QoE Simulation Analysis of VoIP Traffic Support under Nakagami-m Fading Channels

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ABSTRACT: *The expansion of the wireless networks is one of the main reasons of the doubtless growth of the VoIP communications during the last years. This technology has been extensively studied by the research community from several points of views, such as, (i) the effect of delay, packet loss rate, jitter, etc. on the quality of the VoIP calls, or (ii) the systems capacities evaluating different bandwidths and coding schemes. However, very few works detailing the effect of the wireless physical layer can be found in the related literature. In this work we employ the Nakagami-m propagation model in order to evaluate the effect of fading channels on VoIP traffic for IEEE 802.11 systems and its influence on the Quality of user Experience (QoE). The results obtained employing this propagation model, which permit us to characterize the wireless channel with different levels of hostility (fading), are compared with those attained in a Free Space environment. We simulate several scenarios, with different codecs, different packetization time-intervals, and different bandwidths in order to study the impact of fading over system capacity, coverage range, and QoE, which is estimated by the E-Model. From our results, we observe that fading channels provoke a noticeable decrease in the system maximum capacity and coverage range. Furthermore, VoIP communications suffer a noticeable drop on the achieved QoE in scenarios affected by fading channels, mainly affecting to low bit-rate codecs.*

Keywords: VoIP, QoE, Fading

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1. Introduction

Voice over IP (VoIP) is a technology that has received an increasing attention by both, the research community and the end users. The former is focused on analyzing its performance for different scenarios to improve its level of Quality of Service (QoS) and Quality of user Experience (QoE), e.g., [1–4]. For the latter, VoIP is an attractive technology because it offers high quality voice calls at low cost allowing people to connect all over the world. Furthermore, the increasing expansion of Wi-Fi networks (IEEE 802.11), in both public and private domains, facilitates final users the establishment of cost-effective VoIP communications fulfilling higher levels of quality. Regarding quality issues, the design of multimedia services is progressively converging to user-centric approaches. For this reason, not only the QoS from a networking point of view is needed, but QoE should be also satisfied by assessing the level of quality that the customer actually perceives. The most extended methodology used to evaluate QoE is the absolute category rating (ACR) method, which outputs a Mean Opinion Score (MOS) that is a subjective rating of the service. However, using ACR is time-consuming, expensive, and does not permit continuous monitoring of networks. An alternative is the E-Model (ITU-T Rec. G.107). This model takes into account several transmission impairments, such as delay, echo, codec distortion, etc., which generate an additive rating scale called R. R assesses the conversational quality of a voice call and is used to predict customer's QoE, since it can be mapped onto a MOS scale. Despite the considerable work done in the related literature about VoIP performance in wireless networks, even considering QoS and/or QoE, to the best of our knowledge there are few published works addressing the detailed effect of the physical layer on QoE in VoIP traffic. For instance, the physical layer is influenced by path loss, shadowing, and fading, hence affecting the QoE as well.

In this paper, we analyze the effect of fading channels on VoIP traffic over an IEEE 802.11g system with QoE provisioning. We study this effect by simulation for a variable number of VoIP calls, different codecs, different bandwidths, and two VoIP packetization lengths. We use the Nakagami-*m* propagation model [5] to simulate outdoor and indoor scenarios with fading, and the Free Space model [6] to simulate an outdoor free space situation. In order to assess the effect of fading channels on different coding algorithms, two different codecs are used, the ITU-T standards g711 A-law and g726; different coding-rates are also employed, namely, 64 Kbps for g711 and 24, 32, and 40 Kbps for g726. Finally, we investigate the effect of fading on different packet lengths by means of two different packetization time intervals of 10 and 20 ms.

The rest of the paper is organized as follows. Section II reviews the related literature focused on analyzing the support of VoIP calls in wireless local area networks with QoS/QoE provisioning. A description of the simulation environment used in this study is included in Section III. Section IV shows the simulation results and discusses the effect of fading channels. The paper ends summarizing the most important facts.

2. Related Work

VoIP has been extensively studied during the last years. Some relevant examples of works that evaluate the capacity for VoIP traffic in wireless local area networks by simulation or experimentation are [2, 7–12]. In these works, the effect of layer two handoffs, delay constraints, channels with constant bit error rates, packet size, UDP traffic, TCP traffic, overheads from different layers, etc., have been intensively evaluated. From their results, we observe that the capacity achieved (defined as number of valid VoIP calls) is very different, of course, depending on the simulated and/or configured (testbed) network conditions. It is worthy to mention the work done by Shin and Schulzrinne [2], which studied the capacity of VoIP traffic in an IEEE 802.11b network for a wide range of network parameters. They showed that there are several factors such as the buffer size at the access point, the type of retry limit of the IEEE 802.11, or the preamble size that have a significant impact on the performance of VoIP, in particular, limiting the number of VoIP calls supported by the network. By using the g711 codec and a packetization interval of 20 ms, they obtained a capacity of 15 constant bit-rate calls and 38 calls in variable bit-rate VoIP traffic; understanding as maximum capacity the number of VoIP calls so that the 90^{th} percentile of the one-way end-to-end delay is less than 150 ms. However, it should be noted that neither this work nor the previous ones detailed the effect of more realistic physical layers. The physical layer plays a key role that cannot be ignored when we are evaluating QoE, since packet losses are directly impacted by this factor ignoring its origin, and QoE models usually include packet losses as one of their input parameters.

To the best of the authors' knowledge, only the works in [13] and [14] tackled the physical layer effect. In [13], the quality of voice communication over IEEE 802.11a networks is studied. Particularly, their authors included in the simulations a simplified version of the Keenan-Motley partition loss model incorporating a Free Space propagation model and shadowing, representing an office scenario with no line-of-sight. As a result, they showed the cumulative distribution functions of packet error rate for different Signal-to-Noise ratios (distances) and different data rates. Using stationary VoIP users subject to a particular fading realization, they obtained a Mean Opinion Score (MOS) for the VoIP calls using the E-model. They showed how MOS values are much greater if packet errors from fading are not taking into account. A more complete work was done in [14], where authors examined the behavior of 802.11s wireless mesh networks including Nakagami-*m* fading channels. They showed how the SNR should be increased to achieve similar QoE levels in VoIP calls than those achieved with Additive Gaussian Noise channels. In their study, they used the E-Model for QoE evaluation.

