8

Experimental Characterization of VoIP Traffic over IEEE 802.11 Wireless LANs

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1. Introduction

Voice over Internet Protocol (VoIP) technology has become a potential alternative to and also a supplement of the traditional telephony systems over the Public Switched Telephone Network (PSTN), providing a versatile, flexible and cost-effective solution to speech communications. VoIP allows the transmission of voice signals from one party to another one digitally, i.e., the analog voice signal is coded into small packets of digital data and sent over a network.

The traditional telephone network, PSTN, uses the circuit switching technique, in which the network establishes a dedicated end-to-end connection between two hosts. The resources needed to support the communication between these end systems are reserved for the whole duration of the communication, so as to guarantee a given quality of the communication. The main drawback of circuit switching is its lack of flexibility due to the fact that dedicated circuits are idle during silent periods and thus network resources are wasted during these contemplation periods. Unlike PSTN, VoIP networks use packet switching, which sends digitized voice data packets over the networks using many possible paths. The packets are reassembled at their destination to generate the voice signals. Network resources are not reserved in VoIP networks, i.e. voice packets are sent into the network without reserving any bandwidth. On the one hand this method provides more flexibility to the network but, on the other hand, it suffers from congestions. Voice applications are delay intolerant services, therefore voice quality at the end host is not guaranteed in VoIP networks.

Nowadays, the pervasiveness of WLAN networks together with the spread of VoIP capable wireless devices has motivated an extensive use of VoIP applications over WLAN networks. However, the interaction between these two technologies (VoIP and WLAN) is still not well understood and has received much attention from the research community during recent years. When the VoIP communication has to travel through a WLAN link congestion problems are hardened due to the shared nature of radio medium, the error prone channel and the limited bandwidth of the link, which can cause a further degradation of the voice quality.

This chapter focuses on the transmission of VoIP traffic over IEEE 802.11 WLAN networks and it takes an experimental approach to the topic. The objective of the chapter is double-fold. Firstly it aims at illustrating, from a practical perspective, the challenges that VoIP communications face when transmitted over WLAN networks. Secondly, it has the objective of providing experimental evidence of the requirements and practicality of some of the solutions that have been proposed in recent literature to optimize user experience in these scenarios.

Basically, the chapter illustrates the performance of VoIP technology over single-hop and multi-hop WLAN settings using the EXTREME Testbed® experimentation platform (Portolés-Comeras et al., 2006). Throughout the text we show how there exists a fundamental relation between the quality of VoIP calls and the capacity of WLAN networks. Even more, the experimental results show that this relation is difficult to handle in practice as this relation suffers from a sudden ‘breakdown’ by which when the number of calls exceeds a certain volume, most of communications sharing the WLAN network are severely degraded.

Beyond these results the chapter provides elements and hints on how to deal with experimentation settings to study VoIP over WLANs, and validates some state-of-art results and hypotheses from a practical perspective.

The structure of the chapter is as follows. First the background on VoIP networks and IEEE 802.11 WLAN is presented. Then, the problem of transmitting VoIP traffic over WLAN is stated in the two typical WLAN topologies: single-hop and multi-hop, also listing the literature on such topics. Section 4 describes the methodology used for analyzing the problem and the experimental framework is presented. After that, the experimental results are reported in Section 5. Finally, conclusions are drawn in Section 6.

2. Background

2.1 Voice over IP networks

Commonly voice over IP (VoIP) networks allows phone to phone, PC to PC and PC to phone communications through an IP backhaul as depicted in Figure 1. PC to PC is a call in which one PC communicates with another PC. In phone to phone an IP phone communicates with an analog phone. PC to phone is a type of communication in which a PC communicates with an analog phone.

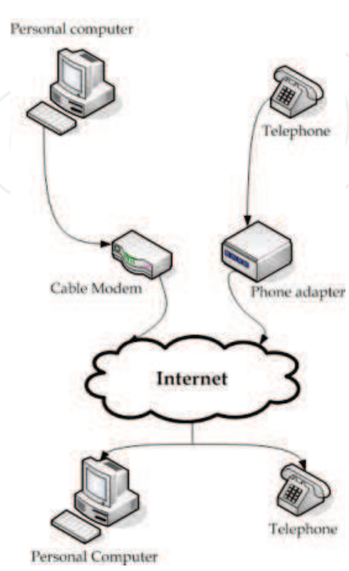


Fig. 1. VoIP network

In the user plane the communication takes place as follows.
At the sender, the voice stream from the voice source is first digitized and compressed by the encoder. Then, several coded speech frames are packetized to form the payload part of a RTP packet. The headers (e.g. IP/UDP/RTP) are added to the payload to compose the packet which is sent to IP networks. The packet may suffer different network impairments (e.g. packet loss, delay and jitter) in IP networks. At the receiver, packet headers are stripped off and speech frames are extracted from the payload by the depacketizer. Playout buffer compensates network jitter at the cost of further delay (buffer delay) and loss (late arrival loss). The de-jittered speech frames are decoded to recover speech with lost frames concealed (e.g. using interpolation) from previous received speech frames. For a better comprehension, Figure 2 reports the different block diagrams involved in a communication in a VoIP system.

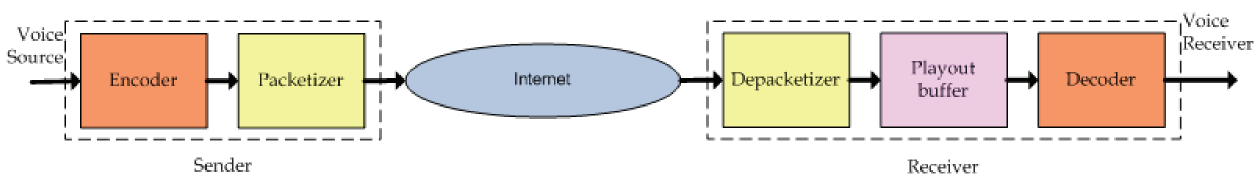


Fig. 2. VoIP System block diagram

As far as the control plane of a VoIP system is concerned, it has to perform the following tasks:
1. Find out the destination (e.g. IP address)
2. Establish the communication with that party and negotiate the parameters needed for the correct use of the session by the involved parties.
3. Potentially send quality reports to the sender to improve the quality of the communication (e.g., using RTCP).

Normally VoIP networks utilize H.323 [H.323] or SIP [RFC-3261] to perform the signaling functions described above.

2.2 Voice quality assessment

One of the most important metrics in VoIP systems is the speech quality experienced by the end users, also referred to as Quality of Experience (QoE) in the following sections.
Voice quality assessment methods can be classified in two main classes: subjective and objective methods.
The most known subjective method is the Mean Opinion Score (MOS). MOS value is obtained as an average opinion of the voice perceived quality based on asking people the grade of the service they received on a five-point scale, i.e. Excellent, Good, Fair, Poor, Bad. This method is internationally accepted and recommended by ITU [P.800]. Nevertheless it suffers from several problems such as highly time consuming, expensiveness, lack of repeatability.
Objective methods can be intrusive or non-intrusive. A typical intrusive method is the Perceptual Evaluation of the Speech Quality Measurement Algorithm (PESQ) based on the P.862 ITU-T standard. It is based on the comparison of a reference and the degraded speech signals to obtain a predicted one way MOS score. Such method is of course accurate, but it is unsuitable for monitoring live traffic because of the need of reference data to generate the reference signal. On the other hand, non-intrusive methods do not need of a reference signal

and seem more appropriate to monitor live traffic. E-Model [G.107] is the most used non-intrusive method. It determines voice quality directly from IP network and terminal parameters.

In our studies we adopted such method to assess perceived voice quality. Next subsection gives an overview of the main characteristics of such method.

2.2.1 E-Model

The European Telecommunications Standards Institute (ETSI) Computation Model (abbreviated in E-Model) has been developed by ETSI in the framework of the ETR 250 and it is used to predict the voice quality non-intrusively for VoIP applications.

The primary output from the E-Model is the "Rating Factor" R, which represents a measure of the perceived quality by the user during a VoIP call. According to ITU-T Recommendation 03/2005, the E-Model defines a region of acceptable quality when R is higher than 70. In Table 1 is reported the relationship between values of R and MOS.

