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VoIP Over WLAN: What About the Presence of Radio Interference?

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1. Introduction

The use of Voice over IP (VoIP) is rapidly accelerating around the world and becoming familiar to an increasing number of people using Skype routinely (Douskalis, 1999). VoIP is also becoming more and more deployed through the so-called Voice over Wireless Local Area Network (VoWLAN) technology (Lin & Chlamtac, 2000), which integrates wired and wireless telephony in the same Internet Protocol (IP) structure, reducing the cost of calls and avoiding the typical problems of the highly variable coverage of the cell phone networks inside buildings. This vivacious scenario is giving to VoWLAN technology an increasing importance, entitled to become even greater in the future with the diffusion of new-kinds of portable devices (e.g. PDAs and social phones) and the availability of more and more Wi-Fi zones everywhere in the world.

In this chapter attention is focused on one of the most critical problems affecting VoWLAN operation, which, if not properly taken into account and controlled, may severely degrade the overall quality of service perceived by the final user. Such an important issue is radio interference in the wireless channel, which may affect the integrity of the signal received by a WLAN terminal and, consequently, cause misinterpretation of the carried digital information. The phenomenon is nowadays becoming more and more critical because of the increasing use of radio terminal equipment deploying the typical frequency band in which WLANs operate, i.e. the so-called unlicensed 2.4 GHz Industrial Scientific and Medical (ISM) band. In the related frequency range, in fact, IEEE 802.11 WLANs (informally known collectively as Wi-Fi) (IEEE 802.11, 1999) must coexist with IEEE 802.15.4 (IEEE 802.15.4, 2003) and IEEE 802.16 (IEEE 802.16, 2001) apparatuses. Moreover, they have to operate in the presence of unintentional spurious signals from electronic devices that either use this band, like cordless phones, microwave ovens, baby monitors, security cameras, or operate in adjacent frequency bands, like a number of wireless appliances whose distribution in modern houses, public and professional contexts is by now widespread.

Some authors tried to investigate on the effects of interference on voice quality in a VoWLAN conversation (Wang & Mellor, 2004; Wang & Li, 2005; Garg & Cappes, 2002; 2003; El-fishawy et al, 2007; Prasat, 1999; Hiraguri et al, 2002). For instance, in (Wang & Li, 2005) the coexistence of Transmission Control Protocol (TCP) and VoIP traffic in a WLAN has been studied in terms of delays and performance loss. In (Garg & Cappes, 2003), experimental studies have been shown on the throughput of IEEE 802.11b wireless networks for user diagram protocol (UDP) and VoIP traffic. In all these contributions, attention is essentially

focused only to interference at network/transport layer, due to the presence of competitive traffic in the same WLAN. Few information is instead typically available in terms of physical layer interference.
In this chapter, the performance of VoIP over WLAN is analyzed under the effect of physical layer interference, in the presence and absence of cross-traffic. The goal is twofold: first to underline the importance of radio interference in the behavior of a WLAN when supporting VoIP applications; second to outline solutions to avoid interference and thus optimizing a VoIP call over a WLAN. To this aim, an experimental approach based on cross-layer measurements is adopted (Angrisani & Vadursi, 2007), describing and commenting meaningful results obtained from a number of experiments conducted by the authors on a testbed operating in a semi-anechoic chamber and emulating two typical real life scenarios. In particular, different network architectures and voice codec typologies are emulated, such as G.711 (ITU-T G.711, 1972), G.729 (ITU-T G.729, 1996), G.723.1 (ITU-T G.723.1, 2006), usually utilized in VoIP applications over WLAN. Experiments are conducted according to a cross-layer approach and monitoring the following parameters: (i) signal to interference ratio (SIR) and jitter at physical layer, (ii) packet loss at network/transport layer, and (iii) mean opinion score (MOS) and R factor at application layer. For each investigated scenario, the presented outcomes will allow the reader to clearly identify and understand the origin of some typical interference phenomena on VoIP services over WLAN. They also allow to experimentally verify the effectiveness of practical and helpful rules, addressed in the chapter, for improving quality losses in a VoWLAN application in the presence of interference at physical and network/transport layer.

2. Preliminary notes

In this section, preliminary notes concerning VoIP and VoWLAN technology, IEEE 802.11 standard and voice quality metrics are introduced with the purpose of recalling some of the terms and parameters used in Sections 4 and 5.

2.1 VoIP

VoIP is a family of transmission technologies for the real-time delivery of voice calls over IP networks such as the Internet or other packet-switched networks. It is playing a fundamental role in the development and use of Internet in the world. It is also greatly contributing to the convergence of different technologies and applications over the same hardware infrastructures. The success of VoIP is especially due to the Internet itself, and in particular to its emerging use all over the world. Internet is in fact becoming a need of primary importance in an increasing number of countries. It is radically modifying styles and behaviors of people, communities and companies in their everyday relationships, activities and businesses. User mobility, real-time interaction, instant messaging, text paging, social networks, voice services, internet access during travels, multimedia exchanging, are only few examples of common needs and applications required by modern people, professionals and industries.
In a traditional VoIP call, terminals are connected through a local area network (LAN), made of cables, switches, hubs, and other similar apparatuses. This topology ensures efficient and reliable communication with strong immunity levels against radio interference; cables are in fact frequently covered by metallic shields and properly connected to the ground in order to avoid the influence of external perturbing radio interference. Nevertheless, many problems still arise, making the use of VoIP services not yet fully reliable. One problem can be attributed to the fact that voice calls require real-time procedures, which cannot fully

be satisfied in an IP-based context. In a IP network, in fact, two terminals are not linked through a physical circuit like in a public switched telecommunication network (PSTN). They instead communicate through a set of data packets, each of which containing a destination address and a fragment of the digitalized voice conversation. The addressed terminal collects the received packet, extracts the useful information, and reconstructs the original signal. This mechanism has to be completed without loss of packets or too long delays, so that to avoid failures in the real-time reconstruction procedure, and consequently artifacts in the voice conversation. Another problem is the use of a cabled infrastructure, which requires a non-negligible effort in terms of installation, reconfiguration and maintenance. In particular, an high number of cables are needed to connect a building, through walls and pipes in the walls and under ground floors or even roads. This means very high costs and long times to wire large areas and buildings. In the design of new buildings, LANs require to accurately predict all the possible needs of future users in such a way as to reduce as well as possible further modifications of the wired plant. This typically leads to an high risk of oversizing the whole infrastructure, and a consequent increase of costs. LANs are also a limiting infrastructure for voice applications; in particular, it obliges users to be physically connected to a personal computer, thus strongly limiting their mobility within the covered area.

2.2 From VoIP to VoWLAN

VoWLAN (Voice over WLAN) is a method of sending voice information in digital form over a wireless broadband network. It represents the conjunction of two important emerging technologies: VoIP and WLAN. In a VoWLAN call, terminals are connected to the Internet through a wireless link and an access point. It consists in the use of a wireless broadband network according to the IEEE 802.11 set of specifications for the purpose of vocal conversation (IEEE 802.11, 1999). VoWLAN is leading to an increasing importance and use of WLANs, which are rapidly wide spreading everywhere in the world, through an increasing number of public and private hot-spots located in public areas, university campuses, factories, sport arenas, and so on. This is also increasing the use of VoIP through an emerging community of people and professionals using Skype routinely and daily.

The use of radio communications allows to efficiently solve the above quoted mobility disadvantages of LANs; in particular they offer the following benefits:

1. a complete absence of cables between terminals and access points;
2. a complete mobility of terminals inside a covered area without the need of interrupting the connection between terminals and server;
3. an higher productivity of employers due to the gained higher mobility;
4. an easy and quick installation of new terminals, without cables to connect; a new user can be added simply by supporting the terminal with a wireless card;
5. a quite null effort to manage the infrastructure and its modifications;
6. cheaper local and international calls, free calls to other VoWLAN units and a simplified integrated billing of both phone and Internet service providers.

The convergence of voice and data over the same wireless devices (*e.g.* laptop, VoIP cordless phones, portable digital assistants PDAs) requires specific solutions to be applied at the following levels:

1. ***Hardware*** An high-speed control processing unit (CPU) is needed in each wireless terminal, able to adequately manage voice streams compression and de-compression tasks. High performance microphones and speakers are also needed to adequately support voice quality.

2. ***Software*** A number of typical problems due to the use of the wireless medium must be solved through the design of proper algorithms. For instance, these algorithms must guarantee the required quality of service (QoS) or to correct the effects of the typical latency of wireless communications.

3. ***Network*** A strong and reliable interaction between WLAN and the traditional telephony network is needed. In this task, real-time is an essential requirement to be satisfied.

4. ***Interference*** The effect of interference can be detrimental on a WLAN performance operating in the already crowded 2.4 GHz ISM band. In this case, no shielding or filtering solutions can be applied. The incoming external signal may lead to the loss of some data packets, hence reducing the possibility to reconstruct the original voice sequence.

Hereinafter, attention will mainly be paid to the effects of radio interference which, as quoted in Sec. 1, represent one of the most critical VoWLAN problems up to now still not completely investigated. The effect of the interference on a WLAN communication can be different and classified into two main classes: (i) the effects arising when interference occupies the frequency band on which the WLAN is starting to transmit. In this case, the network is forced to wait until the interference stops and the channel becomes free again; this phenomenon delays the delivery of packets and may cause disruptive effects on the voice call. (ii) The effects arising when interference acts during a WLAN communication; in this case the interference signal superimposes to the useful one causing errors in the delivered and received data stream. This kind of effect may lead to errors in the de-codification process of data packets with consequent loss of packets and artifacts in the voice call.

