Abstract—In this paper the issue of networks interworking is faced with reference to UMTS cellular networks and IEEE802.11 wireless local area networks. A feasible architectural solution of the integrated network is given and the possibility to serve by means of the WLAN technology also those voice calls that would be blocked by UMTS, in case of resource saturation, is investigated. The performance level provided by the integrated network in a realistic scenario is here assessed considering both IEEE802.11a and the forthcoming IEEE802.11e WLAN technologies for possible integration with UMTS.

I. INTRODUCTION

Third generation cellular networks are already a reality: the provision of high speed communications supporting an increasing number of different services is the challenge faced by UMTS, whose deployment is expected to foster multimedia communications and wideband access to the Internet with cellular terminals.

At the same time wireless local area networks (WLANs) are achieving a great penetration in the mass market as a really effective solution to provide mobile access to the Internet; companies all over the world are already offering WLAN connections in particular locations, such as airports and hotels.

In these areas, hereafter denoted as "hot spots" (HSs), anyone owning the appropriate technology on his laptop can connect to the Internet at a reasonable price and with a satisfactory connection speed.

Nonetheless, the request for bit rate is expected to further increase in the next future and more capacity will be necessary; on these conditions, UMTS and WLAN interworking becomes a really significant issue to be investigated: provided that the final user is equipped with a dual mode terminal, combining the two technologies, thus increasing the "pool" of available resources, would considerably increase both the users’ satisfaction and the network utilization efficiency.

In this paper we investigate the possibility to serve by means of the WLAN technology, which can operate at bit rates considerably higher than UMTS, also those voice calls that cannot be served by UMTS (and would therefore be blocked) because of a saturation of its radio resources.

Obviously a format conversion from circuit-switched (CS) voice flows into packet-switched (PS) Voice over IP (VoIP) flows (and vice versa) is needed in order to make feasible the radio network switching.

The benefit of UMTS-WLAN interworking is here investigated with reference to the Chinese TD-SCDMA UMTS technology [1] and to both IEEE802.11a [2], [3] and the upcoming IEEE802.11e [4] WLAN technologies. Here, in particular, an architectural solution for UMTS-WLAN integration is given and the users’ satisfaction level provided in a realistic scenario by the integrated network is assessed; moreover, the performance improvement due to the adoption of the IEEE 802.11e WLAN technology instead of the simpler IEEE802.11a in the integrated UMTS-WLAN network is evaluated.

The paper is outlined as follows. In section II the scenario under investigation is introduced, in section III the UMTS-WLAN integration is discussed from the network architecture point of view, in section IV the structure of the software platform adopted for our investigations is described, in section V and in section VI the simulation settings and the related numerical results are reported and finally, in section VII the final conclusions are drawn.

II. SCENARIO

The scenario investigated, depicted in figure 1, consists of 18 UMTS Nodes-B with tri-sectorial antennas over an area of 3720x2775 squared meters; the WLAN access point (AP) is assumed co-located with a Node-B (one of the central ones) and can potentially serve users located within a distance of 25 meters; to avoid border effects all merit figures investigated in the numerical results section refer to an area of 100x100 squared meters centered in the AP (see figure 1).

In order to reproduce the variety of services of a real scenario, three traffic classes were considered: voice, web browsing and file transfer (hereafter referred to as FTP traffic).

As for the traffic distribution in the scenario, here we considered the superimposition of a "background traffic", generated by users uniformly distributed in the whole 3720x2775 m² region, and of a "hot spot traffic", generated by users uniformly distributed in the circular hot spot coverage area.

Background traffic: it is generated by "background users" that are assumed to perform phone calls (generating voice traffic) or web-browsing sessions; when located outside the hot spot the "background users" are served by UMTS (if there are available resources, otherwise are blocked); within the hot spot they can be served also by the WLAN.

Hot spot traffic: it is added to the "background traffic" in the hot spot region and it is constituted by additional contributions of voice and web-browsing traffics and a further component of FTP traffic. Within the hot spot, data traffics (both web browsing and FTP) are served by the WLAN, while voice calls are supported by UMTS, if there are available...
resources, otherwise they are blocked (if no voice redirection is considered) or redirected to the WLAN (if voice redirection is considered).

The above reported assumptions are summarized in the first three columns of table I, while in the fourth column the transport level protocol adopted for each kind of traffic is reported.

The straightforward consequence of the above reported assumptions is that the hot spot region is characterized by a higher user density than the surrounding region, as is typical of crowded areas such as, for instance, an airport gate or a shopping mall where, actually, the realization of a hot spot is envisioned.