R-value	MOS	User Satisfaction
90	4.34	Very satisfied
80	4.03	Satisfied
70	3.60	Some users dissatisfied
60	3.10	Many users dissatisfied
50	2.58	Nearly all users dissatisfied

Table 1. Relationship between R-factor and MOS values

The value of R is calculated as follows:

$$R = R_0 - I_S - I_d - I_e + A \tag{1}$$

where R_0 represents the basic signal-to-noise ratio, including noise sources such as noise and room noise. I_S is a combination of all impairments which occur simultaneously with the voice signal (e.g. quantization noise, received speech level and sidetone level). I_d represents the impairments that are delayed with respect to speech (e.g. talker/listener echo and absolute delay). I_e represents the effects of special equipments or equipment impairments (e.g. codecs, packet losses and jitter). A is an advantage factor (e.g. it is 0 for wireline and 10 for GSM).

The present chapter, as explained in the introduction, aims at analyzing how well an IEEE 802.11 WLAN can support VoIP service. Therefore our interest is on the impairments due to codec, delay, losses and jitter introduced by the WLAN, for that reason we can consider the simplified formula below to calculate the R-factor.

$$R = 93.2 - I_d - I_e - I_C \tag{2}$$

with

$$I_d = 25\left\{(1-X^6)^{1/6} - 3(1+\left[\frac{X}{3}\right]^6)^{1/6} + 2\right\} \tag{3}$$

where $X = \log_2 \frac{delay(ms)}{100\ ms}$

I_c, codec impairment

and

$$I_e = I_C + (95 - I_C) \cdot \frac{P_{Pl}}{P_{Pl}/(R_{Burst}+B_{Pl})} \tag{4}$$

where
- P_{Pl}: packet loss probability
- B_{Pl}: packet loss robustness (codec dependant)
- R_{Burst}: models the burstiness of the losses

by assuming that impairments are mainly due to network conditions.

2.3 Background on WLAN

This section will describe the basic mechanisms of the Distributed Coordination Function (DCF) used in IEEE 802.11 WLAN standard. For a detail description, please refer to [ANSI/IEEE Std, 1999].

DCF uses carrier sense multiple access with collision avoidance (CSMA/CA) as medium access protocol. The CSMA/CA protocol is designed to reduce the collision probability between multiple STAs accessing a medium, at the point where collisions would most likely occur. It also assures equal access priority to all stations competing for the radio resource in a given period of time.

In Figure 3 is illustrated a typical scenario when a transmitter and a receiver communicate using CSMA/CA:
- transmitting station has to sense the medium idle for a certain time, called DIFS, before sending its data ;
- receiving station acknowledges the reception after waiting for a different period of time, called SIFS, if the packet was received correctly (CRC is used to check the correctness of the received frames);
- automatic retransmission of data packets is used in case of transmission errors.

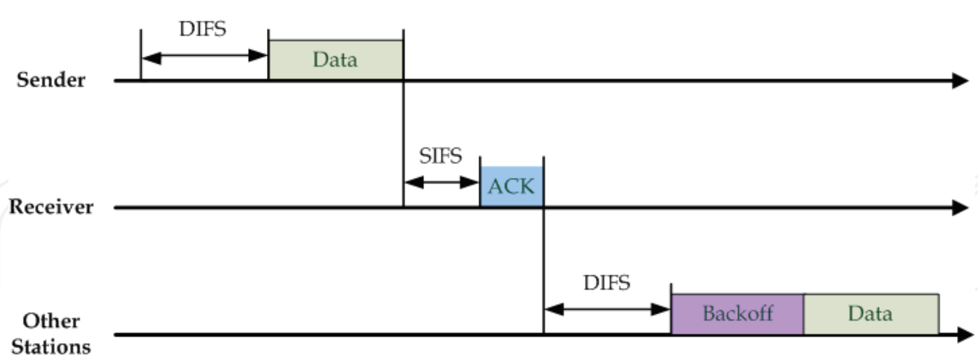


Fig. 3. Sending unicast packets using CSMA/CA

As it can be seen from Figure 3, different inter frame spaces (IFS) are used to provide the priority levels for access to the wireless media. In DCF, two different IFSs are defined; they are listed in the following from the shortest to the longest:
- Short Inter-Frame Space (SIFS), used for an ACK frame;
- DCF Inter-Frame Space (DIFS), used before to transmit data frames and management frames.

The entire process of transmission is as follows.

A STA, desiring to initiate transfer of data, senses the medium to determine the busy/idle state of the medium. As different STAs can sense idle medium, they could simultaneously access to the channel, generating a collision. Therefore, a backoff procedure is implemented in DCF. Before starting a transmission, each node performs a backoff procedure, with the backoff timer uniformly chosen from [0;*CW*] in terms of time slots, where *CW* is the current contention window. If the channel is determined to be idle for a backoff slot, the backoff timer is decreased by one. Otherwise, it is suspended. When the backoff timer reaches zero, the node transmits a data packet. If the receiver successfully receives the packet, it acknowledges the packet by sending an acknowledgment (ACK) after a SIFS. If no acknowledgment is received within a specified period, the packet is considered lost; in this case the transmitter will double the size of *CW*. After attending a DIFS, every station shall generate a random backoff period for an additional deferral time before transmitting and choose a new backoff timer, and start the above process again. When the transmission of a packet fails for a maximum number of times, the packet is dropped.
Figure 4 presents the various step of the previous procedure.

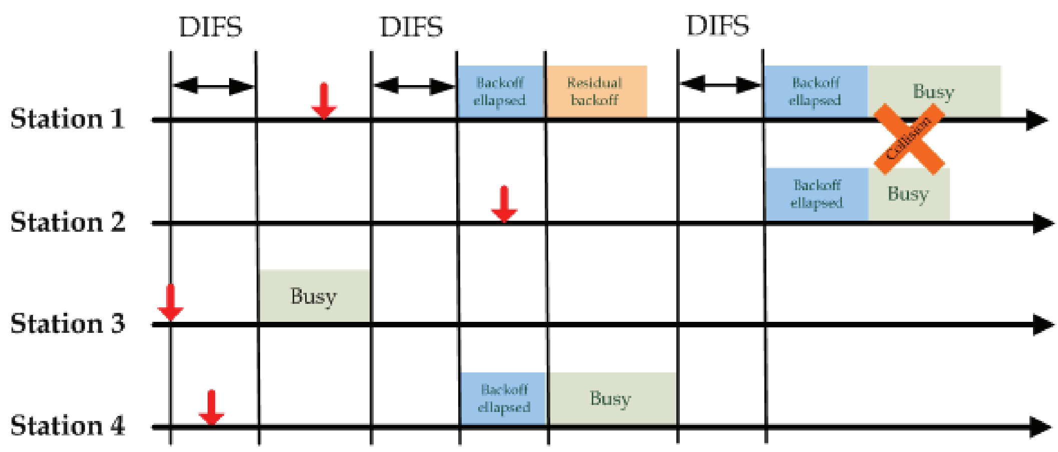


Fig. 4. Backoff procedure using CSMA/CA

According to the IEEE 802.11 standard, the network can be configured into two modes: infrastructure mode or ad hoc mode. In the infrastructure mode, an access point (AP) is needed to participate in the communication between any two nodes, whereas in the ad hoc mode, all nodes can directly communicate with each other without the participation of an AP.

3. VoIP over Wireless LAN

3.1 Introduction

In recent years there has been a growing interest in the use of WLANs based on IEEE 802.11 because of its low cost, simple deployment and free band access. As a matter of fact, nowadays the employ of WLAN in home, in offices and in public sites is a normal practice to provide customers with intranet and Internet access. In particular, WLANs are very popular among enterprises and universities to give workers and students greater flexibility to access the corporate/campus network without being tied to the network by a wire, so that they can access and contribute data far more quickly than before, boosting the productivity of all others who depend on critical information and, hence, increasing the overall agility of the organization.

Meanwhile also Voice over IP is gaining popularity as alternative to the traditional phone line thanks to its lower cost of infrastructure, easier integration of voice and data applications and lower call cost.
It is then easy to understand the increasing interest in studying the way the IEEE 802.11 can support VoIP services.
The main benefits of using VoIP over WLAN are basically three:

- WLAN can offer mobile voice services within a local area such as a campus or organization's facilities. In theory such service is already provided by cell phones, but many facilities experience cell interference or poor interior coverage, making cell phones unusable.
- extend VoIP services such as presence, push-to-talk, localization, conference calling, call transfer, do not disturb, on-line voice mail to WLAN environment
- convergence of data and voice in the same handheld device.