2.3 IEEE 802.11g standard

IEEE 802.11 is a standard used to provide wireless connectivity to fixed, portable, and moving stations within a local area (IEEE 802.11, 1999). It applies to the lowest two layers of the Open System Interconnection (OSI) protocol stack, namely the physical layer and the data link layer. The physical layer (PHY) is the interface between the upper media access control (MAC) layer and the wireless media where frames are transmitted and received. The PHY layer essentially provides three functions. First, it interfaces the upper MAC layer for transmission and reception of data. Second, it provides signal modulation through direct sequence spread spectrum (DSSS) techniques, or orthogonal frequency division multiplexing (OFDM) schemes. Third, it sends a carrier sense indication back to the upper MAC layer, to verify activity on the media. The data link layer includes the MAC sub-layer, which allows the reliable transmission of data from the upper layers over the wireless PHY media. To this aim, it provides a controlled access method to the shared wireless media called carrier-sense multiple access with collision avoidance (CSMA/CA). It then protects the data being delivered by providing security and privacy services. The 802.11 family includes multiple extensions to the original standard, based on the same basic protocol and is essentially different in terms of modulation techniques. The most popular extensions are those defined by the IEEE 802.11a/b/g amendments, on which most of the today manufactured devices are based. Nowadays, 802.11g is becoming the WLAN standard more widely accepted worldwide. It works in the 2.4 GHz band, like 802.11b, but operates at a maximum data rate of 54 Mbps, like 802.11a, with net throughput of about 19 Mbps. In practice, it provides the benefits of 802.11a but in the 2.4 GHz band. The 802.11g hardware is then backwards compatible with 802.11b hardware. It uses the OFDM scheme for the data rates of 6, 9, 12, 18, 24, 36, 48, and 54 Mbit/s, and reverts to complementary code keying (CCK) (like 802.11b) for 5.5 and 11 Mbit/s, and DBPSK/DQPSK+DSSS for 1 and 2 Mbit/s. 802.11g suffers from the same problem of 802.11b, namely it operates in the already crowded 2.4 GHz ISM band (2.4 - 2.4845 GHz). In this band, the standard defines a total of 14 frequency channels, each of which is characterized by a 22 MHz bandwidth. This implies that channels are partially overlapped, and that the number of non-overlapping usable channels is only 3 in FCC nations (ch 1, 6, 11) or 4 in European nations (ch 1, 5, 9, 13). Hereinafter, attention will mainly be paid to IEEE 802.11g standard.

2.4 Voice quality

In a VoIP call, the voice signal is fragmented into a set of data packets and delivered over an IP-based infrastructure. The quality of the voice call at the receiver side depends on the arrival order of the received packets, and on the presence of possible errors. If some packets are erroneously received, or characterized by a too long delay, all the process is delayed. For ordinary applications such as email or web, delays may not represent a critical problem. But, for the case of voice calls, like VoIP, where strict real-time constraints are required, delays can strongly degrade the voice quality perceived by end users.
Voice quality can be subdivided into the following two contributions:

***Listening quality (LQ)*:** the clearness of the voice message perceived by the listener in a given time interval;

***Conversional quality (CQ)*:** the quality of the conversation, including bi-directional phenomena like message delays at the receiver side and echoes.

It also depends on two main factors: (i) distortion, *i.e.* difference of the received signal and the transmitted one, (ii) overall delay, also known as “mouth to ear” delay, which includes all the collected delays. These two factors are strictly related to the network on which the call is sent. For example, a PSTN is typically rather immune to distortion and delays, while an IP network has the drawback to be more susceptible to such phenomena, and ultimately, in the specific case of wireless networks, to interference. A VoIP network has also addition delay contributions due to a number of performed intermediate operations like data coding, packets organization, queue management, de-jitter, etc. Another source of vocal distortion is the use of low bit-rate audio codec. More insights about the most typical impairments affecting voice quality in a VoWLAN conversation will be given in Sec. 3.
Voice quality can be analyzed in two different manners: (a) subjective or (b) objective measurements. Subjective measurements are conducted in terms of mean opinion score (MOS), which is the average result of opinion scores obtained by a group of listeners according to a rating scheme defined in (ITU-T P.800, 1996). The MOS is expressed as a single number in the range 1 to 5, where 1 (bad) is the lowest perceived quality, and 5 (excellent) is the highest perceived quality. It can be estimated only through in-laboratory conducted tests. MOS scores are attributed according to the voice quality perceived by the listeners who participated in tests. Tests are also to be executed in different boundary conditions, *i.e.* by changing the sentences, the deployed language and some listening conditions, which can lead to different MOS values. In fact, MOS scores achieved in different conditions can never be compared one with another. In (ITU-T P.800, 1996), four different test typologies are mentioned:

Conversation opinion test The test is carried out by couples of users using the phone system under test. At the end of conversations, a judgment is expressed by each user, and the average score, called MOSc (conversational MOS), is evaluated;

Listening Test/ACR (Absolute Category Rating) The test is performed by a group of listeners who give a judgment to a set of short sentences listened through the system under test. At the end of test, the average score, called MOS, is evaluated;

Listening Test/DCR (Degradation Category Rating) The test is performed by a group of listeners who analyze the differences between some short sentences taken as reference and the corresponding ones obtained by using the system under test. The result of the test is an average score, called DMOS (degradation MOS), accounting for the degradation effects effectively perceived;

Listening Test/CCR (Comparison Category Rating) The test is the same of DCR, but with the difference that listeners are here not informed about the type of message they are listening, *i.e.* if it is the reference or the corrupted one. The result of the test is an average score, called CMOS (comparison MOS).

Subjective measurements have the drawback to be very expensive and time consuming: they require a laboratory with characteristics satisfying specific requirements, and a number of people to be involved in the tests. This has lead to the development of new measurement techniques based on objective procedures and aimed at giving results similar to those obtainable with subjective measurements.
Objective quality measurements are performed through algorithms and can be intrusive or not intrusive. They are typically easy to implement, low cost and efficient in terms of measurement repeatability. Intrusive methods provide estimates of MOS introducing a voice sample in the network under test. In well-known algorithms like Perceptual Evaluation of Speech Quality (PESQ) or Perceptual Speech Quality Measure (PSQM) the measurement is performed by comparing the original sample with the received one. Non-Intrusive algorithms are instead based on the analysis of the only received voice stream, providing a transmission quality metric that can be used to estimate a MOS score. This method has the advantage that all calls in a network can be monitored without any additional network overhead, but the disadvantage that the effects of some impairment can not be measured. The most known non-intrusive method is the E-model defined in (Schulzrinne et al, 2003), based on the R factor, also known as *Transmission Rating Factor*. The objective of the model is to determine a quality rating incorporating the "mouth to ear" characteristics of a speech path. The range of the R factor is nominally 0-100, even if <50 values are generally unacceptable and typical telephone connections are never higher than 94, giving a typical range of 50-94. In the basic model, the R factor is expressed as follows:

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

where R_0 stands for the signal-to-noise ratio, i.e. the factor R in an ideal case with no disturbances and distortions, I_s is the simultaneous impairment factor, which accounts for the degradation due to simultaneous events like spurious tones and quantization distortions, I_d is the delay impairment factor, due to the delays and echoes, I_e is the equipment impairment factor due to some used devices like the icodec, and A is the advantage factor, which accounts for the tolerance of users to impairments. For instance, the typical tolerance is in the range 5-10 in a cell phone call, and null in a PSTN call. In Table 1, the typical MOS scores and R factors associated to some specific user opinions are shown for the case of a G.711 codec.

Listener Opinion	R Factor	MOS Score
Maximum obtainable for G.711	93	4.4
Very satisfied	90-100	4.3 - 5.0
Satisfied	80-90	4.0 - 4.3
Some users satisfied	70-80	3.6 - 4.0
Many users dissatisfied	60-70	3.1 - 3.6
Nearly all users dissatisfied	50-60	2.6 - 3.1
Not recommended	< 50	1.0 - 2.6

Table 1. MOS and R factor scores

3. Typical impairments

In-channel radio interference can strongly degrade the quality of a VoWLAN voice conversation as found experimentally and documented in Section 5. In order to better understand how interference can provoke this effect, basic notes about the most typical impairments affecting VoWLAN communications are here recalled.

3.1 Delay

One first impairment is due to the presence of delays (also called latency) in the arrival of data packets from the transmitter. In Fig. 1 a simplified scheme of a VoWLAN system architecture is reported, along with a symbol representing the interference.
As shown, a microphone is used to convert the incoming voice message into an analogue voltage signal. The signal is then converted into a digital data flow by means of an analog to digital converter (A/D); this data flow is subsequently fragmented, compressed and organized by a suitable encoder according to an IP-based scheme. Data packets are then modulated and converted into a radio frequency signal compliant with the IEEE 802.11 standard and delivered through an antenna to an access point. This latter one demodulates the incoming radio signal, collects and ordinates the received data packets. Data packets are subsequently delivered to a receiver terminal, which extracts the useful (payload) information, converts it into an analogue signal (digital to analogue, D/A, conversion) and

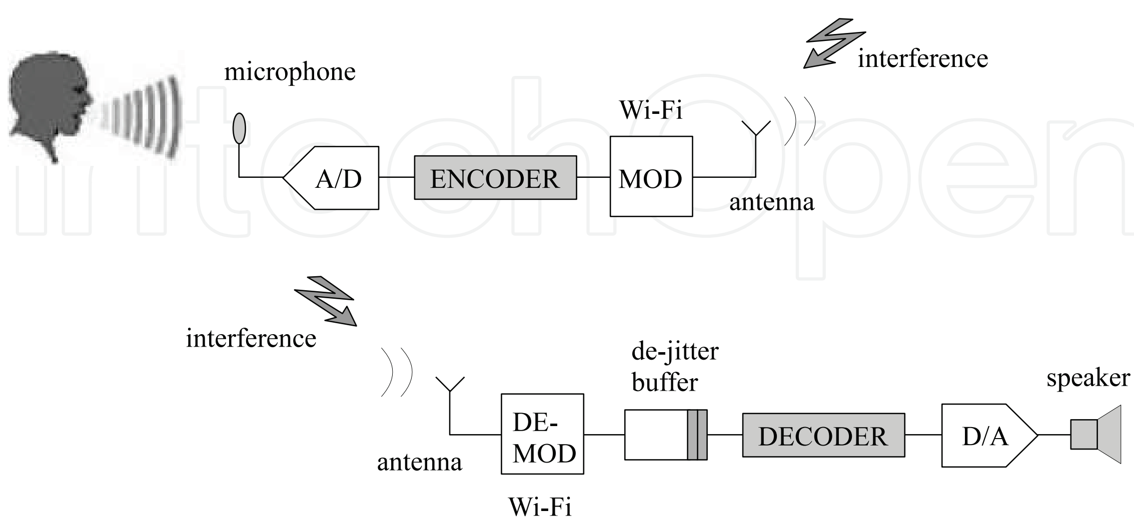


Fig. 1. Simplified architecture of a VoWLAN system under the effect of interference