III. ARCHITECTURAL ISSUES

UMTS and WLAN interworking is a topic issue for beyond-3G systems. Assuming that dual mode terminals are available, two aspects have to be detailed: an architectural proposal for the integrated network and a feasible practical realization.

Regarding the network architecture allowing the interconnection between UMTS and WLANs, different approaches have been proposed so far (e.g., [5], [6]): the mobile IP approach (also called loose coupling), the gateway approach and the emulator approach (also called tight coupling).

The mobile IP approach provides roaming between UMTS and WLAN networks according to the mobile IP paradigm: the two networks are completely independent, the long duration of the inter-system handover procedure could deeply affect the user satisfaction.

The gateway approach does not require mobile IP: UMTS and WLAN are still independent networks, but a gateway between them allows to convey data flows generated by roaming users towards the original transport networks (UMTS GGSN or WLAN router), thus reducing the handover latency.

Finally the tight coupling approach: WLAN can be seen as an alternative UMTS access stratum so that the Internet is uniquely accessed through the GGSN; the handover is faster than the previous cases, but the tighter interworking between UMTS and WLAN can only be realized through an explicit exchange of 3GPP messages.

In this paper the performance of the UMTS-WLAN integrated network are investigated with reference to the tight coupling approach, which is at same time the most challenging and the most promising architectural solution.

Different solutions can be envisioned to realize the tight coupling between the WLAN AP and the UMTS Terrestrial Radio Access Network (UTRAN): the AP could be seen, for instance, as an additional cell or a Node B (let us recall that a Node B can manage more than one cell). Let us observe, however, that while the Iub control plane could be adapted to support an "enhanced" cell or Node B, the user plane implementation would be much more complicated, due to the fact that Uu layer 2 Medium Access Control (MAC) and Radio Link Control (RLC) protocols, mapped over Iub Frame Protocol could not support the higher rates required by a WLAN connection.

Thus, in our opinion, the best architectural solution is moving to a Radio Network Subsystem (RNS) perspective: a Radio Network Controller (RNC) emulator is inserted between the AP and the Core Network (CN) (see figure 2). So doing, we can exploit the essential fact that neither the adjacent RNC nor the CN need to know anything about the internal resource management of RNSs. Moreover, we think that an Iur logical interface between the standard RNC and the RNC emulator is not required: due to the fact that we cannot use macro-diversity between cells under Serving-RNC (UMTS cells under standard RNC) and cells under Drift-RNC (WLAN AP under RNC emulator), Iur user plane would not be used. Regarding signalling for inter-system handover procedure, we could then use Serving RNS Relocation function that manages the Iu interface connection mobility from a RNS to another over Iu, without loss of PDP context or any other Session Management information.

Concerning the data flows, when a user terminal is connected to the WLAN RNS, Iu-CS speech data coming from CN should be converted by the emulator into VoIP flows towards the AP and vice versa; Iu PS data flow should be easier routed from the emulator to the AP and vice versa.

Besides the above discussed architectural solution of the integrated network, which mainly concerns the hierarchical level of the WLAN within the UMTS network, a feasible practical solution has to be investigated in order to conveniently provide a logical and a physical link between the RNC emulator and the CN. Here we imagine that the same network provider will
manage both the WLAN and the UMTS networks, hence it is likely that the UMTS Node B and the WLAN AP will be co-located, to reduce costs and to ease maintenance.

The physical contiguity between the RNC emulator and the Node B suggests to adopt already existing UTRAN interfaces for the physical and logical WLAN-UMTS integration. Due to the fact that Iub/Iu Layer 1 shall comply with the same requirements [1], and that the physical layer provides the same ATM services according to ITU-T I.361, we could multiplex different logical interfaces over the same physical link (see figure 3). In particular, an ATM switch could be inserted in the physical connection already existing between the Node B and the RNC, in order to switch the flows to/from the Node B (logical Iub interface) and the flows to/from the RNC Emulator (logical Iu interface).

In this scenario the ATM switch should be located close to the Node B and consequently also close to the AP, which is also convenient for network management and maintenance. Thus the physical link between the ATM switch and the standard RNC carries two different logical interfaces: the Iub from Node B and the Iu from WLAN RNS. The latter should then transparently cross the standard RNC exploiting the RNC switching function.

Finally, the link between the RNC and the CN convey Iu for both the standard RNS and the WLAN RNS.

It is worth noting that, apart from the short links connecting the AP, the RNC emulator and the ATM switch, no significant physical interconnections have been introduced by this solution; obviously, the capacity of the link between the ATM switch and RNC should be adapted in order to convey also the WLAN traffic.