However VoIP applications are real-time services, then impose upper limits on delay and jitter in addition to usual requirements on packet loss and throughput to achieve an adequate quality. Ensuring the quality of service for VoIP in WLANs is a big concern as the performance characteristics of their physical and MAC layers is much worse than their wireline counterparts. In particular, lower peak transmission rate, lossy medium, interference problems, are some drawbacks of the physical layer. Moreover the DCF mechanism used by IEEE 802.11 WLANs, described in the previous section, can cause frame collisions during the communication, which is another issue to take into account in such environment.
The following two sections aim at giving an overview of the state-of-the-art on the VoIP over WLAN in the two typical WLAN topologies, namely Single-hop and Multi-hop topology.

3.2 VoIP over single hop WLAN

3.2.1 Background and state of the art

An extensive literature on 802.11 WLAN performance evaluation exists, though it is essentially based on theoretical studies for the characterization of the DCF in general conditions.
Stochastic Petri Nets are used in (Heindl & German, 2001) to model the behavior of DCF and then performance measures such as effective channel throughput are derived. In [Bianchi, 2000] the author used a Markov chain to model DCF and evaluate throughput and packet loss as a function of the number of wireless stations. This work does not take care of what type of protocol is above the MAC, hence the assumptions on traffic load do not correspond to load generated by VoIP traffic. In particular, the hypothesis of working in saturated condition is not correct for VoIP traffic as demonstrated in (Zhai et al., 2005).
Prior work that pertains to study VoIP over IEEE 802.11 has focused mainly on Point Coordination Function (PCF) as access protocol, for instance in (Crow et al., 1997) and in (Veeraraghavan et. al., 2001).
The first experimental studies at our knowledge on VoIP over WLAN is in (Garg & Kappes, 2003), where a study on the maximum number of VoIP users supported by a single WLAN access point is presented in a real scenario. They analyze basically how many VoIP users can an access point support, varying the codec. No reference to the QoE perceived by users is given.
Another interesting experimental work is in (Elaoud et al., 2005). The authors introduce an evaluation metric to quantify the performance of a voice call that takes into account both

packet losses and delay, thus providing a study on WLAN voice capacity with and without background traffic. Differences in performance evaluation due to the different VoIP codecs are not taken into account. No comparison with standard voice perceived quality assessment methods is presented.

In (Narbutt & Davis, 2006) the authors analyze experimentally the relationship between voice call quality (using the E-Model) and wireless resource usage by introducing three WLAN bandwidth components, namely load bandwidth, access bandwidth and free bandwidth. However an analysis on the effects of IEEE 802.11 PHY and MAC layer over VoIP quality is missed.

In (Zhai et al., 2006) authors claim that the optimal working point for a WLAN supporting VoIP traffic is when collision probability (p) is equal to 0.1. In that point throughput is maximum and packet delay and jitter are small enough to support voice quality requirements. They also introduce a metric called Channel Busyness Ratio (Rb) for measuring radio resource usage. They claim that in their working region ($p<0.1$), Rb is proportional to the throughput, so as concluding that channel busyness ratio is a suitable metric also to assess VoIP quality. No reference to standard voice quality assessment methods is provided in the paper as well as no measurement of packet loss is presented.

3.3 VoIP over Multi Hop WLAN

3.3.1 Background and state of the art

Studies around the use of VoIP in multi-hop WLAN scenarios aroused together with the popularity of Wireless Mesh Networks (WMNs). Even more, since the very beginning these studies rapidly became highly experimental (e.g. see Armenia, 2005). Among those seminal studies, the work by the authors of (Niculescu, 2006) constitutes an important reference for research studies that followed. The authors illustrate some of the main challenges associated with the transmission of VoIP over WMNs and analyze the performance of some practical solutions to cope with these challenges. Examples of this are the aggregation of flows, header compression, multi-interface configuration studies, and label-based forwarding of packets.

Beyond these initial efforts the literature has focused on some of the challenges that were identified since the beginning.

The study of the capacity of wireless networks received much attention in the beginnings of the decade after the seminal work by Gupta and Kumar (Gupta, 2000). This trend extended to studies related to WMNs and ultimately to studies of VoIP applications over wireless multihop networks. The findings show how the number of hops that flows traverse as well as the number of active nodes in a network affect the overall capacity of the system and, ultimately, the throughput of the applications that run over it. This has fostered the development of practical solutions that predict the utilization that VoIP calls represent to the capacity of a multihop network (see Kashyap, 2007) and used to implement effective call admission control mechanisms such as in (Wei,2006) and (Kashyap, 2007).

A method commonly used in order to make a more effective use of the bandwidth is that of packet aggregation (Niculescu, 2006). VoIP flows generally use constant sized packets to transmit information that make an inefficient use of the bandwidth resources (due to transmission overheads). Aggregation strategies try to mitigate these negative effects aggregating packets from different flows that share common paths of the wireless network. Examples of this are (Kim,2006) (Zhuang, 2006) and (Kassler, 2007).

Finally, also related to optimizing the utilization of bandwidth resources, some authors have worked in header compression solutions together with the aggregation strategy. Examples of this are (Niculescu, 2006) and (Nascimiento, 2008)
Besides optimizing the utilization of bandwidth resources other studies have also paid attention to analyzing which is the impact of the chosen routing strategy on the quality of VoIP calls (Ksentini, 2008) and have designed access strategies (MAC) that are VoIP aware and can help improve the quality of the calls (Yackoski, 2010).

4. Methodology

The work carried out in the framework of this chapter is based on the analysis of experimental tests performed within the laboratory of the IP Technologies area of the CTTC. This section aims at furnishing basic input to understand the operation of the test platform used for characterizing VoIP over WLAN.
The testbed is called EXTREME, which stands for EXperimental Testbed for Research Enabling Mobility Enhancements (EXTREME Testbed®). However, its flexibility has allowed broadening its scope from mobility to any networking scenario that could be of interest to both industrial and research communities. Particular emphasis has been put on those scenarios having a remarkable wireless component, including technologies such as UMTS/HSPA, and 802.11 WLAN.
The main architectural blocks of EXTREME are depicted in Figure 5 as well as its internal and external connections. The core of the EXTREME Testbed® lies on a Central Server. It is the interface between the experimenter/user and the Testbed, hiding its complexity and offering high-level experimentation services to the user. A serie of reconfigurable network nodes can be customized and used as network nodes for experimentation purposes (e.g. traffic emulation, routing, switching, acting as access points, wireless clients, capturing packets...). Each of these machines is connected to both a control network and a data

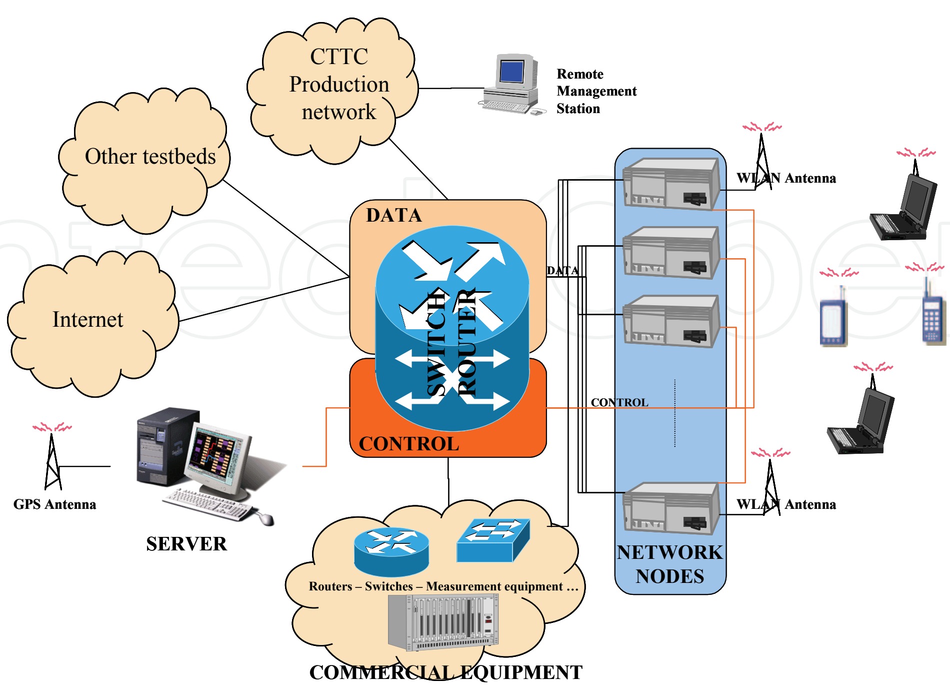


Fig. 5. Block diagram of EXTREME

network. Through the control network they interact with the central server for set up, configuration and data collection purposes (e.g. status, experiment results, etc.). The data network is used during experimentation to support communications. The interconnection pattern of nodes for each experiment is configured in the backbone switch-router.