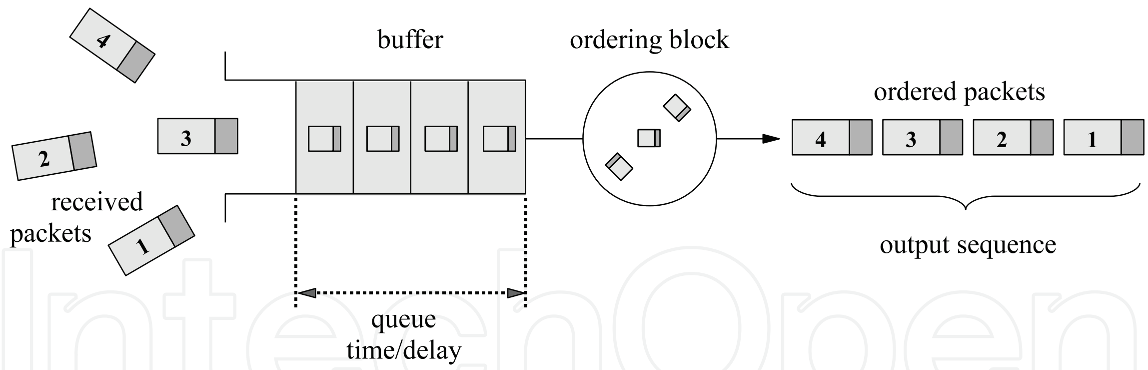


Fig. 2. Procedure deployed to order the data packets at the receiver side

reproduces the voice through a final speaker. In all this mechanism, interference acts on the "on-air" communication between the transmitter (*e.g.* a portable terminal) and the receiver (*e.g.* the access point) antennas.

In case of delays in the arrival of data packets, an unordered sequence of packets approaches the receiver, leading to the possibility of errors and impairments in the reconstruction of the original message. Therefore, a buffering stage is usually adopted at the receiver side, which keeps in memory the packets for a limited time interval, called *queue time*. In the queue time, the buffering stage orders the packets and reconstructs the original sequence. In Fig. 2, a sketch representing this mechanism is shown, in which four unordered data packets are finally arranged according to the desired sequence.

The described queuing mechanism requires the availability of buffers able to process the incoming data flow at a speed faster than the flow rate itself. The adopted queuing strategy is also important to avoid impairments especially in terminals characterized by poor capacity levels (bandwidth). To this aim, different data packet scheduling techniques are commonly adopted:

First in First Out (FIFO) It represents the simplest technique: packets are scheduled according to the arrival order, without any modification;

Weighted Fair Queuing (WFQ) It allows different bandwidths to each data flow according to a pre-assigned queing weight. Each data flow has a separate FIFO queue; this allows an ill-behaved flow (who has sent larger packets or more packets per second than the others) to only degrade itself and not other sessions;

Custom Queuing (CQ) Similar to WFQ, it shares the bandwidth between packet flows proportionally to a pre-assigned traffic class;

Priority Queuing (PQ) It ensures that highest priority data packets are scheduled before the lower priority ones, to which service can even be not guaranteed.

The final overall delay accounts for different contributions, among which we recall:

Propagation delay is the amount of time that a signal takes to travel from the transmitting to the receiving antennas over a medium. It can be computed as the ratio between the link length and the propagation speed over the specific medium. It becomes significant only in the case of long radio link distances;

Processing delay is the amount of delay due to the encoder and decoder processing activities, *i.e.* compression and decompression task, data fragmentation and data packets switching;

Queing delay is the amount of delay occurring both at transmitter and receiver side in the presence of data congestion. At the transmitter side, it occurs when packets are not processed and delivered with sufficient speed. At the receiver side, it occurs when the buffer capacity is not sufficient to manage all the received data;

End-to-end delay is the sum of the previous delays, which, in some particular cases, can be even greater than 500 ms, that is so high to cause superposition of users voices.

These delay terms, along with the ones of microphone, speaker, A/D and D/A converters, compose the so called *mouth-to-ear* delay, which value should never overpass a 150 ms threshold, over which the human ear perceives the presence of delays. In the end-to-end delay a number of parameters and phenomena can act, for instance the length of packets, interference, network traffic. For instance, longer packets are preferable in order to have a less compression of data, hence a shorter processing delay and a overall lower presence of header information. On the other side, shorter packets are preferable in order to obtain a reduced quieting delay to the detriment of processing delay and header size. This latter choice is just that more commonly adopted in a VoWLAN communication.

3.2 Jitter

Jitter is a critical phenomenon affecting communication systems and provoking impairments especially in those operating in real-time mode. It consists of a variation in packet transit delay typically caused by queuing processes at the transmitter and/or receiver side or by defects in the radio channel. It can be measured as the difference between the expected arrival time of a packet and the one effectively observed. The quality of a signal and in particular of a VoWLAN conversation can strongly be degraded by jitter. In order to mitigate jitter effects, a suitable time delay q is commonly added to each packet so that to equalize the time between packets. This task is commonly performed by a device at the receiver side, the so-called *dejitter buffer*, as sketched in Fig. 3, which adds to any received packet a suitably modulated time delay q. These intervals q make uniform the time cadence of arrival data packets, mitigating the effects of jitter.
The size of the dejitter buffer and the maximum delay q are typically chosen in such a way as to minimize the overall delay d among output packets and enhance the receiver ability to compensate the jitter. For instance, high q levels typically means a better ability of the

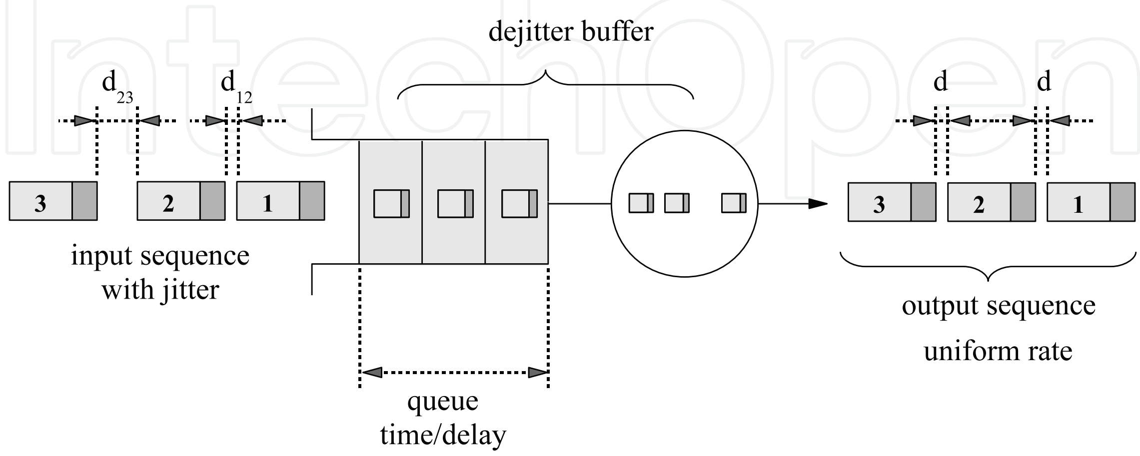


Fig. 3. Procedure deployed to compensate the non-uniform delays (jitter) of incoming data packets

device to compensate the jitter, but also longer *d*. Similarly, high buffer sizes typically mean a better ability to compensate jitter, but also the introduction of longer delays due to the dejitter operation. An efficient solution is typically the use of dynamic buffers, which size can be modulated according to the network status and the jitter value. This control is typically realized by measuring the delay introduced by the network (in particular its variance) and by stretching or reducing the length of the silence interval between consecutive packets. A typical measure of the dejitter time interval is instead 50 ms.

3.3 Packet loss

Packet loss is a frequent and critical phenomenon affecting data communication networks. In these networks, in fact, packet losses and errors are not typically tolerated. This requires the use of suitable and known mechanisms and strategies to replace missing data or to avoid/correct errors.

In the case of VoIP, the loss of one or more data packets or voice samples, as well as the presence of errors in the received stream, can be more tolerated. In fact, the final voice quality must be sufficient to satisfy a good listener, which means that one or more errors as well as packet or sample losses can be tolerated. This makes the two above quoted impairments (*i.e.* delays and jitter) more critical than packet loss in VoIP and VoWLAN networks.

A commonly adopted solution to mitigate packet loss is the so-called *packet loss concealment*. It consists of repeating the latest obtained sample in spite of the missing (not arrived) one. In particular, a time interval is assigned to each expected sample, at the end of which if the sample has not arrived yet the previous one is reproduced. For the G.729 codec, an overall 5% of average loss per call can be tolerated. Further and more sophisticated strategies conceal the missing samples by interpolating tha values assumed by the adjacent received ones.

The data packets structure should carefully be chosen by taking into account the following two issues: (i) the use of buffer queue of high dimensions reduces the effect of packet loss; (ii) long packets and buffer queue increase the overall delay, causing voice degradation.

In case of vocal code schemes like G.711, holes of 32-64 ms or longer may provoke loss of phonemes, and thus lead to disruptive voice degradation. Shorter holes in the range 4-16 ms or lower can instead be tolerated by any listener. The decrease of packet size below forty bytes is not always applicable because of the protocols IP, UDP and RTP and in particular of the size of the header they require. On the other hand too long packets may lead to too long delays, even beyond the ITU recommended levels.

3.4 Echoes

Echoes is a typical effect of phone conversation consisting in a series of delayed repetitions of voice sequences provoking distortions at the listener's ear. The phenomenon can be considered tiresome for delays longer than 25 ms. Due to a non-ideal impedance matching of the communication system elements, it is typically generated inside the gateway and the listener terminal. It can also be due to the resonance effect between microphone and speakers of a user VoIP or VoWLAN terminal when placed and operating close one with another. Echoes can be reduced by optimizing the system impedance matching: in these cases, echoes levels 50 dB lower than the useful signal one can be considered a very good target. They can also be mitigated by using efficient echo cancelers, *i.e.* digital devices implementing adaptive finite impulse response (FIR) filters and compensating the effects of echoes.