In contrast to these works, we evaluate the effect of fading on the QoE of VoIP calls, including a more realistic and complete physical model represented by the use of the Nakagami-*m* propagation model. We compare delay, packet loss, and MOS values obtained with the E-model to assess the maximum allowed capacity, i.e., not exceeding the recommended limits for delay and packet losses according to the ITU-T Rec. G.114 and G.1010, and achieving a MOS value that represents an "*acceptable for most users*" quality level. Compared to [14], we test the robustness of low bit-rate codecs by using g726, assessing the performance of VoIP on diverse 802.11g infrastructure scenarios, namely, indoor and outdoor with different distances among the nodes and the access point.

3. Simulation Environment

In order to estimate the performance of VoIP for different scenarios, we conduct a set of simulations. The framework chosen for the simulation study is Omnet++ v4.2.2 and the Inet framework [15]. In addition, we employ the *VoIPTool*, which is included in the aforementioned version of Inet framework. This tool allows the generation and transmission of realistic VoIP packet streams, as well as the customization of relevant parameters for a VoIP communication, such as packet length, coding rate, or VoIP header size. *VoIPTool* consists of two modules, namely, a traffic generator and a sink; the former creates a stream of VoIP packets from an arbitrary sound file using different audio codecs; as input, we use a set of the ITU-T Test Signals (Rec. P.50). On the other hand, the VoIP sink decodes the transmitted stream and produces two different wave files, namely, a file representing the original sound file, and the other, the encoded and transmitted one.

In our simulations, we use the ITU-T standard codecs g711 A-law (64 Kbps) and g726 for three different coding rates (24, 32, and 40 Kbps). We have chosen these codecs with the aim of comparing the effect of fading on two different kind of coding schemes: non-compression and low bit-rate codification. In order to assess the quality of the VoIP calls, an E-Model implementation has been added to the VoIP receivers following the guidelines of the ITU-T Rec. G.107 and G.113.

The simulated scenario is an Ethernet-to-wireless (802.11g) network topology illustrated in Figure 1. To study the performance for different available bandwidths, two different transmission rates of 11 and 54 Mbps included in the 802.11g standard are used. The access point buffer is set to store 100 frames and the retry limit is established as the 802.11 standard long value, i.e., 7. The wireless card modules are set to a transmission power of 5 mW and a sensitivity of -85 dBm, with a carrier frequency given by the 802.11 standard, i.e., 2.4 GHz, and a signal-to-noise threshold of 4 dB.

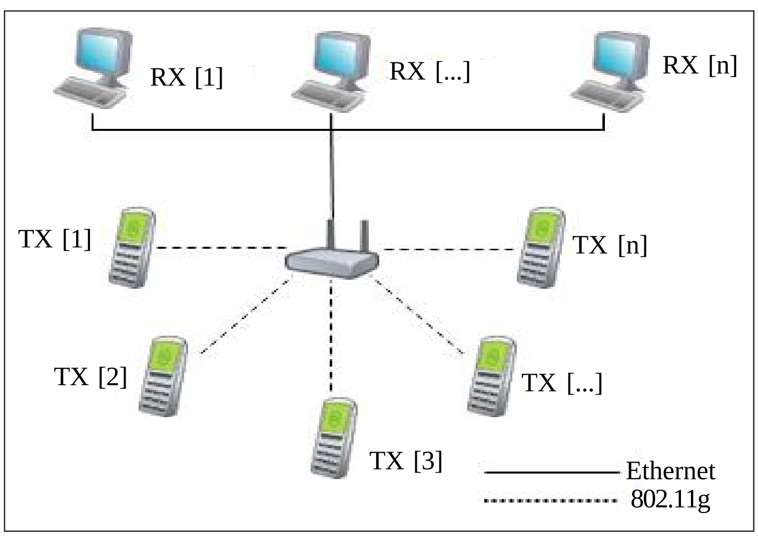


Figure 1. Simulation scenario (TX are the wireless transmitters, one for each VoIP call; RX[n] are the wired receivers, one for each VoIP call)

The effect of fading channels is assessed by carrying out simulations for the Free Space and Nakagami-*m* propagation models. The latter, is characterized by the shape factor m, the smaller the m value the greater the level of fading. If $m = 1$, the Nakagami-*m* model represents the Rayleigh model, which assumes the inexistence of a dominant contribution along a line-of-sight between transmitter and receiver. When $m > 1$, the Nakagami-*m* closely approximates to the Rice model, which introduces less fading in the transmission channel. Taking into account the radio parameters and these propagation models, we evaluate the system for a variable distance between the wireless VoIP nodes (*TX* in Figure 1) and the access point, ranging from 25 m to 475 m. We assume a VoIP header (i.e., RTP) of 12 bytes. In the simulations, we use two different packetization lengths, namely, 10 and 20 ms. The starting time for each VoIP call is chosen randomly with a uniform distribution in a time range of (0, 8*s*). The audio sources have a duration of 8*s*. Additional 802.11g parameters are set according to Table 1.

4. Results

In this section, we show the simulation results obtained to study the VoIP system behavior for different scenarios, analyzing separately the obtained results for the two different packetization intervals under consideration. We measure the MOS value, the delay, and the Probability of packet loss (Ppl) for every call to analyze how the distance, and therefore the fading channels, affects the system, in terms of capacity and conversational quality. In order to set each individual call as valid, three metrics are usually used in the bibliography: MOS value, one-way delay, and/or Ppl. Following the guidelines of the ITU-T Rec. G.114 and G.1010, we define a call as valid if the final MOS value attained for this call is over 3.1 (see Table 2). Furthermore, the maximum one-way delay introduced by the network and Ppl accepted are 80 ms and 5%, respectively. The former is calculated as follows; the one-way end-to-end delay of voice packets is supposed to be less than 150 ms (ITU-T Rec. G.114). Assuming the codec delay to be about 30-40 ms at both sender and receiver, and that Ethernet connections are simulated as ideal links, i.e., do not add extra delay to the communication, then the wireless network should contribute less than 80 ms delay.