Commercial equipment (traffic generators, networks emulators, and measurement equipment) is integrated in the testbed. External connectivity is offered through the CTTC production network and through connections to external production and research networks.

Regarding operating system (OS) and application software, for control and core development purposes mainly open source software is used, but any type of software can be integrated in the platform.

In the context of networking experimentation, an experimenter in the EXTREME Testbed® follows the following phases:

- Experiment design. The researcher defines the experiment at a high-level using some description files.
- Autoconfiguration software maps this high-level description into a physical topology.
- Experiment load and configuration. The autoconfiguration tools are in charge of controlling the node booting process, disk image loading, and execution of configuration files into the nodes.
- Execution of the experiment. The experiment execution control software is in charge of executing, at the chosen instant, the applications, in selected nodes, to follow the intended experimentation.
- Data collection for the EXTREME Measurement Architecture (EMMA) analysis, which can also be programmed using the execution control software.

The EXTREME Measurement Architecture (EMMA) provides a framework for the researcher to monitor the system under test while freeing him/her from the low-level details of the configuration of the traffic generation and capture tools.

A control server is at the core of EMMA, carrying out a series of functions:

- Interaction with the user. It provides a single point of interaction between the user and EMMA.
- Configuration of all transmit, receive, and monitoring (wired and wireless) nodes, i.e. the user is able to configure sources and destinations, flow characteristics, parameters to measure for both active and passive measurements.
- Scheduling of measurement events by the user.
- Gathering of traces and/or computations carried out at monitoring and/or end-nodes.
- Presentation of graphical results based on information gathered.
- Rapid evaluation and integration of new hardware and software monitoring tools.

EMMA allows multiple instances to be created, each monitoring a different experiment when more than one scenario/experiment are running concurrently in the reconfigurable testbed.

Active measurements, based on injecting synthetic traffic flows, serve to characterize paths segments, and potentially end-to-end paths.

Passive measurements, based on analyzing a copy of the traffic at a given point or observing certain variables without affecting real traffic, serve to characterize the traffic or other operational parameters at a particular point in the network.

The experimenter can specify how many flows are generated, source and destination of these flows, their characteristics, and where the monitoring machines are placed in the network, capturing and/or analyzing these flows.

5. Experimental results

5.1 VoIP over single hop WLAN

5.1.1 Introduction

The following section aims at explaining by showing numerical results achieved through experimental tests, how well an IEEE 802.11 WLAN in infrastructure mode (i.e. single hop) can support VoIP services. The analysis studies the effects of network congestion due to a high number of transmitting nodes and of wireless channel errors on the voice quality perceived by the user.

In the infrastructure mode, all nodes send and receive their traffic through an Access Point (AP). In section 2.3, IEEE 802.11 access protocol has been introduced. It is important to highlight here that the AP has no preferential access to the medium with respect to the other nodes. Taking this into account, as the number of nodes grows, different events which can damage the user QoE occur in the network:

- the number of contending stations rise, then causing increase in packet collision probability and consequently in packet error probability and packet delay because of the back-off procedure and packet retransmissions.
- the high number of packets to be delivered by the AP, i.e. downlink packets, suffer higher delay than in the uplink as well as possible losses due to AP transmission buffer overflow. For that reason, downlink is considered the bottleneck in VoIP over WLAN systems.

Moreover in a real scenario, the error prone nature of wireless channel (due to path loss, multipath and fading) can increase the packet error probability; also several WLANs can coexist in the same area, interfering each other and worsening the general network performance.

As introduced in section 2.2, several network metrics have a combined impact in the voice quality perceived by the users. In particular, the weight of packet delay, packet losses and codec impairment on the user QoE (modeled through the E-Model) is analyzed in the following subsections.

5.1.2 Voice quality performance on congested networks

This subsection intends to experimentally study the effect of network congestion due to a high number of nodes on the voice quality experienced by the users of such network. EXTREME testbed, detailed in section 4, has been configured for such experiments. The network under test has been built with ten nodes running Fedora 10 Linux OS with 2.6.17.11 kernel version. Each node is equipped with two Atheros based WLAN cards using MadWifi driver. Test traffic is generated using MGEN software [MGEN] and the synchronization of all the machines is achieved with high frequency NTP updates through the control network. This mechanism provides accuracies up to 200 μs in average and 400 μs as maximum value. One node acts as an AP and the other nine can act as a maximum of eighteen VoIP users, one per card. For each sender–receiver pair, packets are captured in both sides, so the end to end delay can be calculated and lost packets detected.

Table 2 resumes the traffic generated by each considered codec, namely G711, G723.1 and G729. A voice call is composed of one constant bit rate uplink flow and one constant bit rate downlink flow.

Codec	Bitrate (kbps)	Packet inter-arrival time (msec)	RTP Packet size (bytes)
G711	64	20	160
G723.1	6.3	30	24
G729	8	20	20

Table 2. Characteristics of the traffic generated by the voice codecs

The three codecs has been tested with the built network configured to work at 1Mbps and 2Mbps physical rate. Basically, for any given codec-data rate pair, tests with incremental number of voice users have been performed until reaching the voice capacity threshold for that pair. We refer to voice capacity as the maximum number of voice users the WLAN can support with R-factor higher than 70.

Results are shown in form of piled bar diagrams. The darkest bar represents the average R-factor experienced by all the users in downlink, which is considered the bottleneck of the system, as stated in the above. As mentioned in section 2.2.1, the highest this value can get is 93.2. Several impairments can produce the experienced R-factor to be lower than the maximum value, namely delay, codec and packet loss impairments. Codec impairment models the speech degradation due to compression in the voice codification and is represented by a white bar. Delay and packet loss effects are modelled in their respective impairments, represented in dark colour (delay) and light colour (packet loss). The impairments and R-factors are only plotted when they are higher than 0.

Figure 6, Figure 7 and Figure 8 show respectively the results obtained for the G711, G723.1 and G729 codec. In all the cases studied, the R-factor has a sudden breakdown in a given point: the input of only one user more generates an abrupt falling in the QoE of all the active users, regardless of the codec in use. This behaviour is typical in systems where no mechanism to guarantee quality of service (e.g., packet scheduler, congestion control) is implemented, as in the IEEE 802.11 standard. Another important result that can be noticed in the three figures is that the impairment due to packet losses is higher than the impairment due to packet delay at the breakdown, with the exception of the scenario of G711 codec at 1 Mbps. Such scenario can be considered a border line situation due to its very low number of nodes in the network that implies a very low number of contending stations and consequently a very low value of collision probability and packet error probability. Therefore, in general, it can be claimed that packet error delivery due to collisions and AP buffer overflow have a great impact on the performance of the network. Methods to control packet losses have to be introduced to allow WLAN to guarantee voice user QoE also in congested situations.

Finally, from the three figures, different values of the R-factor can be observed in a non-congested situation. These values strongly depend on the codec impairment. G723.1 has the highest and G711 the lowest in our set up.