4. Measurement testbed

A number of experiments have been conducted with the aim of investigating on the effects of radio interference in the behavior of a WLAN when supporting VoIP applications. Experiments have been carried out by using a real testbed, operating in two different scenarios, in the following denoted as *A* and *B*. The testbed enlists an IEEE 802.11g wireless network (WLAN) supporting VoIP applications. Additional interference sources have been introduced in the proximity of the WLAN to emulate typical in-channel interference arising in real-world environment. Tests have been conducted within a protected and controlled environment, i.e. a shielded semi-anechoic chamber compliant with electromagnetic compatibility requirements for radiated emission tests.
In Figs. 4 and 5 the testbed deployed in the two analyzed scenarios is sketched; an its photograph is also shown in Fig. 6. It enlists the following elements:

1. an 802.11g access point, AP, D-link DI-624+;
2. a notebook, NB1, from Hewlett Packard, equipped with a Intel Pentium III processor, 296 MB RAM, Windows XP, and a 802.11g D-link DWL-G650 adapter+;
3. a notebook, NB2, from ACER, equipped with a 1.4 GHz Intel Centrino processor, 512 MB RAM, Windows XP, and a 802.11g D-link DWL-G650 adapter;
4. a notebook, NB3, from IBM, equipped with a Intel Pentium IV, Linux Ubuntu, and a 802.11g D-link DWL-G650 adapter;

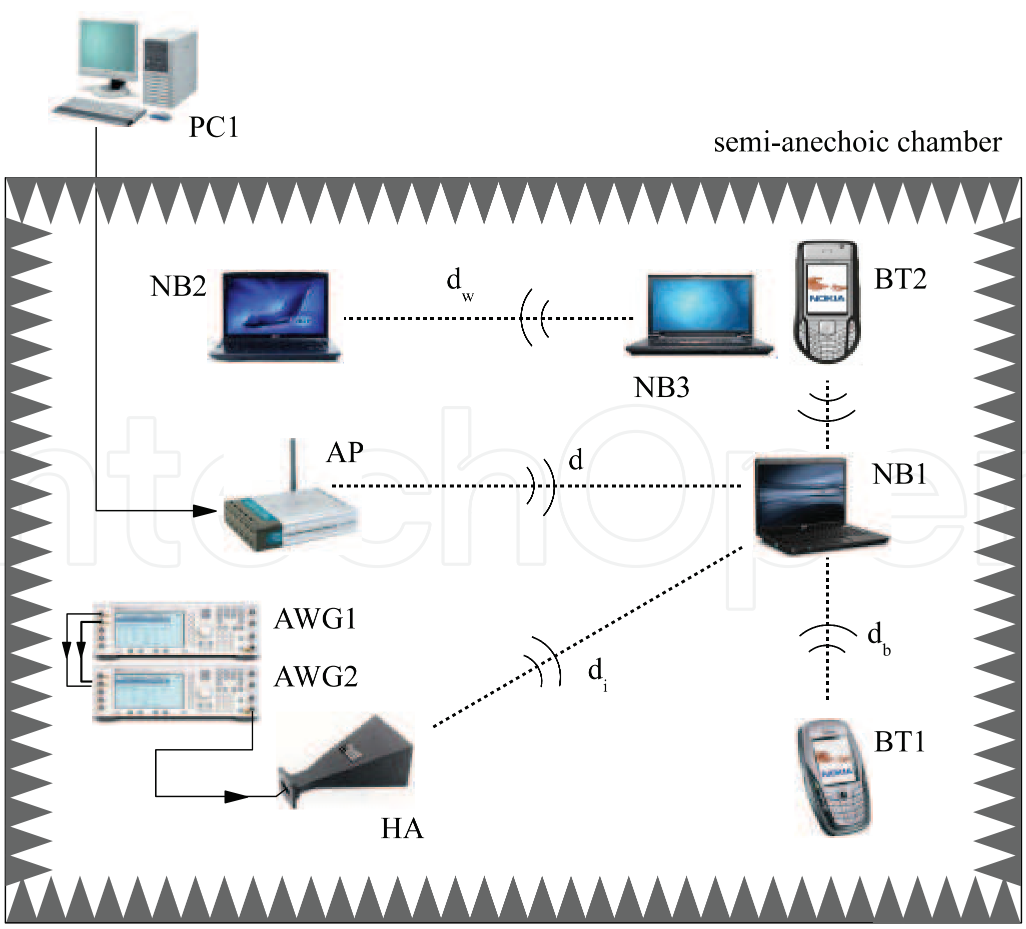


Fig. 4. Testbed configuration deployed in scenario A: wired-wireless VoIP communication

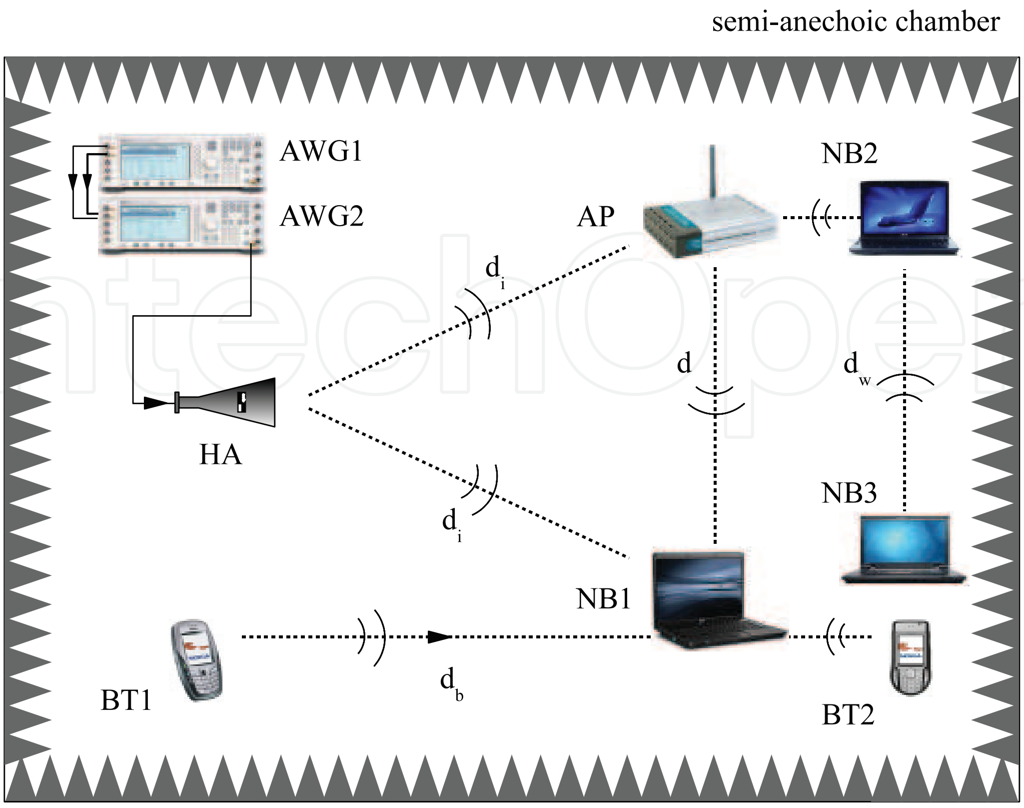


Fig. 5. Testbed configuration deployed in scenario B: wireless-wireless VoIP communication

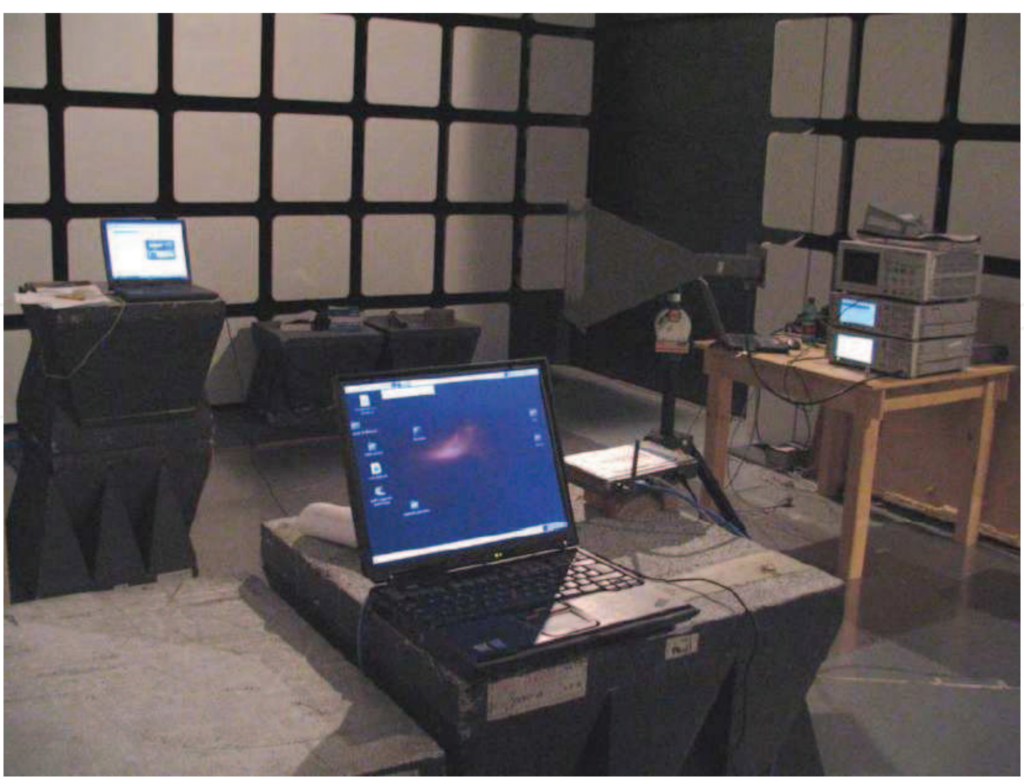


Fig. 6. Test site and adopted instrumentation

5. a computer desktop, PC1, equipped with a 1,4 GHz Intel Pentium IV, 1-3 GB MB RAM;
6. two mobile phones: a Nokia 6600, BT1, and a Nokia 6630, BT2, each equipped with a 1.1/class 2 Bluetooth transmitter;
7. two arbitrary waveform generators, AWG, Agilent Technologies E4431B ESG-D (250 kHz - 6 GHz frequency range) and E4438C ESG (250 kHz - 6 GHz frequency range);
8. a microwave horn antenna, HA, from Amplifier Research with 0.8-5 GHz frequency range;
9. a real-time spectrum analyzer, RSA, Tektronix RSA 3408, connected with a receiving microwave horn antenna Schwarzbeck BBHA9120D (1 - 18 GHz frequency range).