Finally, between the standard RNC and the RNC emulator, another logical link could be added (the path would be the same used for Iu by WLAN RNS): this link could be dedicated for Operations and Maintenance (O&M) function; we expect that a Common Radio Resource Management between different RNSs and particularly between a standard RNS and a WLAN RNS could be carried out through this O&M link, leading to an increase of network performance.

IV. SIMULATORS PLATFORM OVERVIEW

In order to investigate the performance of the UMTS-WLAN integrated network, we developed a simulation tool aimed at reproducing the behavior of the two interworking technologies, with particular attention to the physical (PHY) level aspects. It has to be pointed out, in fact, that when dealing with radio communications, phenomena such as, for instance, interference (e.g. the inter-cell interference in an UMTS scenario) or the time correlation of the frequency selective fading due to multipath propagation (typical of an indoor WLAN scenario) deeply affect the systems performance, either increasing the bit error rate or influencing the behavior of the MAC and the Transport Control Protocol (TCP).

Many of these aspects are often neglected or considered in simplistic ways in commonly used network simulators (such as, for instance, Opnet [7] or ns-2 [8]); moreover, the simulation of integrated interworking networks obviously requires the adoption of interworking network simulators, thus preventing from the adoption of commercially available tools. These considerations motivated the development of a dedicated simulation tool which is the integration of UMTS and WLAN simulators we previously developed and utilized for our research activities (e.g., [9], [10]).

In particular, we realized a platform with a server-core simulator (hereafter called Upper Layers Simulator, ULS) and one or more client simulators (Lower Layers Simulators, LLS): the ULS takes care of the user-related information, such as its position and movements, and of the end-to-end aspects of each connection, such as the TCP or UDP dynamics and the generation of the application-level traffic. Being related to the end-to-end aspects of communications, the ULS structure is independent on the particular technology (UMTS, WLAN, ...) adopted to establish the user connection.

All aspects related to the technologies adopted, hence related to the MAC and physical layers, are managed by the LLSs, which are the client simulators and are specific for each technology, so that our simulation platform provides the presence of so many LSSs as technologies adopted in the investigated scenario (see figure 4).

The ULS and the LLSs are distinct executables and can run independently on different personal computers (PCs), nonetheless ULS and LLSs can communicate one with the
others, thus simulating the networks interworking, through the TCP sockets of the PC operating system.

In the following, a brief description of the ULS and the WLAN and UMTS LLSs simulators is given.

A. Upper layers simulator

The main tasks of the ULS are hereafter reported:

- it establishes the initial instant of each new traffic session, according to the statistics of the traffic class it belongs to, as well as its position within the investigated scenario;
- it generates the bit-flows up(down)loaded by each user in each traffic session according to the statistics of its class of traffic;
- it reproduces the transport protocol behavior, both UDP and New Reno TCP are implemented;
- it selects through which technology should each user connect to the network on the basis of user-defined rules and the available network information; it can also decide to reject a connection or to move it from a LLS to another (that is, from a given technology to another) at any time, thus simulating the network interworking;
- it finally collects all the simulation outcomes and generates the outputs (throughput, packet delivery delays,...) from an end-to-end point of view.

In our simulation platform each LLS manages its own time axis; the ULS, for its part, communicates to the LLSs the next instant in which some event concerning the upper levels happens (sessions begin, start of bit transfers, TCP timeouts, etc.). This way, when the time counter of a LLS reaches that instant, the related LLS simulation stops and a "call" to the ULS is performed asking for the event-related information and providing to the ULS, at the same time, information on the low levels events. After the ULS reply, the simulation of the calling LLSs restarts, and the consequent actions (packet queuing, transmission, ...) are taken.

The above described stop-and-wait procedure is the basis for the coordination among LLSs, which is obviously needed when simulating interworking networks: in case an ULS event is of interest for more than one LLS, no ULS reply is granted to the calling LLSs until all the interested LLSs have stopped waiting for it, then the reply is issued on the basis of the LSSs reports provided to the ULS. If follows that although the LLSs are not synchronized (they could be running on different PCs), the faster LLSs periodically stop and wait for the LLSs they are interworking with.

B. WLAN LLS

The WLAN network simulator (that is, the WLAN LLS) carefully reproduces the MAC protocols of both IEEE 802.11a [2], [3], and IEEE 802.11e [4] technologies as well as the IEEE802.11a PHY level behavior.