5.1.3 Field trials

In this section the effect of the wireless channel errors is experimentally analyzed. Wireless channel can introduce errors in the network for two reasons basically:

- path loss, multipath and fading
- interference from other networks transmitting at the same carrier and residing in the neighbourhood

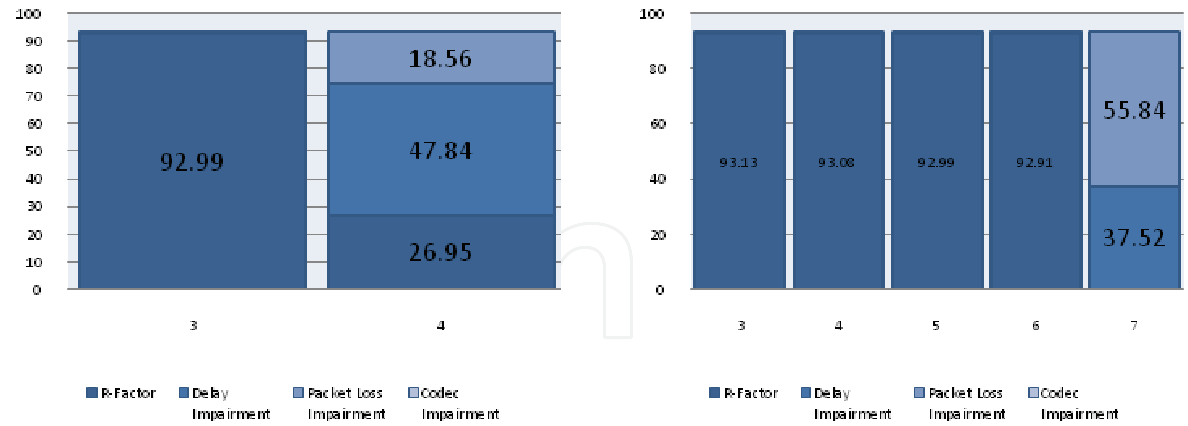


Fig. 6. R-Factor vs Number of users at 1 Mbps (left) and 2 Mbps (right) for G711 codec

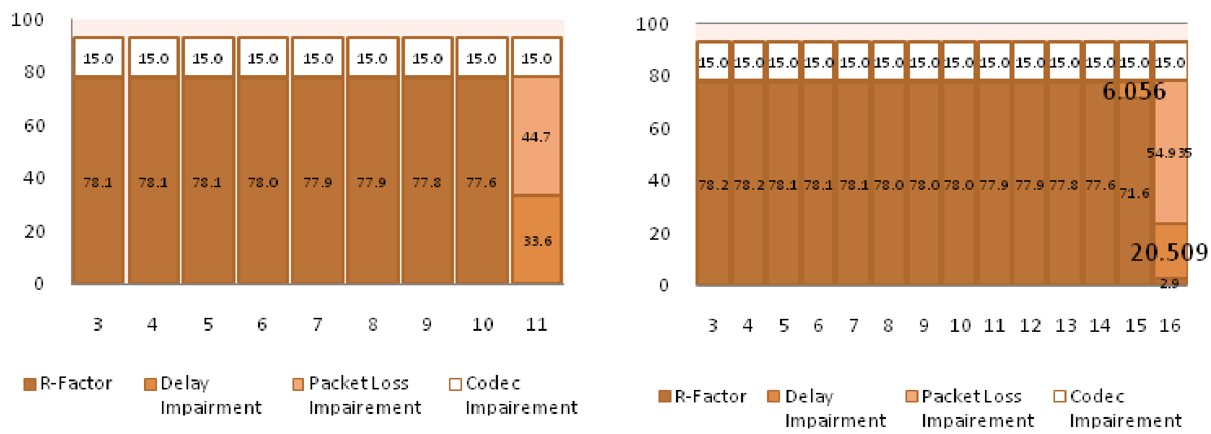


Fig. 7. R-Factor vs Number of users at 1 Mbps (left) and 2 Mbps (right) for G723.1 codec

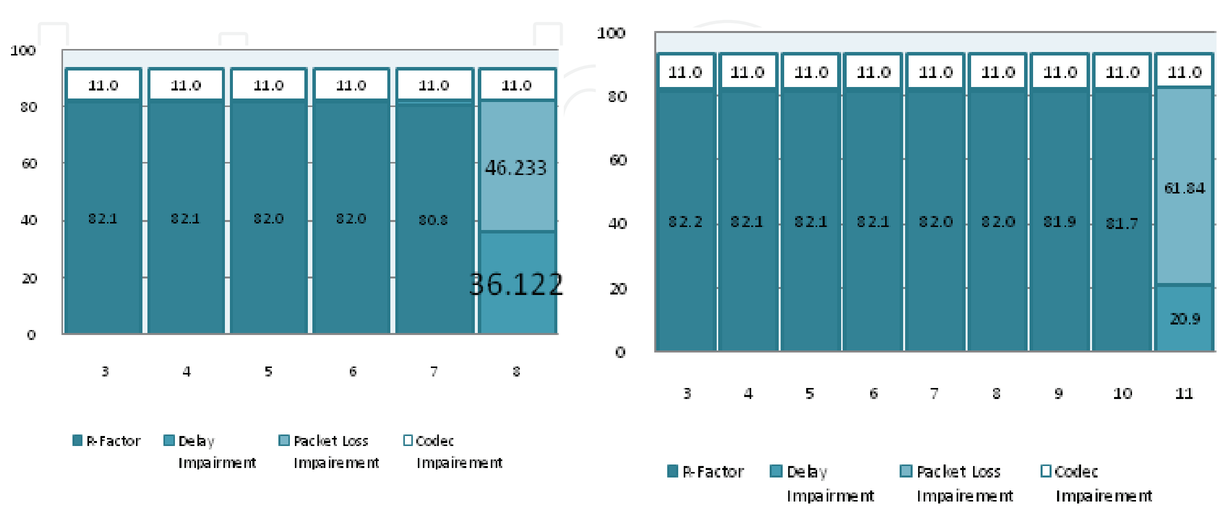


Fig. 8. R-Factor vs Number of users at 1 Mbps (left) and 2 Mbps (right) for G729 codec

For this scope the test platform has been modified. As a matter of fact, the results presented in section 5.1.1 have been measured in an in-lab controlled environment and using RF cables. Those cables isolate the system under test from external interferences and provide an almost ideal radio channel. Then, we decide to remove those cables and expose the test platform to the several production WLANs running in our campus, both in adjacent and overlapped channels and to a real radio channel. Only G711 codec has been tested, since in these trials we were not interested in studying the difference among different voice codecs. The tests have been run during several mornings, with no control on the interfering networks.

Results of the field trials are shown in Figure 9. Performance is reduced in general. Especially in 2Mbps scenario voice capacity is reached with three voice users less than in the case described in the previous subsection. Also, the R-factor breakdown is deeper than in the previous case: both delay and losses impairment increase in these trials due to wireless channel errors that produce retransmission and back-off procedure with higher frequency. It is important to highlight that even in the scenario considered in this subsection, loss impairment is the one with more weight on the quality experienced by the user. This result confirms the necessity to introduce methods to control network congestion in WLANs to guarantee user QoE. As an example we mention Call Admission Control, which is a successful scheme that has been used in cellular network to provide high quality voice services.

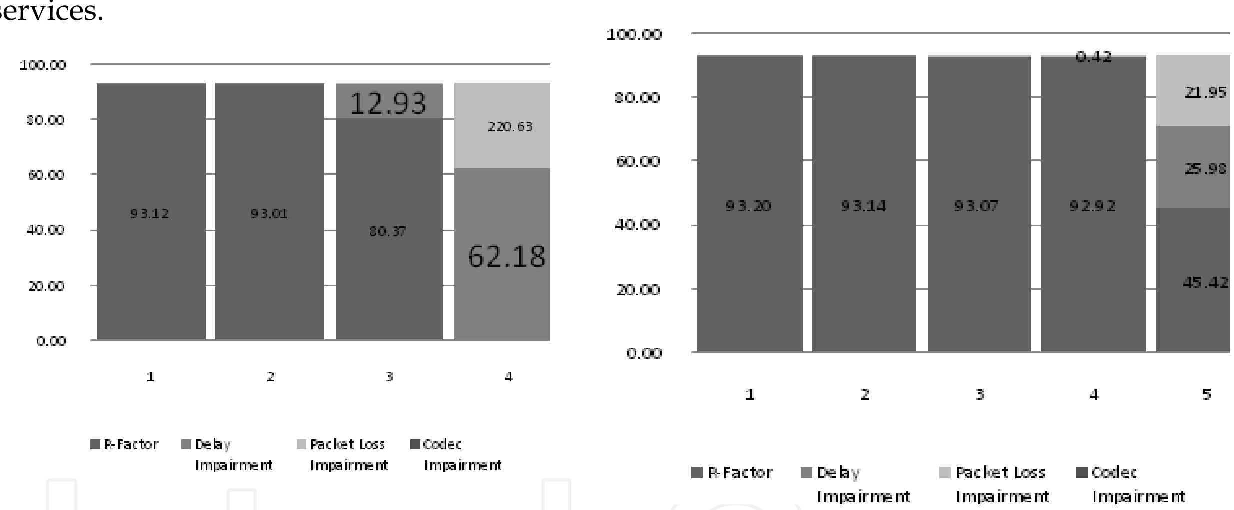


Fig. 9. R-Factor vs Number of users at 1 Mbps (left) and 2 Mbps (right) for G711 codec

5.2 VoIP over multi-hop WLAN

5.2.1 Scenario considered

As mentioned before, studies in wireless multihop networks highly depend on the scenario considered. This section proposes a target scenario for the study. The results obtained may be extended to several other scenarios but generalization is beyond the purpose of this paper.