4.1 Scenario A

In this first measurement scenario, a wired-wireless configuration is emulated between two VoIP terminals: PC1 and NB1. As can be seen in Fig. 4, PC1 is placed outside the camber and communicates to the AP, inside the chamber, by means of a wired link. AP subsequently forwards the received data stream to NB1 by means of a WLAN connection. In this context, the use of VoIP over IEEE 802.11g is only analyzed in the download stage. The analyzed voice signals are pre-recorded voice messages suitably generated in order to let the voice quality measurement algorithms easily and efficiently detect the presence of voice impairments in the message such as latency, jitter, and packet loss. In particular, NB1 is placed at a distance d = 2.25 m from AP, while PC1, outside the chamber, is located within a shielded room. This allows the analysis of the effect of the interference, purposely generated inside the room, acting on the only wireless link. At the position of NB1, in case of null interference, the measured power level from AP is nearly -25 dBm.
The following interference are instead considered:

a. Bluetooth signal, generated by the couple BT1 and BT2, communicating with each other at a reciprocal distance $d_b = 4$ m, and with BT2 placed close to NB1. The power levels they generate provide a signal to interference ratio (SIR) at the WLAN receiver side (NB1) equal to 4 dB;

b. Additive White Gaussian Noise (AWGN), radiated by the antenna HA at a distance $d_i = 1.3$ m. In this case, the SIR level at NB1 is suitably varied changing the power at the AWG generator output connector;

c. Wi-Fi data traffic over the same frequency channel, generated by the couple of Wi-Fi terminals NB2 and NB3, placed at a reciprocal distance of $d_w = d$ and in the proximity of AP and NB1, respectively. In particular, NB3 is used to generate and transmit data traffic, at different data rate, and NB2 to receive it.

4.2 Scenario B

In this second measurement scenario, a wireless-wireless configuration is emulated between the following two terminals: NB1 and NB2. As represented in Fig. 5, NB2 generates VoIP traffic that NB1 receives through the intermediate AP. In this case, the VoIP over IEEE 802.11g call is analyzed at both upload and download stage. The architecture of the testbed is very similar to that of scenario A, with the exception of: PC1, here not considered, NB2, which generates VoIP traffic toward AP and receives interfering data traffic from NB3, and HA, placed at a distance $d_i = 3$ m from both AP and NB1 and oriented as shown in Fig. 5. The same interference sources of scenario A are instead considered.

4.3 Measurement instrumentation and software tools

Measurements have been conducted according to a cross-layer approach, which consists of several measurements, to be concurrently carried out at different layers of the ISO/OSI stack. The approach aims at experimentally correlating the major physical layer quantities to those characterizing key higher layer parameters (e.g. network/transport layer, application layer), allowing an efficient assessment of communication networks performance and drawbacks (Angrisani & Vadursi, 2007) and (Angrisani et al, 2007). In particular the following three layers have been considered: physical layer, through estimates of in-channel power and signal to interference ratio (SIR) at the receiver side, network/transport layer, by means of jitter and packet loss (percentage of lost packets) measurements, and application layer, through R factor and MOS estimates.

To the purpose, suitable measurement instrumentation and software tools have been deployed. In particular, physical layer measurements of in-channel power and SIR have been executed by using the RSA, in channel power mode, and the 1 - 18 GHz horn antenna (Bertocco & Sona, 2006). Network/transport and application layer estimates have instead been carried out by means of specific software tools, *e.g.* D-ITG, WRAPI+ and D-Link Air Plus Xtreme G Wireless Utility. D-ITG is a distributed Internet traffic generator (Botta et al, 2007), whose architecture allows to generate traffic and vary parameters such as inter-departure time, packet length, etc. It also allows measuring several QoS parameters at both the sender and receiver sides, and reporting a complete report of measured parameters over the entire measurement time. WRAPI+ is a real-time monitoring tool that enables a user to assess the values assumed by some performance parameters of a WLAN. In particular, it provides a complete report of information concerning the IEEE 802.11b/g network behavior in a given time interval. D-Link Air Plus Xtreme G Wireless Utility is a tool available from the D-Link DWL-G650 board allowing the monitoring of further parameters of the WLAN like for instance bit-rate and the received power level at both AP and NB1.

5. Experimental results

The two measurement scenarios have been investigated in five different configurations: 1) without interference, 2) Bluetooth interference, 3) AWGN interference, 4) WLAN concurrent data traffic, and 5) both AWGN interference and WLAN concurrent data traffic. Tests have been performed by considering only one audio codec for 1-4 configurations (G.711), and three different ones for the case of both AWGN interference and WLAN concurrent data traffic (G.711, G.723.1, G.729).

5.1 No interference

In the first experiment, a VoWLAN communication has been emulated in the absence of interference. VoIP calls have been generated by using pre-registered messages delivered from PC1 to NB1 (scenario A), and from NB2 to NB1 (scenario B). Measurements have been executed at the only receiver side. The obtained results show that:

- jitter is negligible;
- packet loss is nearly equal to zero;
- R factor reaches the value of 93, i.e. the maximum level for G.711 compression mode;
- MOS is equal to 4.4, i.e. the quality of the voice calls, at the received side, is more than satisfactory.

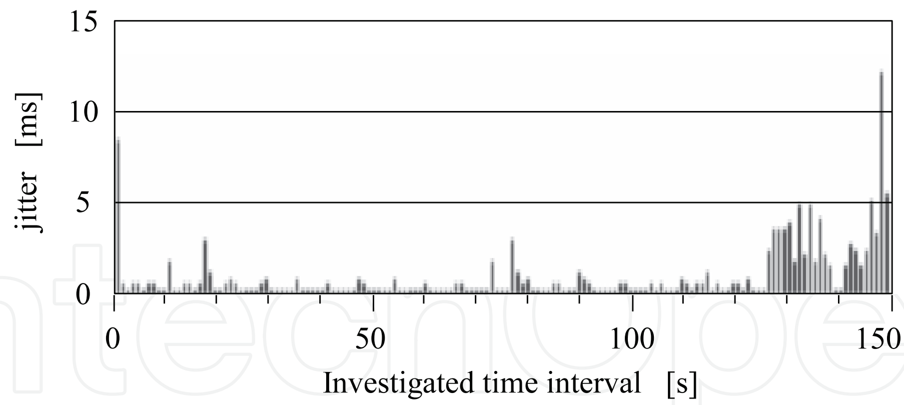


Fig. 7. Jitter estimated in the absence of interference

A diagram of the measured jitter levels in the case of scenario A is shown in Fig. 7 for an investigated time interval of 150 s length. Quite similar values have been obtained in scenario B. Fig. 7 shows the presence of delays, which origin can be attributed to the impairments in the deployed devices. However, the values they assume can be considered very low with respect to the maximum threshold of 150 ms that can be tolerated in a voice conversation, without significant loss of perceived quality (Douskalis, 1999).

5.2 Bluetooth interference

In the second experiment, the behavior of a VoWLAN communication has been studied under the effect of Bluetooth interference. As well known, Bluetooth devices radiate small power levels, *i.e.* typically in the range 0 through 20 dBm. Nevertheless the distance at which they commonly operate from computers and Bluetooth devices, like printers, mouse, and keyboards, is typically rather small, below 1 m, and the frequencies they use belong to the same ISM band, partially occupied by Wi-Fi networks. Therefore, despite the low levels of power, the effects of Bluetooth terminals on the analyzed VoWLAN application can not be a priori excluded.
Table 2 summarizes the results of the experiments conducted in the two scenarios. The table shows that in both the scenarios, the effects of interference are negligible both in terms of network/transport layer parameters, *i.e.* packet loss and jitter, and of application layer parameters, i.e. R factor and MOS. In fact, despite a reduction of the R factor and MOS with respect to the case of non-interference, the quality of the VoIP call is in the class "very satisfied". Quite the same values have been obtained at different positions of the Bluetooth terminals within the room and locating BT1 close to AP.

	Scenario A mean value	Scenario A st. dev	Scenario B mean value	Scenario B st. dev
packet loss [%]	0.060	0.003	0.100	0.009
jitter [ms]	0.100	0.003	0.600	0.003
R factor	90.090	0.010	90.090	0.010
MOS	4.380	0.020	4.370	0.050

Table 2. Effects of Bluetooth interference and R factor scores

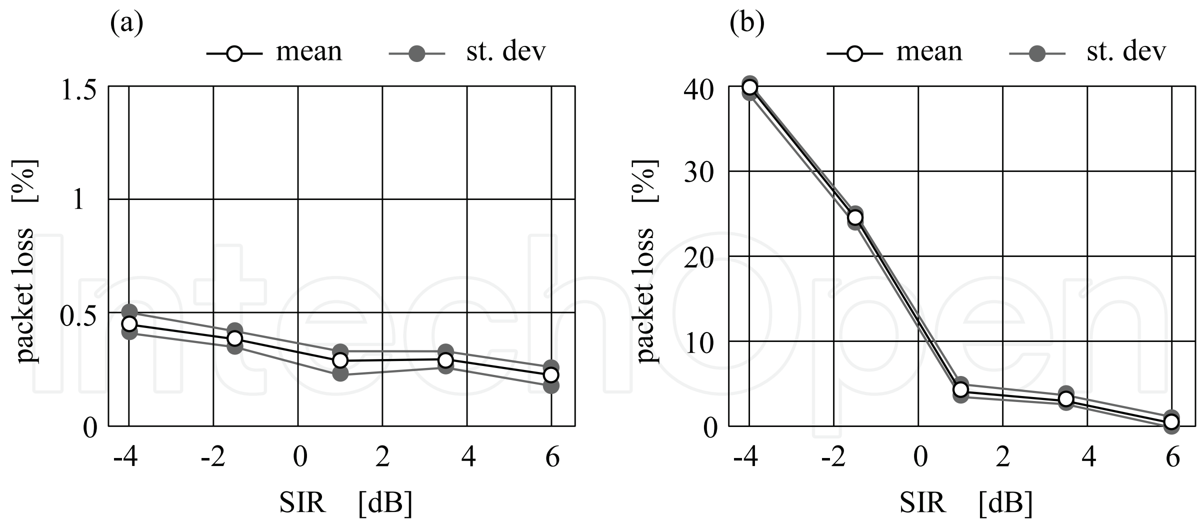


Fig. 8. Measured packet loss vs signal to interference ratio (SIR): (a) scenario A, (b) scenario B

5.3 AWGN interference

A third set of experiments have been conducted by considering the only effect of AWGN interference, affecting the VoWLAN streaming. The obtained results are summarized in Figs. 8, 9, and 10.