Parameters	Bytes	11 Mbps	54 Mbps
SIFS, DIFS, SLOT (μs)	-	{10, 50, 20}	{10, 50, 20}
CW_{MIN} (slots)	-	31	31
PLCP preamble (μs)	-	144	4
{PLCP, MAC, SNAP} headers (μs)	–, 28, 8	{48, 20, 36, 5.81}	{16, 4.15, 1.18}
IP + UDP + RTP headers (μs)	40	29.09	5.92
Voice (g711, 10 ms) (μs)	80	58.18	11.85
Voice (g711, 20 ms) (μs)	160	116.36	23.70
Voice (g726-40, 10 ms) (μs)	50	36.36	7.4
Voice (g726-40, 20 ms) (μs)	100	72.72	14.8
Voice (g726-32, 10 ms) (μs)	40	29.09	5.92
Voice (g726-32, 20 ms) (μs)	80	58.18	11.84
Voice (g726-24, 10 ms) (μs)	30	21.18	4.44
Voice (g726-24, 20 ms) (μs)	60	43.62	8.88
ACK (μs)	14	10.18	2.07

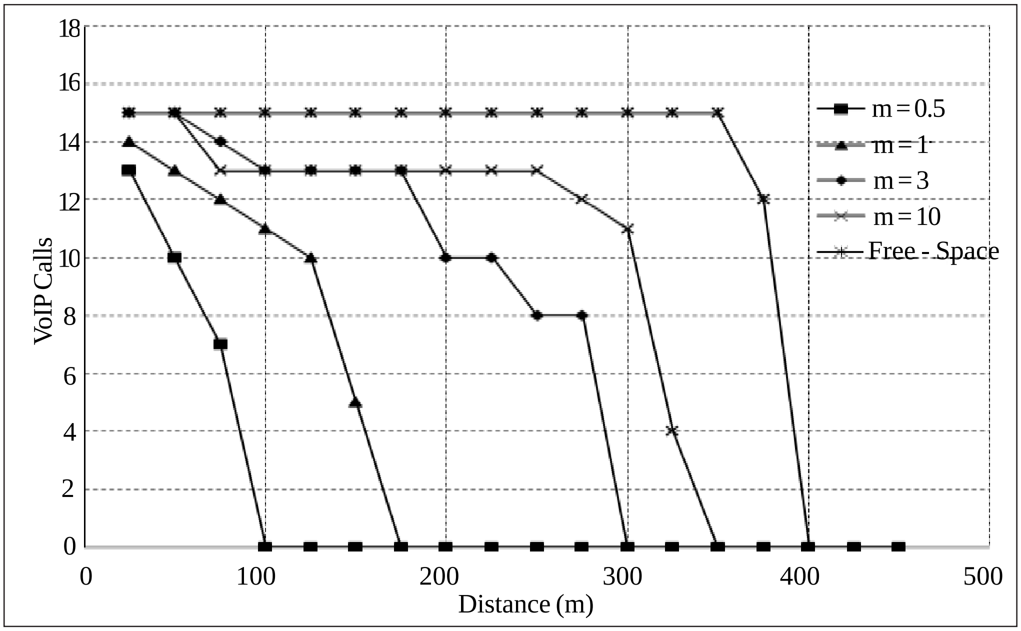
Table 1. 802.11g Parameters

4.1 Packetization of 10 ms

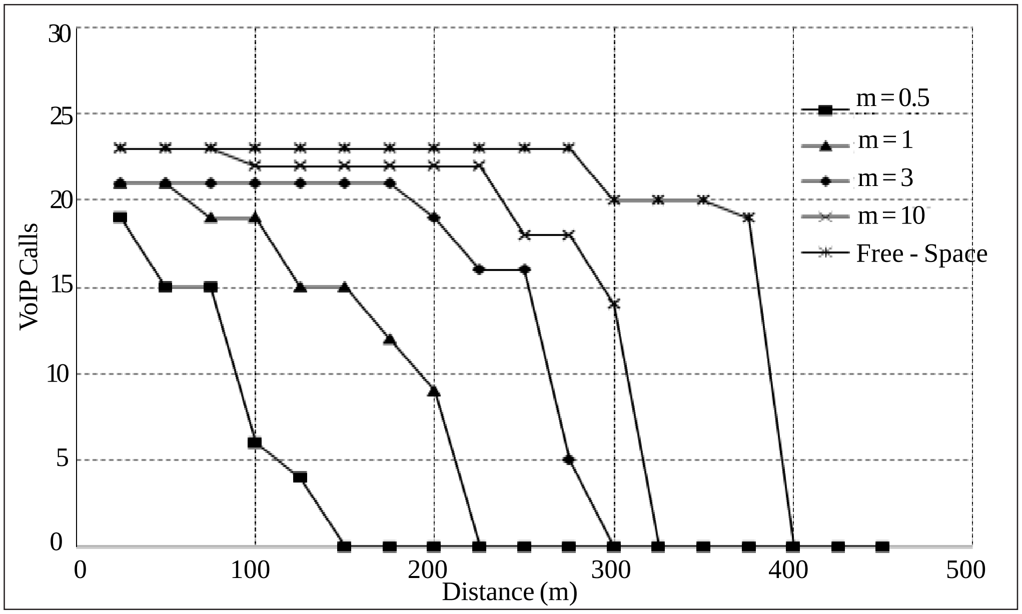
For the g711 A-law codec, Figure 2 (a and b) shows how the number of simultaneous valid VoIP calls supported by the network decreases as the distance between VoIP sources and the access point increases for transmission rates of 11 and 54 Mbps. Different system responses are noticed depending on the propagation model used and the value of shape factor *m*, which

R	User Satisfied	MOS
90 - 100	Very Satisfied	4.34 – 4.50
80 - 90	Satisfied	4.03 – 4.34
70 - 80	Some Users Dissatisfied	3.60 – 4.03
60 - 70	Many Users Dissatisfied	3.10 – 3.60
50 - 60	Nearly All Users Dissatisfied	2.58 – 3.10
0 - 50	Not Recommended	1 – 2.58