Figure 10 presents the targeted scenario. A number of VoIP terminals collaborate in order to route individual VoIP calls between each one of the rest of terminals and peers residing in an external network. A wireless gateway is the one interconnecting the rest of the network and the terminals holding VoIP calls.

There are a number of considerations to be taken into account to completely define the scenario. Firstly, all VoIP communications are established between a wireless terminal and a node that resides somewhere in the backbone side of the network in the figure. Secondly, all terminals support establishing a single VoIP communication but are able to forward VoIP calls from other terminals. This can be thought as the communications infrastructure inside an office, where, in order to reduce costs, a single wireless gateway is installed and its coverage is extended through collaborative relaying. Third, the mobility of terminals is reduced. In general, users holding each one of the terminals remain in the same position for a reasonably long time. However, they might occasionally move, which leads to triggering a route re-discovery process. Fourth, VoIP terminals use the IEEE 802.11 WLAN protocol to form the multi-hop network. Finally, the mean duration of a call is considered to be of 2.6 minutes, as reported in (Lam, 1997).

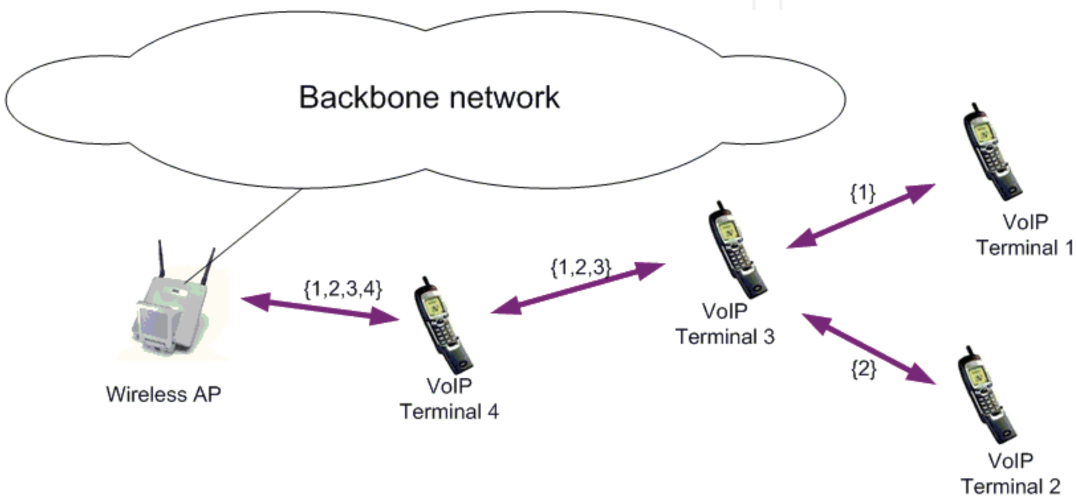


Fig. 10. Scenario: call relaying

The following sections present a study of the relation between VoIP quality and this multihop scenario. First, the section introduces the experimentation setup, then experimental data is used to show the relation between VoIP quality and number of hops and users. Finally, the section analyses the impact of state-of-art aggregation strategies on the quality of VoIP calls and the inter-relation between the disconnection time caused by route-rediscoveries and quality degradation from an end-user perspective.

5.2.2 Experimentation setup

All experiments have been carried out within the EXTREME framework described in section 4. All computers involved in this scenario are Pentium IV PCs with 512MB of RAM memory. They all run Linux operating system with Kernel 2.4.26.

Figure 11 shows the experimental setup used to obtain the results described in this section. Several nodes of the EXTREME cluster are each equipped with a PCI based wireless card carrying the popular Prism chipset (specifically, Z-COM ZDC XI-626 WLAN cards). All wireless cards are interconnected using coaxial wiring and propagation losses are emulated using RF attenuators, as depicted in the figure. This serves a double purpose. Firstly, it isolates the experimentation environment from external interferences. Secondly, it allows us to build a highly controlled environment where transmission range and carrier sense range can be adjusted in accordance to our needs for each one of the stations. Specifically, we adjust attenuations so that each one of the stations can only transmit to its neighboring ones but can 'sense' those stations that are two hops away.

When required wireless terminals start UDP traffic flows emulating VoIP traffic and with destination the VoIP sink in the figure. At the same time the VoIP sink starts an equal flow in the reverse direction completing the bidirectional VoIP communication. VoIP flows are emulated using the MGEN tool (MGEN). The reason to choose this application is double fold. On one side the traffic source can activate an option called "precise on" that efficiently controls real-time generation of packets. On the other side the traffic sink is able to store received packets in a format that allows convenient packet loss count and latency computation afterwards.

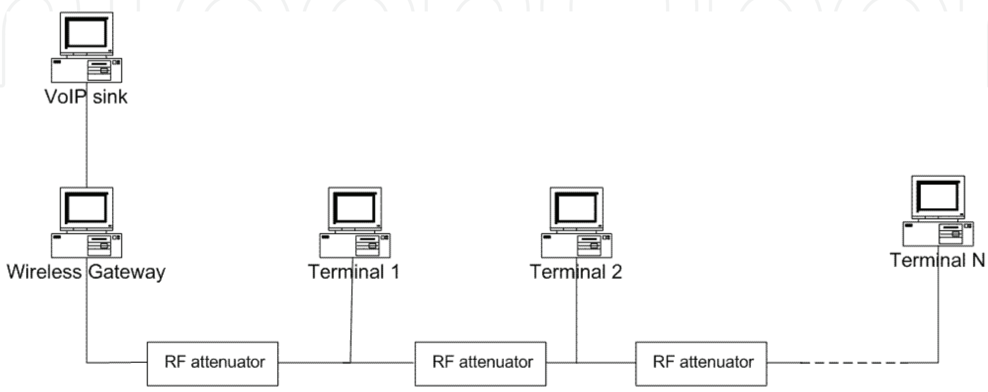


Fig. 11. Experimental Setup

5.2.3 VoIP capacity constraints due to number of hops and users

Since the seminal work by Gupta and Kumar (Gupta, 2000), the research community has put large efforts into optimizing the use of resources in wireless networks in order to optimize their utilization. The capacity of wireless networks highly depends on factors such as the density of wireless nodes and the distance between communicating nodes (e.g. the number of hops that a flow may traverse). Even more, when working with practical systems, frequency reuse distances as well as the applications that are being used play a very important role in order to make an efficient use of network resources.

This section illustrates, experimentally, the inter-dependence between the capacity of a wireless multihop network and various configuration options of WLAN devices in the presence of VoIP applications. The results give some insights into the effects of configuration options (such as carrier sense range, VoIP codec chosen, WLAN hardware used, number of active users, etc.) to the quality of VoIP calls supported.

Impact of the number of hops on a single-flow VoIP quality

Here we analyze the impact of the number of hops traversed on the quality of a VoIP call. Figure 12 shows the R-factor obtained when computing the E-model at the VoIP sink when the terminal it communicates with is located from 1 to 7 hops away. Note that, while it is a bidirectional communication, we do not print the curve obtained at the wireless terminal, as it is practically the same as the one obtained at the VoIP sink node. Note also, that all nodes are configured to transmit at 2Mbps physical rate.