In Fig. 8, a relevant effect of interference in terms of packet loss can be noted in the case of scenario B and for $SIR < 1$ dB. Below this threshold, here denoted as SIR_{max}, packet loss grows rather quickly upon the decreasing of SIR, while for greater values it slowly lowers from 5 to 0 %. Much smaller values have instead been obtained in the case of scenario A, for any investigated SIR value. A similar difference between scenario A and B and for $SIR < 1$ dB can be observed in Figs. 9 and 10. Specifically, in the scenario A, for any considered SIR, the estimated values of jitter are rather low (below 2 ms) with respect to the tolerated limit of 150 ms, and the obtained R factor and MOS scores belong to the highest Table 1 category, *i.e.* "very satisfied". The only exception is for $SIR = -4$ dB, for which the voice quality can be considered "satisfied". In the scenario B, an abrupt growing of the estimated jitter levels

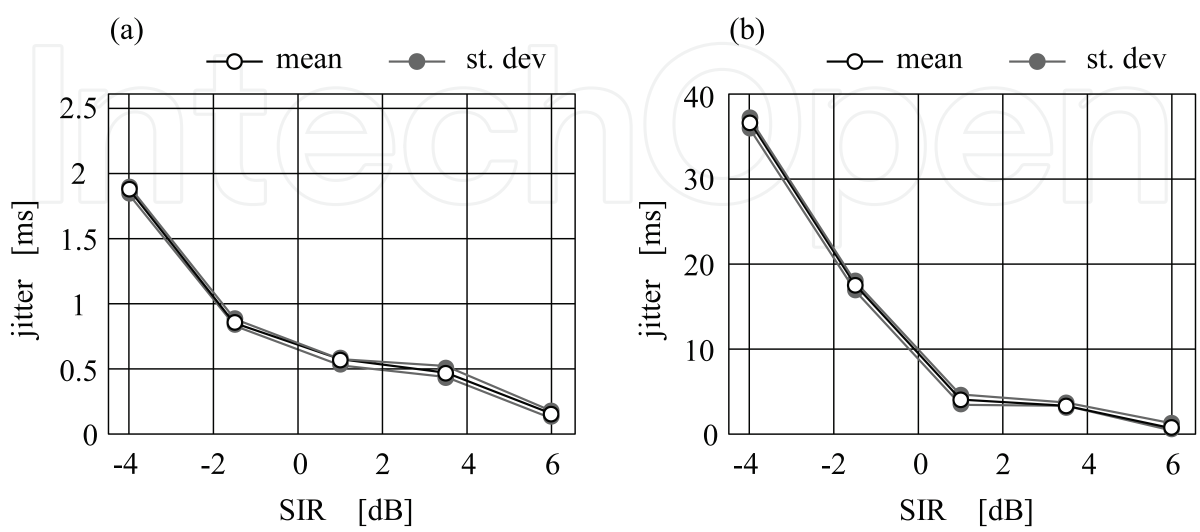


Fig. 9. Measured jitter vs signal to interference ratio (SIR): (a) scenario A, (b) scenario B

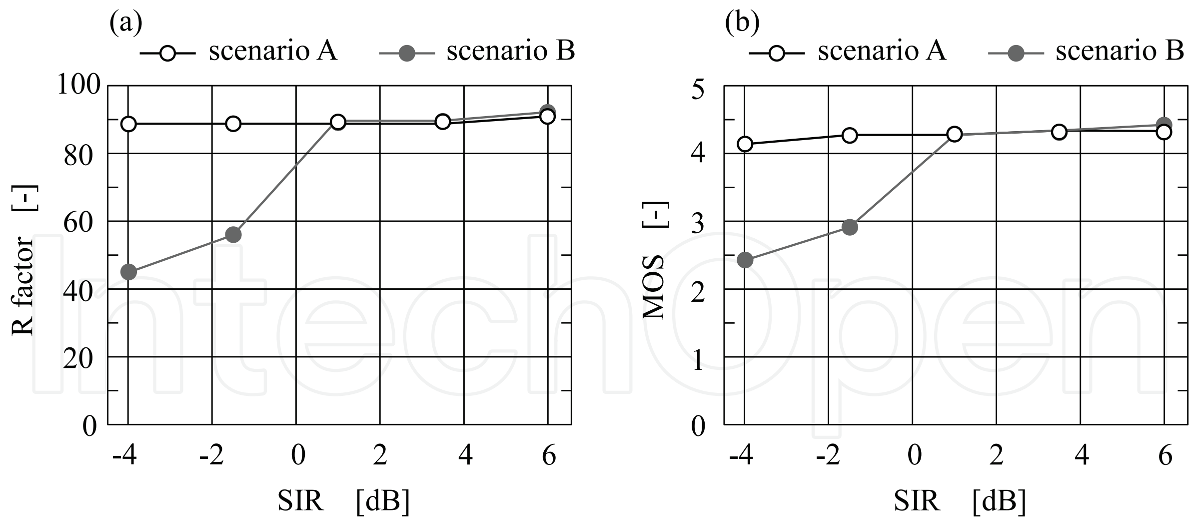


Fig. 10. Measured R factor (a) and MOS (b) vs signal to interference ratio (SIR)

can be noted for SIR levels below the threshold $SIR_{max} = 1$ dB. When $SIR = -4$ dB, the jitter assumes a non-negligible value (37 ms) with respect to the tolerated limit (150 ms), and even greater values are expected for $SIR < -4$ dB. Also in this case, R and MOS belong to the "very satisfied" category.

5.4 WLAN data traffic

In the fourth set of experiments, measurements have been conducted in the presence of a second interfering WLAN, here denoted as WLAN*, constituted by the couple of terminals NB2 and NB3 of Figs. 4 and 5. When WLAN* transmits, the WLAN under test is forced to wait until the end of the interference, and thus to defer the delivery of VoIP packets. The obtained results are summarized in Figs. 11, 12, and 13 for different WLAN* data rate from nearly 22 up to 46 Mbit/s. In the diagrams, the vertical line represents the maximum allowed data rate of the WLAN under test at medium access control (MAC) layer.

In the experiments, the power radiated by NB3 has been chosen higher than the reference

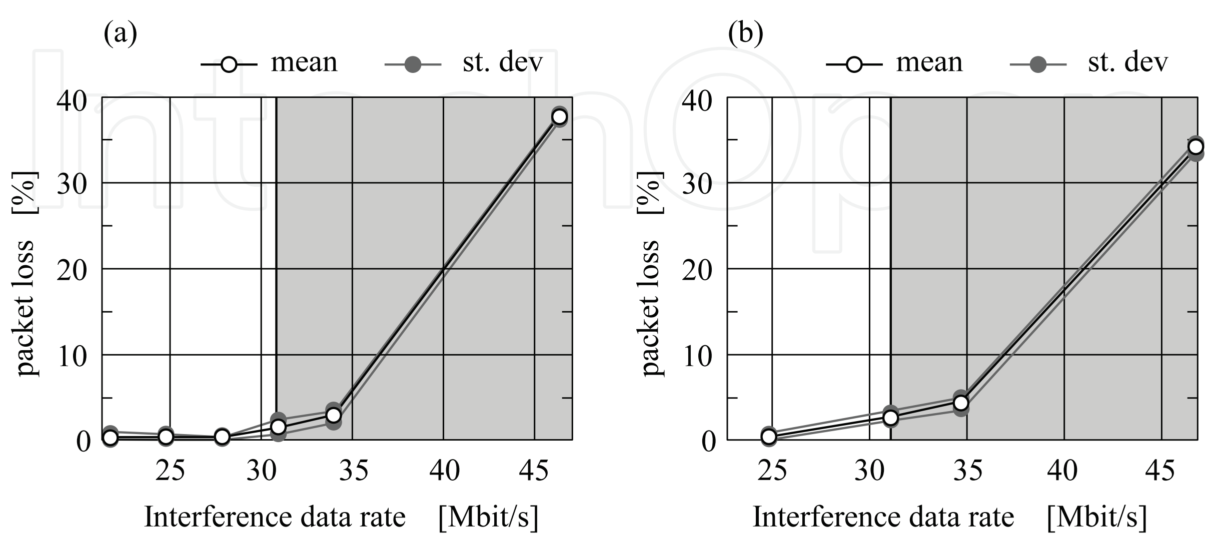


Fig. 11. Measured packet loss vs WLAN* traffic data rate: (a) scenario A, (b) scenario B