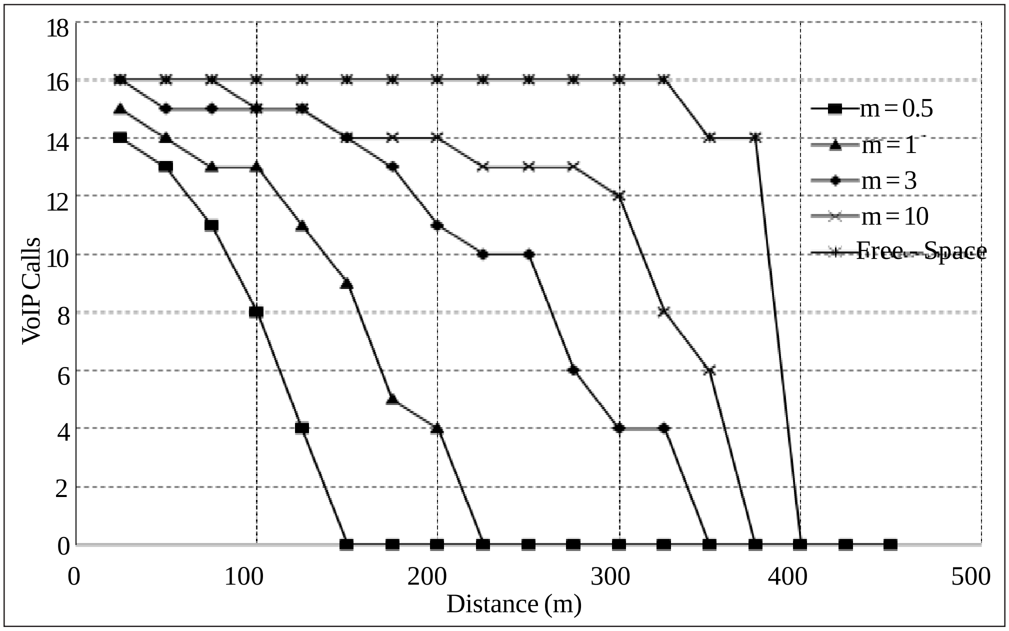
Table 2. Match Between, MOS, and User Satisfaction



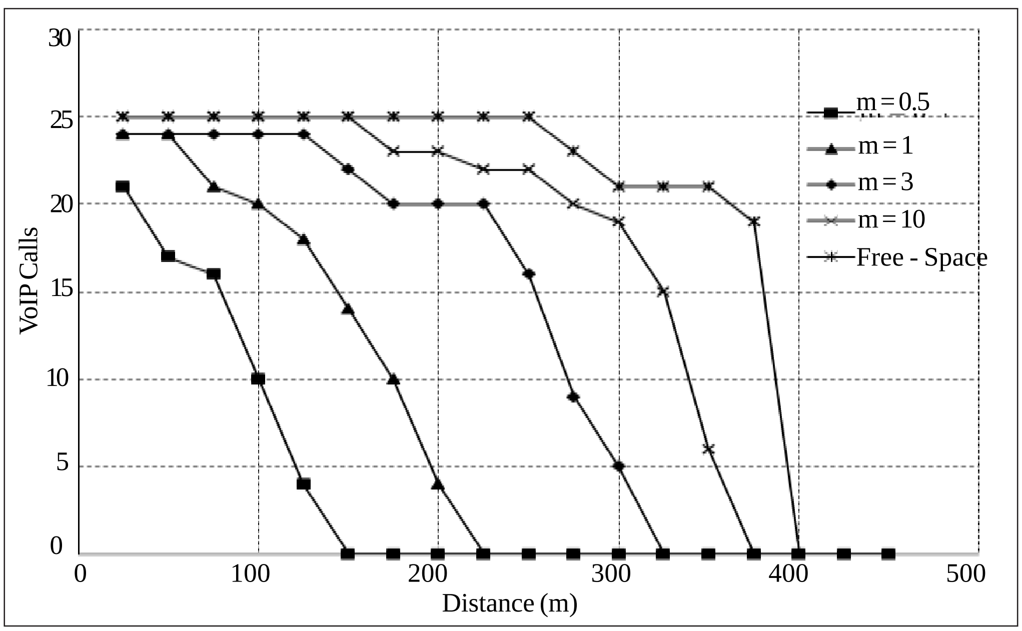
(a). Codec g711 A-law for 11 Mbps



(b) Codec g711 A-law for 54 Mbps



(c) Codec g726 (24 Kbps) for 11 Mbps



(d) Codec g726 (24 Kbps) for 54 Mbps

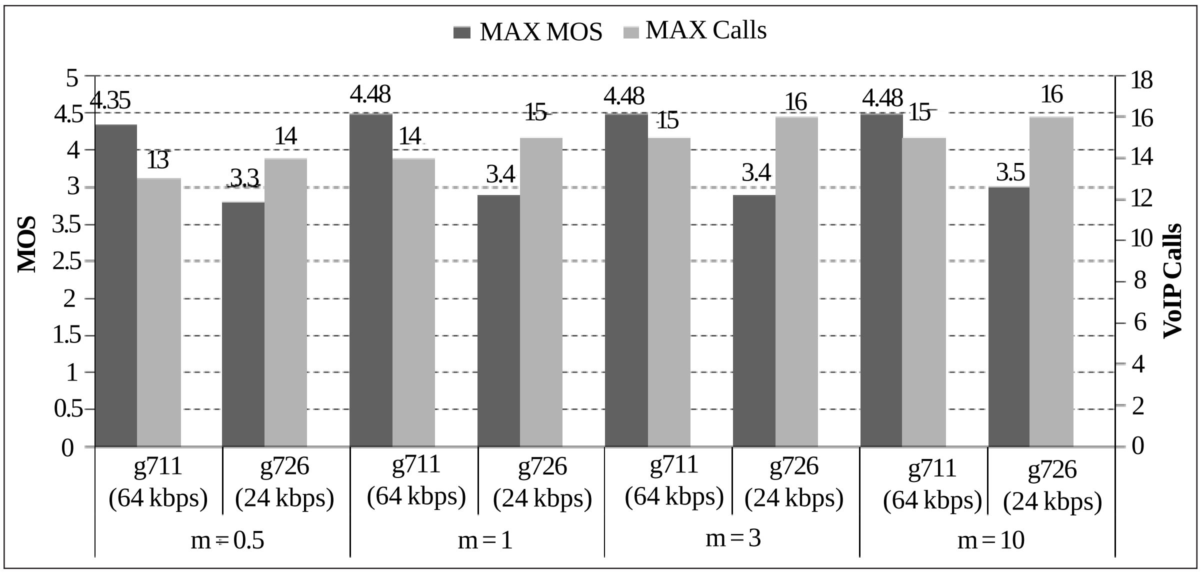
Figure 2. VoIP calls supported in the wireless IEEE 802.11g network and packetization of 10 ms

characterizes the Nakagami model as discussed above. In our simulations, *m* takes the following values: 0.5, 1, 3, and 10. As shown in Figure 2a, the Free Space model reaches the longest distance (375 m) maintaining the maximum number of valid calls, and suffers a very sudden fall from the maximum to zero in a shorter gap of distance. On the other hand, the Nakagami model shows a smoother fall on the capacity, being the work range (i.e., the range accepting valid VoIP calls) much lower than the one obtained for the Free Space model, specifically, 325, 275, 150, and 75 m when *m* takes values of 10, 3, 1, and 0.5, respectively. Moving wireless nodes to distances greater than those, no valid calls are supported by the system.