This figure differs from the results reported in (Ganguly, 2006) as here a single VoIP flow can be sent over a larger number of hops. The main reason for this lies in the following observation. With the interference model adopted here (carrier sense range only reaches two hop distance) we assure that, at any time, any node only contends for channel access with at most four other terminals (those within carrier sense range), so its available channel

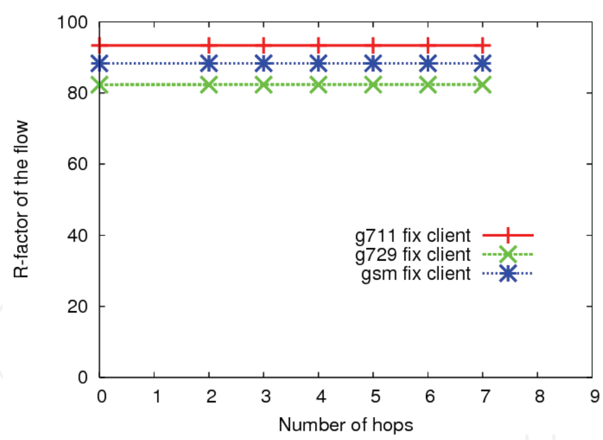


Fig. 12. R-factor vs number of hops traversed by a single flow as observed at the VoIP sink node

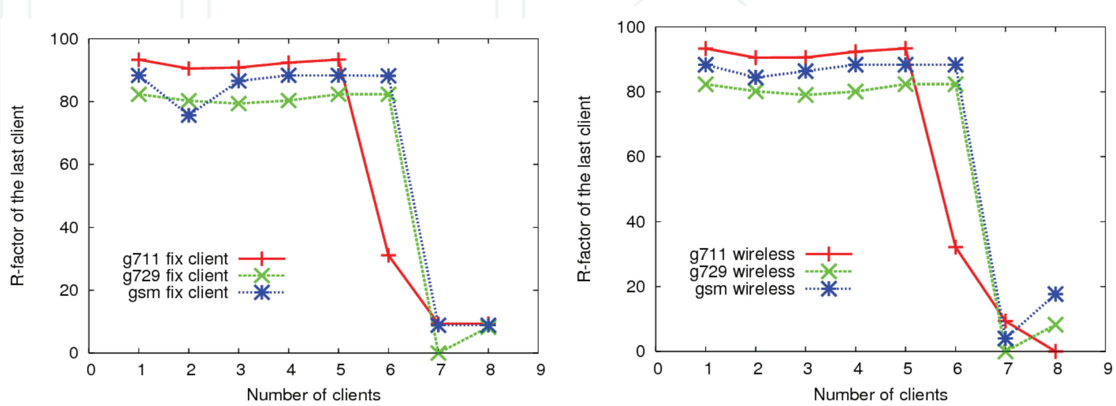
resources (bandwidth) will, at most, be divided by four (Jangeun, 2003). The resulting per node throughput capacity is constant and sufficient to support a VoIP call, no matter the number of hops packets have to traverse.

As a result, one can say that a conscious interference-aware deployment of a wireless multihop network can help, in the absence of other background traffic, increasing the maximum number of hops that a VoIP flow can traverse without suffering any quality degradation.

Impact of the number of hops on multi-flow VoIP quality

In this case we study the quality of voice conversations when each one of the terminals present in the network starts a VoIP call with the VoIP sink node. Going back to section 5.2.1, each one of the terminals communicates with the VoIP sink node via the neighboring node closest to the wireless gateway, which relays all VoIP messages in both directions. The idea is to study the maximum number of terminals supported in such a scenario. Note that this is a chain topology, so that there is only one route possible from each one of the terminals and the VoIP sink node.

Figure 13a and figure 13b, plot the quality of the VoIP call between the last terminal and the VoIP sink, at the VoIP sink node(a) and the wireless terminal(b) respectively. They plot the



(a) (b)

Fig. 13. R-factor of the last terminal call, as observed (a) at the VoIP sink and (b) at the wireless terminal for a different amount of active terminals

voice quality versus the total number of terminals (clients) maintaining a VoIP conversation with the VoIP sink node. Note that the number of terminals is equivalent to the number of hops traversed by the VoIP flows going to (and from) the last wireless terminal.
The figures show how, when using the G.711 codec, the system can only sustain up to 5 VoIP calls in the scenario described with an acceptable quality (R>70). This raises when using G.729 or GSM codecs, as the system can sustain up to 6 calls.
Two observations can be raised from the figures. Firstly, the breakdown previously reported in (Falconio, 2006) is shown experimentally. When the network saturates, the R-factor suffers a sudden breakdown preventing any communication between the last node and the VoIP sink. Secondly, when using different voice codecs, a different number of hops is reached. This suggests that a voice codec adaptation might be an efficient strategy to support a higher number of voice calls in saturated wireless multihop environments.
It is worth mentioning here, that not only communications with the last terminal in the chain but practically all the rest of the VoIP communications fall into unacceptable voice quality situation when the network enters saturation (i.e. when the last terminal starts VoIP transmission). This change is also abrupt, in a breakdown manner, which challenges the design of admission control mechanisms.

A side observation: Impact of the hardware used

Figure 14 compares the performance of two different card models when working in the multi-hop scenario. The figure shows how a multi-hop network built using Atheros based cards can support a lower number of nodes than when built using Prism cards. Such results can be explained using the results in (Portoles-Comeras, 2007), where it was shown that Atheros cards when detecting the absence of stations competing for the medium, does not perform backoff with the objective of increasing the throughput. However, in this case such a mechanism leads to having a more unbalanced multi-hop network that can sustain a lower number of VoIP calls.

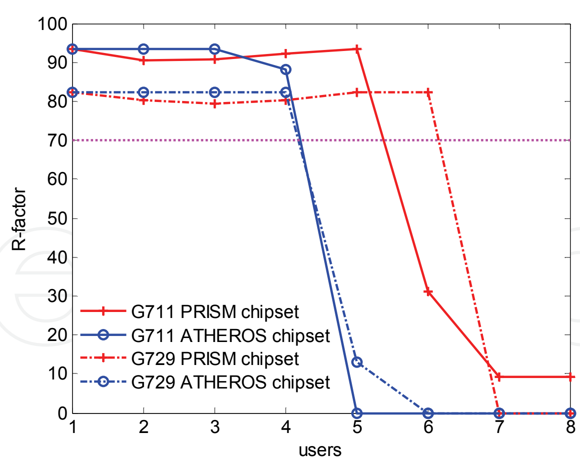
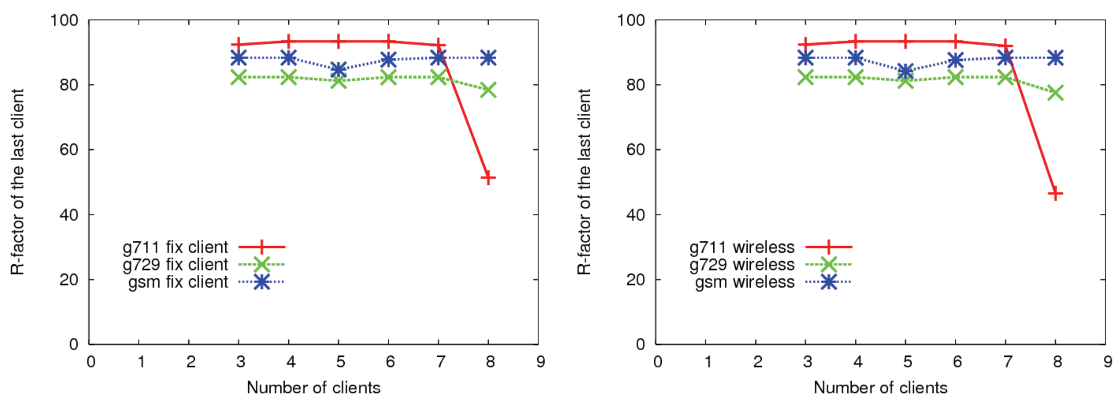


Fig. 14. Comparison of the number of VoIP calls supported in a multi-hop WLAN scenario when using the g711 and g729 codec

5.2.4 Benefits of aggregation: an illustrative example

In order to increase the number of VoIP calls supported in a mesh networking deployment, the authors in (Niculescu, 2006) propose using an aggregation strategy to be applied at the networking stack of each one of the wireless mesh nodes. While this strategy presents promising results, some considerations should be taken, regarding its application in our targeted scenario. Firstly, this solution implies modifying the networking stack of all the terminals to be used in order to include the proposed algorithm. This modification must be done at the OS level which increases complexity of the task. Secondly, considering that short buffering capacity is expected in the relaying terminals, no much forwarding opportunities might arise for packet aggregation.