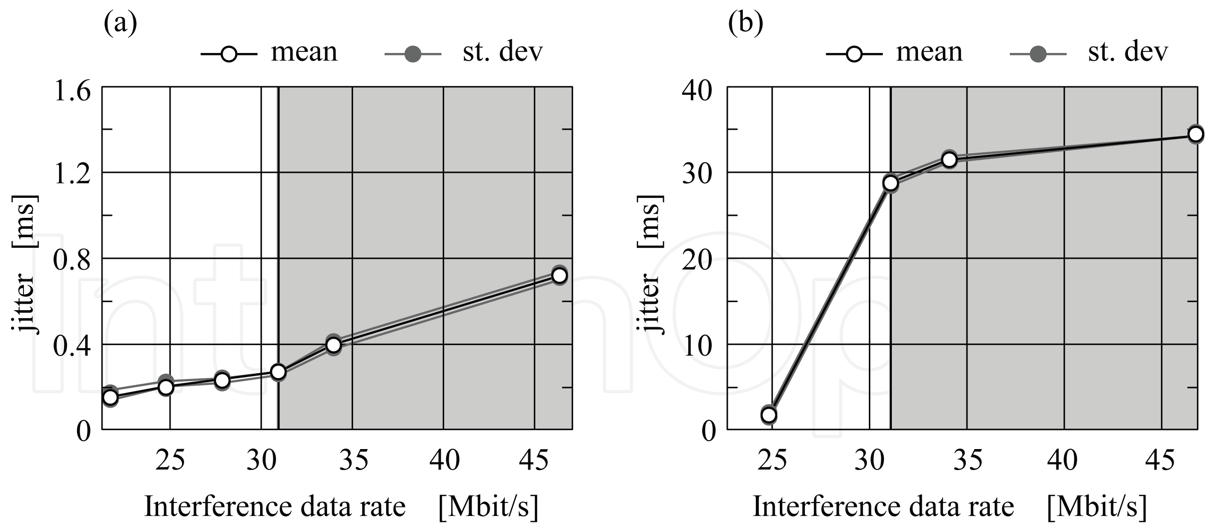


Fig. 12. Measured jitter vs WLAN* traffic data rate: (a) scenario A, (b) scenario B

threshold used by the AP to verify the status of the channel (free or busy). In Fig. 11, the detrimental effects of interference can be noted only for high data rates, beyond the vertical line, in the grey region (overload network). Beyond this limit, packet loss abruptly increases upon the growing of the data rate in both the scenarios, up to maximum values of 40 and 35 ms, respectively. In terms of jitter, Fig. 12 shows that the competitive data traffic can degrade the jitter but only in the scenario B. In fact, in the wired-wireless configuration, the estimated jitter is negligible (lower than 1 ms), while in the wireless-wireless setup it rapidly grows when the interference data rate approaches the network capacity at MAC layer. It can also be noted that for data rates lower than a threshold R_{max} of nearly 25 Mbit/s, the jitter is quite negligible. Fig. 13 finally confirms that also at application layer the effect of competitive WLAN data traffic is perceivable only at the highest data rates (greater than R_{max}). Below this threshold, a maximum voice quality can be obtained, while beyond R_{max} abrupt degradations are observed.

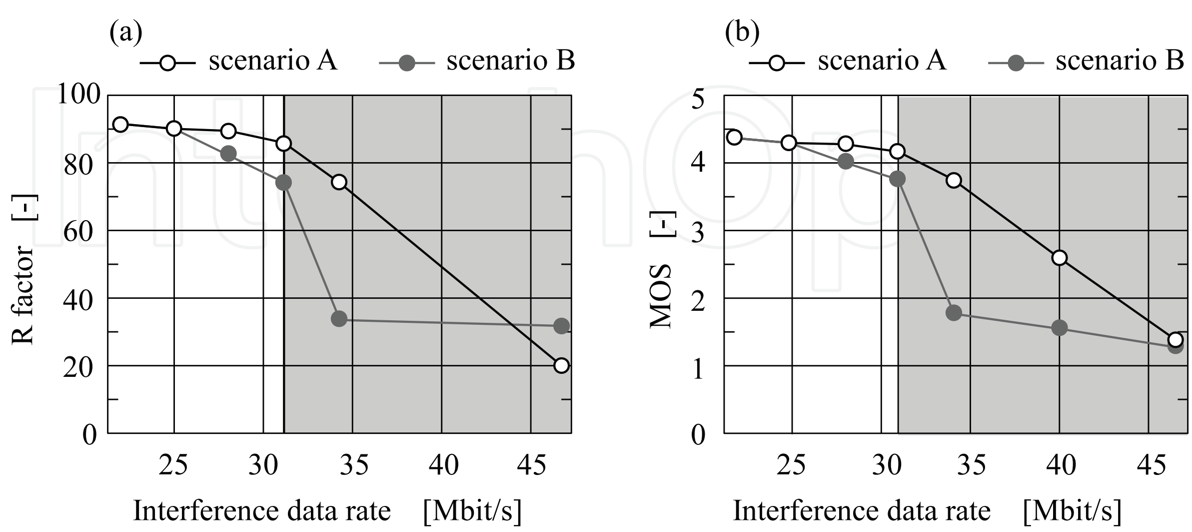


Fig. 13. Measured R factor (a) and MOS (b) vs WLAN* traffic data rate

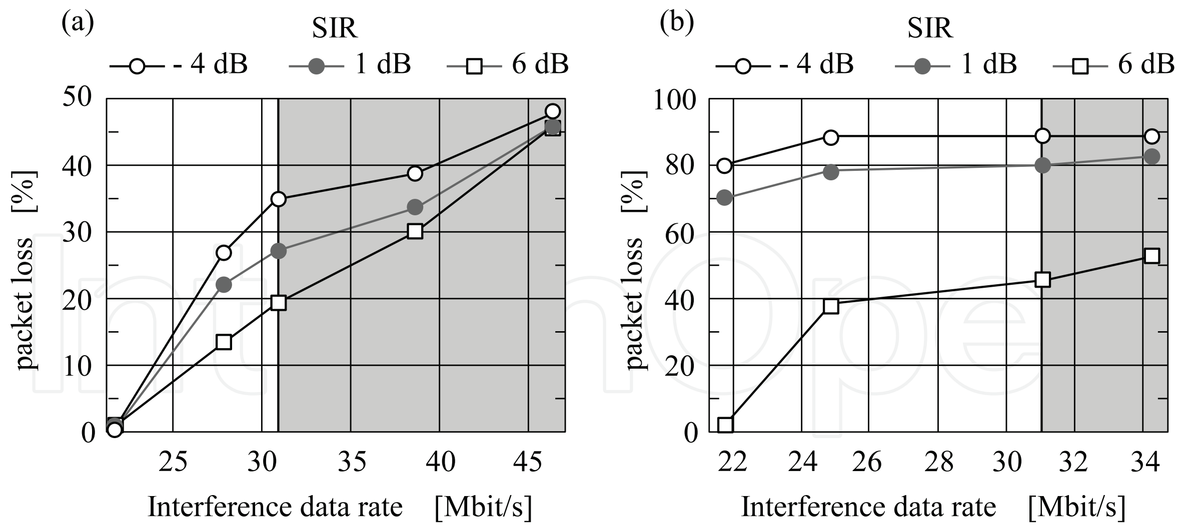


Fig. 14. Measured packet loss vs WLAN* traffic data rate for different SIR: (a) scenario A, (b) scenario B

5.5 AWGN interference and WLAN data traffic

A final group of experiments has involved the case of both AWGN interference and concurrent WLAN data traffic simultaneously operating. To this aim, the interfering sources, *i.e.* the AWGN generator and WLAN*, have been set in the same way as described in subsections 5.3 and 5.4. Measurements have been executed once again at different layers, specifically in terms of packet loss, jitter, R factor and MOS upon the varying of WLAN* traffic data rate in the range from nearly 22 up to 46 Mbit/s and for three different SIR levels: -4, 1, and 6 dB. The obtained results for both scenarios A and B and G.711 audio codec are summarized in Figs. 14, 15, and 16.

From the comparison of Figs. 11 and 14 results, some considerations can be drawn:

1. in the scenario A, the maximum data rate R_{max} beyond which packet loss abruptly increases, changes from 25 to nearly 22 Mbit/s. In the scenario B, this effect is even more

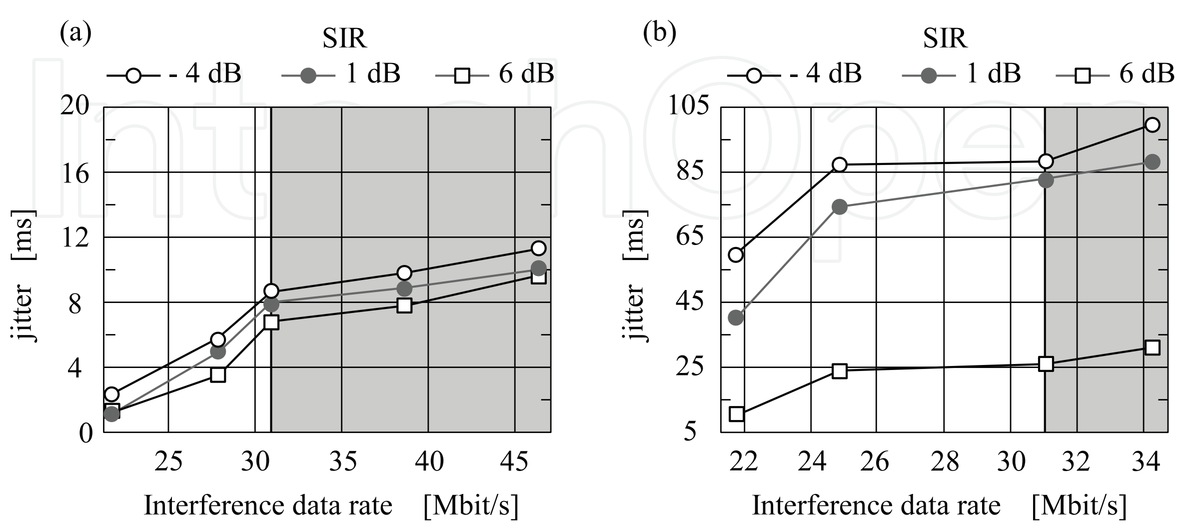


Fig. 15. Measured jitter vs WLAN* traffic data rate for different SIR: (a) scenario A, (b) scenario B