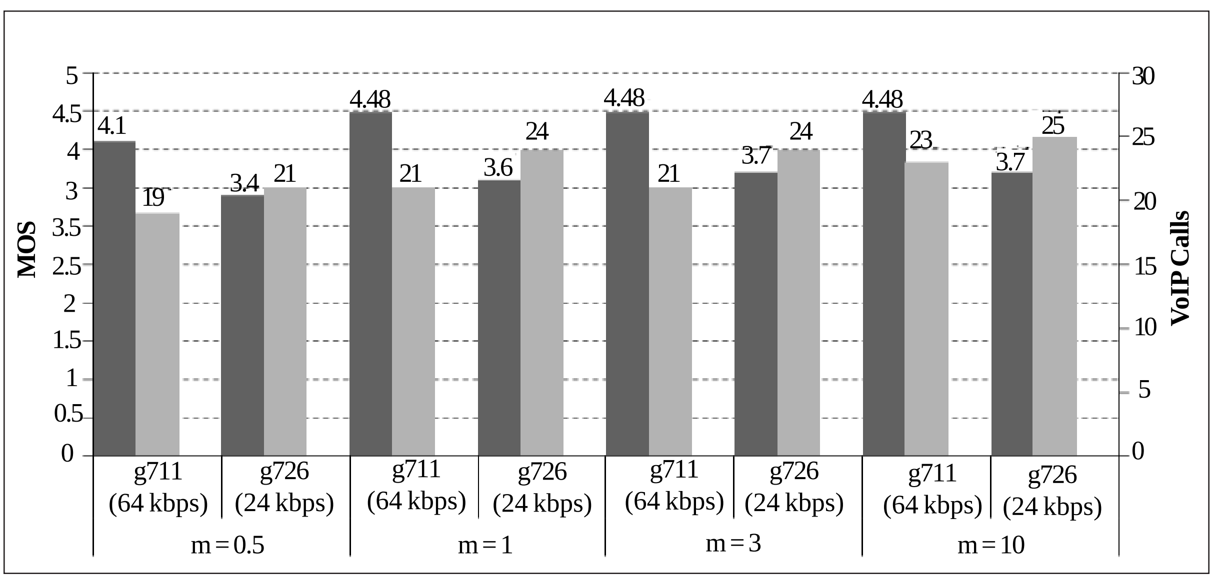
Observing the behavior for a transmission rate of 54 Mbps, Fig. 2b shows that the distance ranges with valid calls are longer for

54 Mbps than for 11 Mbps for scenarios with severe fading (125 and 200 m for *m* values of 0.5 and 1, respectively). This behavior is not noticeable when fading level is less
pronounced, i.e., in Free Space and Nakagami-m scenarios with greater values for *m*. In this case, the same work range is reached using both 11 and 54 Mbps. As expected, the gradual decline on the coverage range observed when decreasing *m* is related to the Nakagami-m fading channels. Lower values for *m* represent greater levels of fading, with a negative effect on the work range.

In addition to coverage range, the maximum VoIP capacity of the system is also affected by fading. We study this effect by analyzing the maximum capacity reached on each scenario in the shortest distance between wireless nodes and the access point, i.e., 25 m. Scenarios with *m* taking values of 0.5 or 1 do not reach the same system capacity attained in Free Space scenarios (or in those with greater values for *m*). All the simulated scenarios, with *m* equals to 0.5, give support to (at least) one call less than the other scenarios. Furthermore, the system capacity is not constant with distance, i.e., a progressive fall in the number of valid