Here we propose, as an alternative, doing the aggregation at the VoIP application itself. When the quality of the call being maintained is detected to have poor quality (e.g. through RTCP notification and run-time R-factor computation) the application can alternatively choose to aggregate various voice packets into one, prior to the send process. This process reduces the amount of resources required to keep the communication. The number of packets to be sent is reduced and this leads to reducing the amount of overhead to send them. This strategy has, however, an impact on the end-to-end delay of packets. In order to conduct aggregation some packets are delayed in purpose. However, as explained above, the end-to-end delay is not, generally, an issue in the targeted scenario, so there exists a margin of tolerance.



(a) (b)

Fig. 15. R-factor of the last terminal call, as observed (a) at the VoIP sink and (b) at the wireless terminal for a different amount of active terminals

Figure 15a and Figure 15b show similar plots as those in figure 13. In this case, however, stations are applying the aggregation strategy proposed. Each one of the terminals aggregates at the application layer two VoIP packets into one and sends them together to the next hop towards the VoIP sink node. The extra delay suffered by some packets due to the aggregation process is accounted for in the computation of the R-factor value. However one might notice that as the end-to-end delay is still low (<150ms) the R-factor value does not reflect any change. The figures show, however, how the aggregation effectively serves the purpose of supporting a higher number of active VoIP terminals in the network chain. These results suggest the possibility of including aggregation strategies at the application layer instead of the lower layers, as this extends the maximum number of terminals supported in our target scenario.

5.2.5 The impact of route-rediscovery latency on voice quality

The bursty loss resulting from the transient disconnection suffered by the VoIP terminal during a route re-discovery process, and the regency of the user after re-establishing regular communications, are factors to be considered in order to analyze the appropriateness of a route re-discovery process.

Figure 16 plots the R-factor value perceived by a VoIP versus the time elapsed since the route discovery disconnection finished[1]. This is plotted for various disconnection times, ranging from 200ms (typical in infrastructure based WLAN networks) and 5 seconds (a value considered well beyond acceptance for real time communications). Plotted curves show that when the disconnection time is below one second the user does not perceive unacceptable quality degradation. Even when the disconnection takes around 2 seconds the user 'forgets' about the disturbance at about 15 seconds after the VoIP communication is re-established. Note, however that this values do not account for mean end-to-end packet losses and delays that should be included for completeness in the curve.

Observing the curve one can notice that a long disconnection is preferable to several shorter frequent ones, as the user may rapidly forget about a single disconnection but would not tolerate frequent shorter ones. Once a protocol and route rediscovery have been designed, curves in this plot may serve to evaluate the possibility to support quality VoIP calls.

For completeness, figure 17 plots similar curves for the G.729 codec case. When using this codec the user is less tolerant to disconnection times and a maximum of 1 second occasional disconnections is tolerated after which it takes around 15 seconds for the user to 'forget' about the annoyance. These plots suggest again the use of codec adaptation strategies in order to adapt the communication to the network conditions. While G.729 might be more attractive in order to support a higher number of calls in a network, it is less recommended when the route re-discovery process incurs high latency.

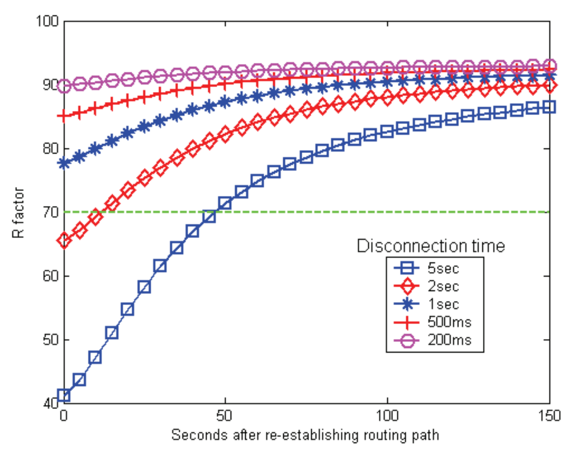


Fig. 16. Impact of route rediscovery disconnections on the quality of VoIP calls when using the G.711 codec

[1] Note that in order to introduce the time component in the calculation of the R-factor in figures 16 and 17, we have used the number of samples correctly received depending on the time elapsed since the route-rediscovery process. The E-model Standard definition does not include this way of using it but, currently, there does not exist any suitable subjective VoIP metric that includes time as an input parameter.

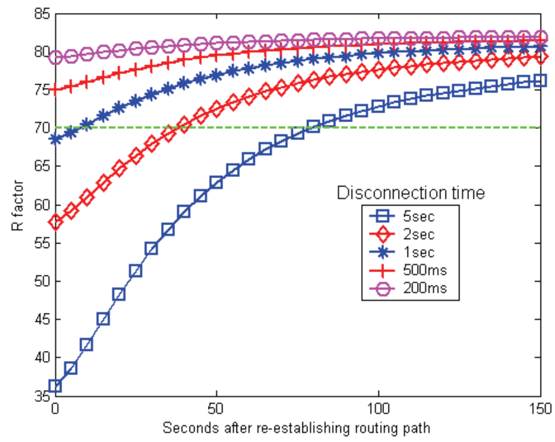


Fig. 17. Impact of route rediscovery disconnections on the quality of VoIP calls when using the G.729 codec

6. Conclusions

The main aim of this chapter has been to study the problem of the transmission of VoIP traffic over the IEEE 802.11 WLANs. The approach of the chapter has been practical and experimental: the challenge that VoIP communication faces when transmitted over WLAN networks has been pointed out by providing experimental evidence of the requirements and practicality of some of the solutions that have been proposed in the literature to optimize user experience in such environment.

Single-hop and multi-hop WLAN topologies have been experimentally studied using the EXTREME Testbed®. The objective of the experiments has been to show the relation between the quality of the voice calls and the capacity of the WLAN in terms of VoIP users supported with a acceptable quality (R>70).

In the single-hop scenario the effect of congestions and of channel errors on voice quality has been analyzed. The tests showed the decisive impact of packet losses due to collisions and to errors introduced by the wireless medium on the quality experienced by the user, which has been always higher than the impairment due to delay, regardless the codec used for the communication. This result supports the necessity of the introduction of methods to control congestions in WLANs. Recently, the "802.11e" standard has been introduced by IEEE to manage QoS in WLANs. Anyway, the new proposed access protocol is more oriented to the assignment of different priority to different types of traffic based on the service requirements than to the introduction of a congestion control mechanism. The definition of Call Admission Control schemes, as in cellular networks, is a valid alternative to guarantee VoIP quality in WLANs.

The results, gathered using the multi-hop set up, reveal that beyond the overhead that IEEE 802.11 WLAN protocol introduces, the number of users, the number of hops to traverse and also the specific deployment strategy (taking into account the carrier sense range) constitute determinant factors affecting the capacity of the network in terms of VoIP users supported.

The experimental results also show how a deployment strategy has to take into account the specific hardware used to support wireless communications, as this decision may also have effects on the VoIP quality perceived by end-users.

Using the multi-hop scenario, the chapter also shows how the aggregation of VoIP packets is a suitable strategy to reduce the impact of IEEE 802.11 overhead on the global network capacity as it can effectively increase the number of VoIP calls supported without penalizing the VoIP quality perceived by end users.

Finally, the multi-hop analysis presented introduces a methodology to determine the impact on VoIP quality of the route re-discovery process usually associated to wireless multi-hop deployments. Depending on the specific multi-hop scenario this process will occur with higher or lower frequency. The methodology introduced allows tuning any engineering as it provides some bounds on the maximum time route-rediscoveries can take.

7. Acknowledgement

This work has been partially funded by the Catalan Regional Government under grant 2009SGR-940.

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