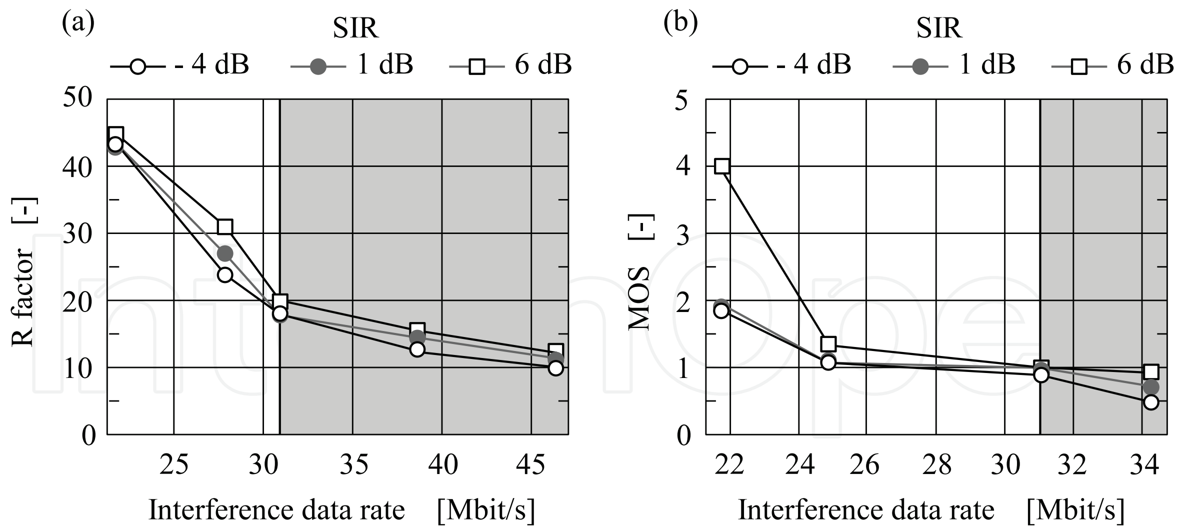


Fig. 16. Measured R factor (a) and MOS (b) vs WLAN* traffic data rate for different SIR

visible; in fact, R_{max} is nearly 22 Mbit/s for $SIR = 6$ dB and << 22 Mbit/s for $SIR \leq 1$ dB, as shown by the two upper curves, which, in the range 22 - 34 Mbit/s, assume very high values (≥ 70 dB). In this case, further measurements should be performed at lower data rate to determine the corresponding values R_{max} below which packet loss becomes negligible or even null;

2. in terms of jitter, Fig. 15 shows that scenario A is rather immune even to the simultaneous presence of AWGN interference and concurrent data traffic. In fact, the estimated jitter curves appear very close one with another with values not higher than 12 ms, that means quite negligible with respect to the 150 ms threshold. A different effect can instead be noted in the wireless-wireless setup, where packet loss significantly worsens upon the increasing of AWGN interference intensity. Also in this case, the effect of AWGN interference is clearly visible for SIR values equal to or lower than 1 dB;

3. Fig. 16 finally shows that at application layer the simultaneous presence of both competitive WLAN data traffic and AWGN interference is very detrimental even with data rate values in the range $22 - 25$ Mbit/s and for any considered SIR value. The obtained R factor highlights that "very satisfied" levels of voice quality cannot be obtained for concurrent data rate levels higher than 22 Mbit/s.

Further tests have been performed at the same setup conditions but with different audio codecs, i.e. the aforementioned G.723.1 and G.729. The following results have been observed:

A. In terms of packet loss, G.711 is the audio codec that provides better results. In particular, a nearly 10% worsening of packet loss is observed for both G.723.1 and G.729 regardeless of the considered intereference data rate.

B. G.711 is also better in terms of jitter, which, for the G.729 codec, assumes very high values, even up to nearly 75 ms for an interference data rate equal to 35 Mbit/s.

C. The R factor is quite the same for G.723.1 and G.729 codecs, and much higher for G.711. For instance, in the scenario B and with 25 Mbit/s of interference data rate, the estimated R factor is 85 for G.711 and nearly 67 for G.723.1 and G.729 codecs.

D. Similarly, MOS is quite the same for G.723.1 and G.729 codecs, and much higher for G.711. For instance, in the scenario B and with 25 Mbit/s of interference data rate, the estimated MOS is 4.3 for G.711 and nearly 3.8 for G.723.1 and G.729 codecs.

6. Conclusion

A number of experimental results have been presented in order to investigate on the interference effects of Bluetooth signals, AWGN and WLAN competitive data traffic on IEEE 802.11g WLAN supporting VoIP applications. Cross layer measurements performed in terms of SIR, jitter, packet loss, R factor and MOS have been carried out with the aim of analyzing the best configurations of parameters like the interfering WLAN data rate and the measured SIR at the receiver side. For instance, in both the analyzed scenarios, i.e. wired-wireless and wireless-wireless WLAN, the maximum interfering WLAN data rate R_{max} and the minimum SIR, SIR_{min}, values have been estimated.

It has been demonstrated that the use of VoIP over WLAN can strongly be interfered by the presence of in-channel noise-like signals, such as AWGN, and of competitive data traffic generated by a near operating WLAN exploiting the same frequency channel. Therefore, parameters like SIR and WLAN interference data rate should always be carefully monitored and, if possible, adjusted beyond or below the thresholds R_{max} and SIR_{min}, respectively, to be estimated as suggested in the chapter. The use of G.711 codec is also suggested against the simultaneous effect of both concurrent data traffic and radio interference.

Many other measurement sessions could be performed to investigate on further interference phenomena here not considered for more conciseness. For instance, the analysis could be extended to the study of the interference effects due to burst-like signals or real life ones. It could also be very interesting extending the study to many other system parameters, like for instance those concerning system's quality of service.

7. References

Lin, Y. B. & Chlamtac, I. (2000). *Wireless and Mobile Network Architectures*, John Wiley and Sons, ISBN 978-0-471-39492-1, New York, US.

Douskalis, B. (1999). *IP Telephony: The Integration of Robust VolP Services*, Prentice Hall, ISBN 978-0-13-014118-7, New Jersey, US.

IEEE Standard 802.11 (1999). *Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications*, IEEE computer society.

IEEE Standard 802.15.4 (2003). *Wireless Medium Access Control (MAC) and Physical Layer (PHY) Specifications for Low-Rate. Wireless Personal Area Networks (LR-WPANs)*, IEEE computer society.

IEEE Standard 802.16 (2001). *IEEE Standard for Local and Metropolitan Area Networks - Part 16: Air Interference for Fixed Broadband Wireless Access Systems*, IEEE computer society.

Garg, S. & Cappes, M. (2003). An Experimental Study of Throughput for UDP and VoIP Traffic in IEEE 802.11b Networks, *Proceedings of Wireless Communications and Networking*, pgs 1748-1753, New Orleans, LA, US, March 2003.

Angrisani, L. & Vadursi, M. (2007). Cross-layer Measurements for a Comprehensive Characterization of Wireless Networks in the Presence of Interference, *IEEE Trans. on Instrumentation and Measurement*, Vol. 56, No. 4, 2007.

Wang, X. G.& Mellor, G.M. (2004). Improving VOIP application's performance over WLAN using a new distributed fair MAC scheme, *Proceedings of Advanced Information*

Networking and Applications, pgs 126-131, ISBN: 0-7695-2051-0, March 2004, Fukuoka, Japan.

Wang, W. & Li, S.C.L. (2005). Solutions to Performance Problems in VoIP Over a 802.11 Wireless LAN, *IEEE Trans. on Vehicular Technology*, Vol. 54, No. 1, Jan 2005, pgs 366-384.

Garg, S. & Cappes, M. (2002). *On the Throughput of 802.11b Networks for VoIP*, TechnicalReport ALR-2002-012, Avaya Labs, 2002.

El-fishawy, N. A. & Zahra, M. M. & El-gamala, M. (2007). Capacity estimation of VoIP transmission over WLAN, *Proceedings of Radio Science Conference*, pgs 1-11, March 2007, Cairo, Egypt.

Prasat, A. R. (1999). Performance comparison of voice over IEEE 802.11 schemes, *Proceedings of Vehicular Technology Conference*, pgs 2636-2640, Vol. 5, Sept. 1999, Houston, Tx, US.

Hiraguri, T. & Ichikawa, T. & Iizuka, M. & Morikura, M. (2002). Novel Multiple Access Protocol for Voice over IP in Wireless LAN, *IEEE Int. Symp. on Computers and Communications*, pgs 517-523, ISBN: 0-7695-1671-8, July 2002, Taormina, Italy.

ITU-T Recommendation G.711 (1972). *Pulse Code Modulation (PCM) of Voice Frequencies*, 1972.

ITU-T Recommendation G.729 (1996). *Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear Prediction (CS-ACELP)*, 1996.

ITU-T Recommendation G.723.1 (2006). *Digital Terminal Equipments - Coding of Analogue Signals by Methods Other Than PCM. Dual Rate Speech Coder for Multimedia Communications Transmitting at 5.3 and 6.3 kbit/s*, 2006.

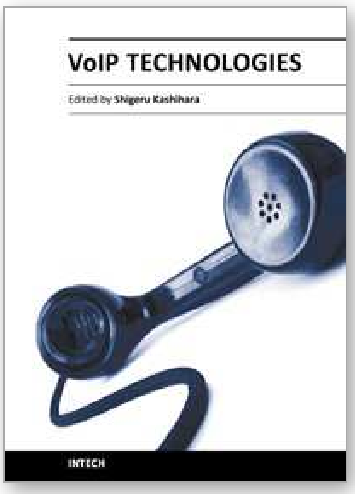
ITU-T Recommendation P.800 (1996). *Methods for Subjective Determination of Transmission Quality*, 1996.

Schulzrinne, H. & Casner, S. & Frederick, R. & Jacobson, V. (2003). *RTP: A Transport protocol for Real-Time Applications*, RFC 3550, July 2003.

Angrisani, L. & Bertocco, M. & Fortin, D.& Sona, A. (2007). Assessing coexistence problems of IEEE 802.11b and IEEE 802.15.4 wireless networks through cross-layer measurements, *IEEE International Instrumentation and Measurement Technology Conference*, paper n. 7326, ISBN: 1-4244-0588-2, May 2007, Warsaw, Poland.

Botta, A. & Dainotti, A. & Pescape, A. (2007). Multi-protocol and multi-platform traffic generation and measurement, *INFOCOM 2007 DEMO Session*, May 2007, Anchorage, Alaska, USA.

Bertocco, M, & Sona, A. (2006). On the power measurement via a superheterodyne spectrum analyzer, *IEEE Trans. on Instrumentation and Measurement*, pgs. 1494-1501, ISSN: 0018-9456., Vol. 55, No. 5, 2006.



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This book provides a collection of 15 excellent studies of Voice over IP (VoIP) technologies. While VoIP is undoubtedly a powerful and innovative communication tool for everyone, voice communication over the Internet is inherently less reliable than the public switched telephone network, because the Internet functions as a best-effort network without Quality of Service guarantee and voice data cannot be retransmitted. This book introduces research strategies that address various issues with the aim of enhancing VoIP quality. We hope that you will enjoy reading these diverse studies, and that the book will provide you with a lot of useful information about current VoIP technology research.