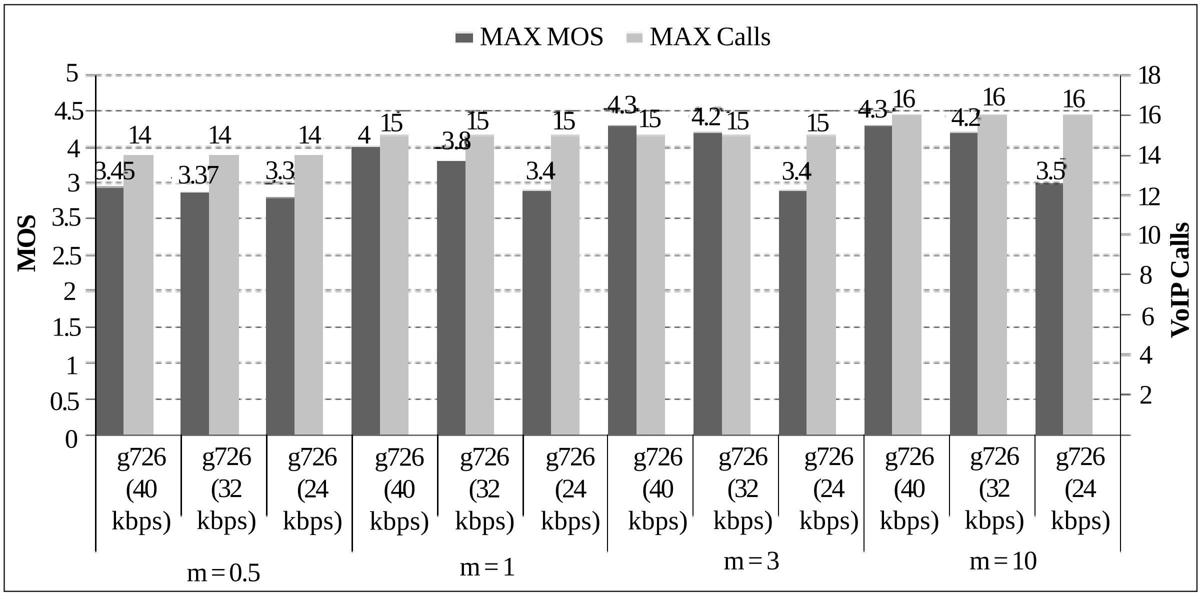


(a) 11 Mbps

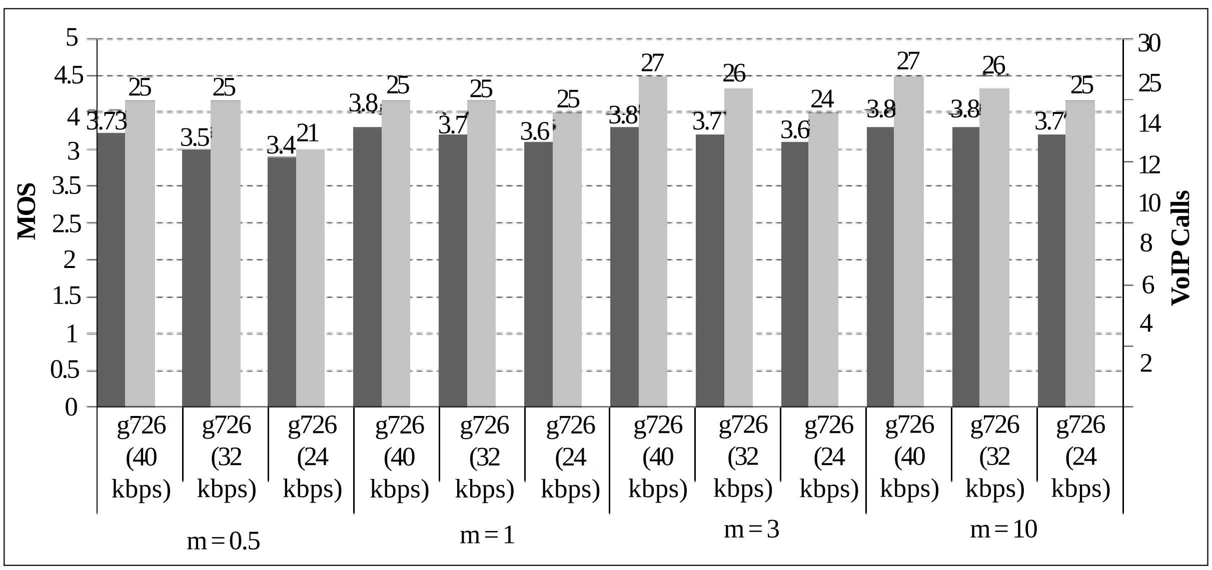


(b) 54 Mbps

Figure 3. Comparison of maximum VoIP capacity and MOS obtained in the VoIP calls with packetization of 10 ms and codecs g711 A-law and g726 (24 Kbps) for the different propagation models under study



(a) 11 Mbps



(b) 54 Mbps

Figure 4. Comparison of maximum VoIP capacity and MOS obtained in the VoIP calls with packetization of 10 ms and codec g726 at 24, 32, and 40 Kbps for the different scenarios under study

calls supported by the system is noticed when the distance between the access point and the wireless nodes increases. Figure 2 (c and d) shows the system response when using codec g726 at a coding rate of 24 Kbps for two different bandwidths of 11 and 54 Mbps. The same behavior as regards the system work range and capacity is observed. Therefore, no matter the codec used, fading affects the VoIP system in both aspects, (i) the decrease of maximum distance reached with valid calls, and (ii) the maximum number of valid calls accepted by the system.

Regarding the effect of the type of codec on the maximum system capacity, it is noticed a small increase using low bit-rate codec g726 compared to the results obtained for g711. As mentioned above, the maximum number of VoIP calls has been assessed taking into account the achieved MOS, the one-way delay, and the Ppl. Accepting a call as valid with a MOS limit of 3.1, we obtain the following results. For Free Space model at 54 Mbps, g711 (64 Kbps) achieves 23 valid calls against 25 calls for g726

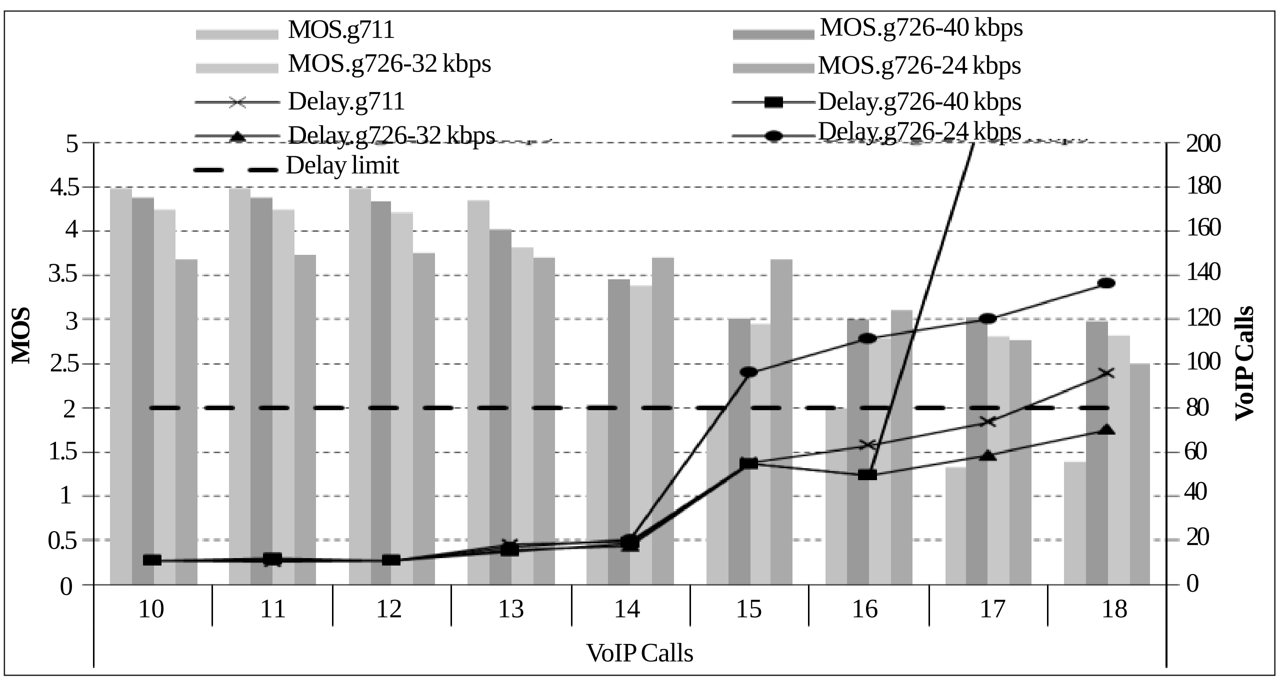
(24 Kbps). Using the most severe fading scenario (m = 0.5), only 19 and 21 calls, respectively, are accepted as valid as shown in Figure 2 (b and d). On the other hand, at 11 Mbps, 15 valid calls are achieved for g711 and 16 calls for g726 (24 Kbps), for no-fading Free Space scenario. However, in a Nakagami environment, with m = 0.5, 13 calls for g711 and 14 calls for g726 (24 Kbps) are achieved (see Figure 2 a and c). We could expect a greater difference between g711 and g726 (24 Kbps) in the number of valid calls accepted. The small increase in the capacity obtained using a low bit-rate codec is due to the legacy overheads and preambles introduced by 802.11g, which mask the effect of low bit-rate codecs.

