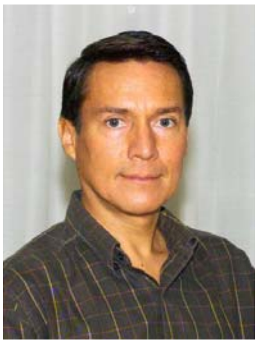
Lizzie Edmea Narváez-Díaz [lendiaz@uady.mx] received the degree in Computer Science from the the Autonomous University of Yucatán (UADY) in 1997. She received the Master of Computer Science degree from Monterrey Technological Institute (ITESM), Campus Cuernavaca, in 2007. Currently, she is a full time teacher at the Autonomous University of Yucatán since 2000 at the Network department in Tizimín México. She has participated in development projects of software engineering. Currently is giving courses on wireless networks in the professional programs in the UADY.

Jesús Arturo Pérez-Díaz [jesus.arturo.perez@itesm.mx] obtained his B.Sc. degree in Computer Science from the Autonomous University of Aguascalientes (1995). He worked as a system and network administrator in the Aguascalientes city hall. In 1997 he started his PhD degree in the University of Oviedo, Spain. He studied the program New advances in Computer Science Systems, and during these years he researched in mobile agents and published around 12 research papers in magazines and international conferences. During the same period of time he took part in a couple of European research projects and became a full associate member of the European founded research project AgentLink. He got his PhD Degree in Computer Science in the University of Oviedo in 2000, afterwards he became associate professor of the Computer Science Department in the same University from 2000 to 2002.

During these years he gave courses and made research in network and Internet security. Nowadays he is a researcher and professor in the ITESM – Campus Cuernavaca México and member of the Researchers National System, his research field focus in network security and wireless communications.

Currently Dr. Pérez-Díaz has gotten the recognized Cisco certifications CCNA and CCAI, which allows him to give Cisco certification courses. Additionally he is giving courses on wireless communications and network security in the postgraduate programs in the ITESM, also he has supervised master and PhD theses in the same field. He has given security courses in some European and South American Universities.



Carlos Felipe García-Hernández [cfgarcia@iie.org.mx] obtained his B.Sc. degree in Communications and Electronics from the University of Guanajuato, México, in 1983 and his M.Sc. degree in Telecommunications Systems from the University of Essex, England, in 1986.

He has National and International Publications, He has given Conferences and advised several B.Sc. and M.Sc. Theses. Also, He has attended Professional Training Courses in the U.S.A. and He gives Professional Courses in México, B. Sc. Courses in ITESM Cuernavaca {Monterrey Technological Institute - Cuernavaca Campus} since 1996 and Postgraduate Courses in CENIDET-SEP {National Center of Research

and Technological Development - Public Education Bureau} since 1987. He works at the Electric Research Institute {IIE}, Control and Instrumentation Department in Cuernavaca México, as a full time Researcher and Project Manager on Radio Communications and Telecommunications since 1983 and He has been in charge of several projects on Wireless and Mobile Networks.

Mr. García-Hernández was a National Researcher Level-I {Presidential Appointment made to Outstanding Researchers} from1987 to 1993, He is an IEEE Senior member, a CIGRE Corresponding member {International Council on Large Electric Systems} and a CIME member {Mechanical, Electrical and Electronics Professional Engineers Association}. Furthermore, He is a Professional Engineer and a Telecommunications Consultant, certified with registration No. 555 from the SCT {Telecommunications and Transport Bureau} and the COFETEL (Federal Telecommunications Commission) with the specialty on Radiocommunications since 1993 and therein He is also certified as a Class II Radio Amateur with registration No. 10013 since 1999 with call-sign XE1RFG.

Victor Manuel Chi-Pech [vmcpech@uady.mx] obtained his degree in Computer Science from the Autonomous University of Yucatan (UADY) in 1996 and his M. Sc. degree in Wireless Network from Monterrey Technological Institute (ITESM), Campus Cuernavaca, in 2007.
Victor Chi works since 2000 in the Autonomous University of Yucatan, as a full time professor. He has participated in development projects of software engineering. Currently is giving courses on wireless networks and software engineering in the professional programs in the UADY.