Codec	**Tx Rate & Packetization interval**			
	11 *Mbps*		**54 *Mbps***	
	10 *ms*	**20 *ms***	**10 *ms***	**20 *ms***
g711	15	27	23	44
g726 (40 Kbps)	16	29	27	51
g726 (32 Kbps)	16	29	27	51
g726 (24 Kbps)	16	29	25	50

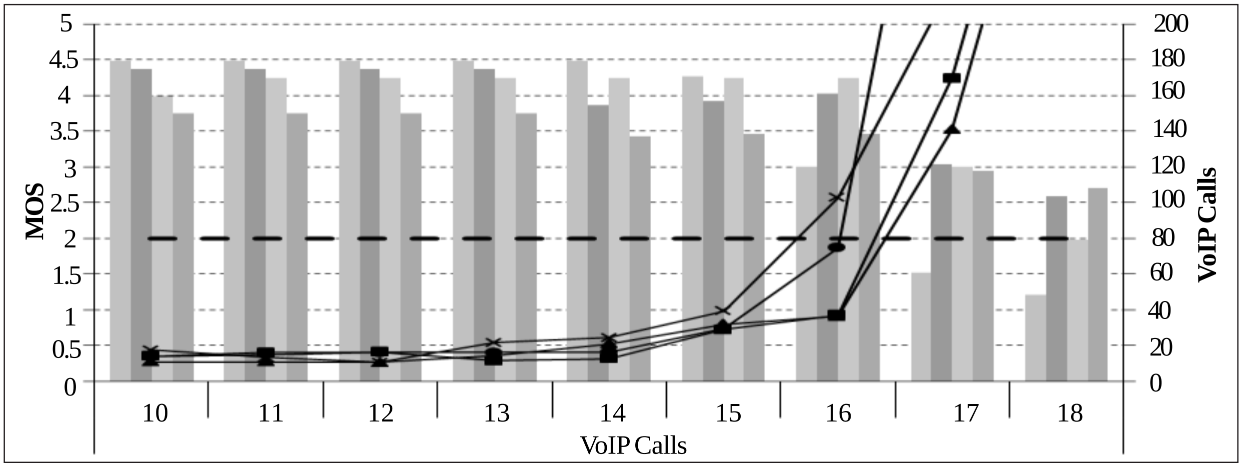
Table 3. Capacity of the System for the free space mobile obtained through simulation (In VoIP Calls)

Concerning quality in the communications, we study the effect of the type of codec and the fading channels over the MOS. The maximum MOS reached for each call decreases when low bit-rate codecs are used in comparison with the MOS obtained for g711. Focusing on the previously analyzed codecs, i.e., g711 and g726 with a coding rate of 24 Kbps, a comparison between the maximum number of VoIP calls accepted by the system and the maximum MOS value attained (for the different propagation models and transmission rates used) is shown in Figure 3. Although the capacity reached is greater using codec g726 (as discussed above), observe how the MOS attained in both cases shows a big difference; for instance, MOS of 4.35 for g711 and MOS of 3.3 for g726 (24 Kbps), at 11 Mbps, when m = 0.5 (see Figure 3a).

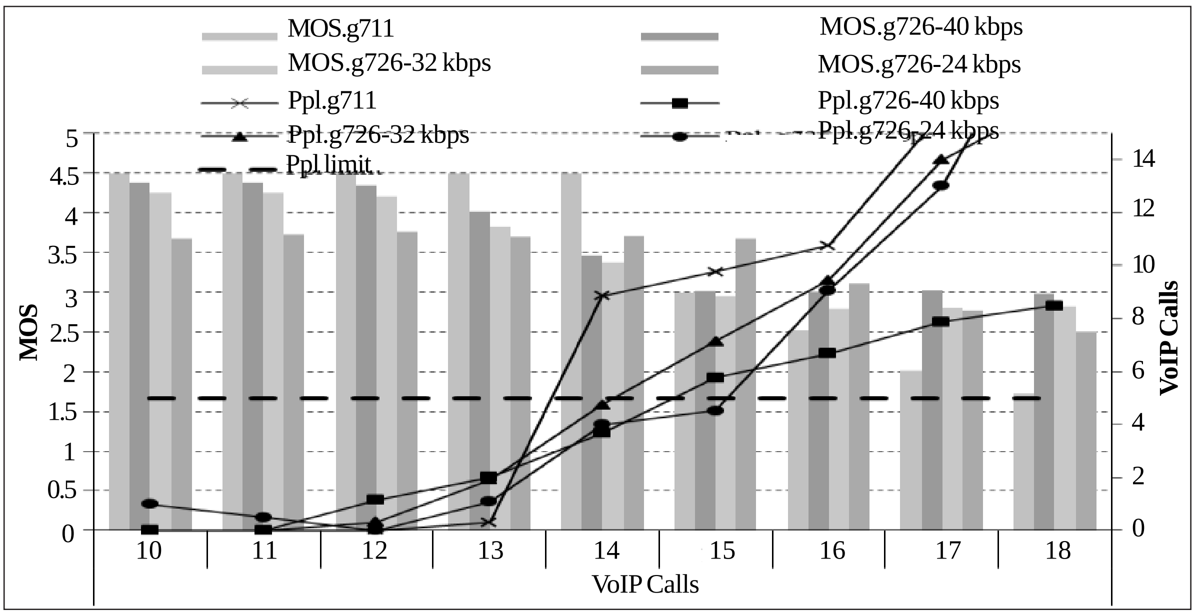
The quality of the VoIP calls is not only affected by the codec used, but also by the effect of fading channels. As shown in Figure



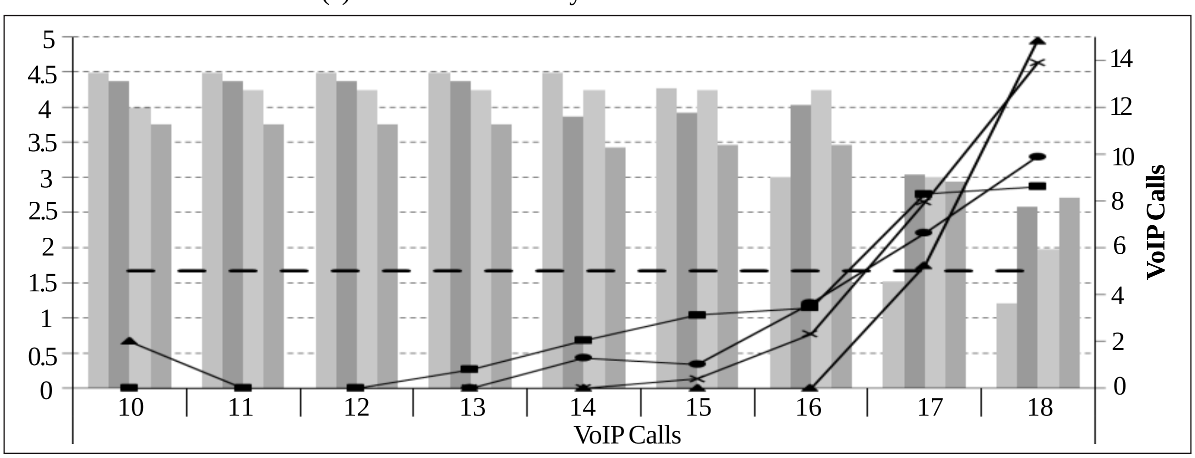
(a) MOS and delay with m = 0.5



(b) MOS and delay with Nakagami $m = 10$



(c) MOS and Probability of Packet Loss with $m = 0.5$



(d) MOS and Probability of Packet Loss with $m = 10$

Figure 5. MOS and delay measured in Nakagami-*m* scenarios for 11 Mbps and a packetization of 10 ms. For each group of bars, the bar sequence is as follows, g711, g726 (40 Kbps), g726 (32 Kbps), and g726 (24 Kbps)

3b, the maximum MOS value reached for codec g726 decreases with lower values of *m*, i.e., a higher level of fading. On the other hand, codec g711 shows a greater strength against fading, and almost the maximum MOS value of 4.5 is reached in all scenarios.

A comparison between the capacity and MOS reached for the three coding rates employed for codec g726 is shown in Fig. 4. The capacity does not exhibit a big increase regarding coding rates. It is worthy to mention the results obtained for 54 Mbps (Figure 4b). Observe how the attained capacity using a coding rate of 24 Kbps is lower than the ones obtained with the other coding rates. This is related to the low MOS value assessed for every call, which make us to discard active calls because their level of quality are lower than the MOS limit established, i.e., 3.1.

In order to study the effect of fading channels over the key network metrics affecting the communication quality, we analyze the one-way delay and packet loss rate. Figure 5 (a and b) depicts the evolution of MOS and delay as the number of calls accessing the system increases. We represent these metrics in the transition distance for each scenario, i.e., when the system capacity decay from its maximum to lower levels. As mentioned previously, we set the maximum one-way delay limit to 80 ms. Observe that, when delay exceeds that value, the MOS drops under 3.1, so VoIP calls are not accepted as valid ones (Figure 5a). The same behavior is detected by analyzing the Ppl. We set a Ppl limit of 5%, meaning that communications that suffer rates over this limit should not be accepted as valid. As it is shown in Figure 5 (c and d), when the Ppl exceeds that threshold, quality estimation fell below 3.1, so this number of calls are set as unacceptable. Therefore, a call can be set as invalid if just one of the parameters, delay or Ppl, reaches its threshold. For instance, let us analyze the case of codec g726 at 40 Kbps. When $m = 10$ at 11 Mbps, it reaches a maximum capacity of 16 calls. Observing Figure 5b, notice that the delay limit is overpassed with 17 calls on the system, rising from 36.94 ms (16 calls) to 169.7 ms (17 calls). A similar behavior holds for Ppl (Figure 5d), achieving a packet loss rate of 3.43% for 16 calls and 8.28% for 17 calls. In addition, the MOS value for 17 calls is 3.0, so that, this number of calls is taken as not supported by the system.

4.2 Packetization of 20 ms

In order to study the effect of fading channels on larger packets, a 20 ms packetization has been also employed in the simulations. Refering to the drop of the system capacity with the distance, the same behavior as for the lower packetization interval is noticed, i.e., scenarios with lower values of *m* support less number of valid calls and their coverage range decreases. In addition, due to the increase on the voice payload, an increment on the system capacity is observed in both, fading and non-fading scenarios. Table 3 shows the results obtained for the Free Space environment. Observe the difference of capacity attained between codec g711 and g726 using a packetization interval of 20 ms at 54 Mbps. In contrast with what happened using a packetization of 10 ms, a remarkable difference is obtained due to the legacy preambles and overheads loss weight in front of the voice payload.

Regarding the quality achieved for each codec used, we have obtained the same levels of quality as for the packetization of 10 ms, again, showing a decrease in scenarios with greater level of fading. Comparing the overall performance between the two packetization time-lenghts, the 20 ms packetization shows more profiteable results, because without decreasing the level of quality the capacity almost doubles compared with the 10 ms packetization.

5. Conclusion

We evaluated, via computer simulation, the effect of the physical layer on the performance of VoIP communications over 802.11g networks from a QoE perspective. To this end, we used the Omnet++ network simulator to measure the effect of fading introduced by Nakagami-*m* propagation model over differents codecs, packetization time-lengths and transmission rates. We compared the results obtained using this well-known model for various levels of fading with those attained with the Free Space model. We demonstrated that fading has a severe influence over the maximum system capacity, coverage range, and QoE of the VoIP calls. We also showed that low bit-rate codecs, e.g., g726, allow an increment on the capacity of the system and on the coverage range in comparison with those obtained using g711 A-law. On the other hand, the low bit-rate codec robustness to fading is lower than that showed by no-compression codec, revealed as a decrease on the quality (MOS) of the communications. Finally, the impact of using different packetization intervals was studied, obtaining advantageous results, in terms of capacity, for larger intervals. As a general conclusion, we have demonstrated that the effect of fading is a key metric that should not be ignored when designing or analyzing the performance of wireless VoIP systems. As future work, cross-layer techniques, taking into consideration the physical layer, will be employed in order to improve the QoE performance of VoIP on wireless scenarios.

6. Acknowledgment

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