WLAN Medium Access Control Protocols. As for the MAC protocol of both IEEE 802.11a and IEEE 802.11e, here we considered the Distributed Coordinated Function (DCF) mode (see [2] for details) which is based on the Carrier Sensing Medium Access with Collision Avoidance (CSMA-CA) strategy; the hidden terminal problem is supposed to be negligible, hence the two-way handshake procedure [2] is assumed in the following.

Let us recall that IEEE802.11e specifications [4] define only the MAC level strategies, which can be combined with anyone of the IEEE802.11a, IEEE802.11b or IEEE802.11g PHY layers; here, when dealing with IEEE 802.11e we mean the combination of IEEE802.11e MAC level and IEEE802.11a PHY level.

In table II the values of the IEEE802.11e MAC protocol parameters adopted in our simulation have been reported (see [4] for details on their meaning).

WLAN Physical level Protocol. As for the PHY issues, let us recall that IEEE802.11a is based, at the Physical level, on eight operating modes adopting in the 5 GHz band the Orthogonal Frequency Division Multiplex Access (OFDM) to counteract the effects of frequency selective fading [3]. Each mode is characterized by a different combination of the modulation scheme and of the punctured convolutional code rate.

All IEEE 802.11a PHY level aspects (propagation, modulation, channel coding, ...) have carefully taken into account by the WLAN LLS, in particular:

- according to ETSI recommendation, the multipath channel is generated in a time and frequency correlated way following [11]. In this work we considered the channel model A (see [11]) that corresponds to a small office environment (the most challenging one) with 18 Rayleigh distributed paths. The time-correlated channel variations are taken into account considering an user speed of 3 km/h for the Clarke [12] Doppler spectrum of each path;
- the Auto Rate Fallback (ARF) [13] link adaptation algorithm is assumed to select the operation mode among the allowed (i.e., the combination of modulation scheme and coding rate); packets are discarded after 7 consecutive failed transmissions; hard decision decoding is assumed.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Voice</th>
<th>Web browsing</th>
<th>Ftp</th>
</tr>
</thead>
<tbody>
<tr>
<td>AIFS</td>
<td>1</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>CWmin</td>
<td>8</td>
<td>16</td>
<td>32</td>
</tr>
<tr>
<td>m</td>
<td>2</td>
<td>6</td>
<td>6</td>
</tr>
</tbody>
</table>
according to the technical specifications of most of commercially available WLAN devices, here we assumed 19dBm for EIRP, 2dB for the antennas' gains and 12dB for the receiver noise figure. The signal power has been assumed decaying with the fourth power of the distance.

\section*{C. UMTS LLS}

The UMTS simulation tool we developed reproduces the main characteristics of the TD-SCDMA radio interface \cite{1}, from Layer 1 to Layer 3.

Simulation step is assumed equal to the time slot duration (675 $\mu$s). Main features are:

- channel pathloss model is the Walfish-Ikegami: both line of sight and non line of sight conditions are considered; in the formula $K_1 + K_2 \log_{10}(d)$, $K_1$ and $K_2$ have arbitrary values, depending on the scenario; here we assumed $K_1=15.3$ and $K_2=37.6$;
- the shadowing is modelled by means of log normal random variables with zero mean and an exponential correlation function; the spread here assumed for the lognormal shadowing process is $\sigma = 5dB$;
- fast fading: different ITU channels are simulated (pedestrian, vehicular, indoor, both A and B); here the pedestrian channel model has been assumed with an user speed of 3.5Km/h.;
- both fast closed loop power control (rate 200 Hz) and slow outer loop power control are implemented; each bearer service is characterized by a proper value of initial target signal-to-interference ratio and different Transmitted Power Command (TPC) step sizes can be selected;
- multi-slot and multi-code combinations are supported.

Transport Channel is the service provided by physical layer to Layer 2 protocols; basic procedures are defined for MAC, providing a logical channel service, and RLC, providing a Radio Bearer service. Layer 2 entities work with larger time intervals: the transport block duration (10, 20, 40 or 80 ms).

MAC main task is the priority management of the shared resource among users while RLC provides segmentation functionality and different reliability modes: for CS speech connections RLC transparent mode is used and the quality of service is estimated by averaging bit error rate measurements over long periods (i.e., 0.5-1.0 s); in case of PS sessions the simulator evaluates whether the transport block belonging to a data packet is correctly received or not through the value of the related measured block error rate.

For PS calls a pair of RLC protocol instances in acknowledge mode is used, providing a reliable radio bearer service, including error correction by automatic retransmission.

At top of Layer 2, Radio Resource Control (RRC) block implements main Layer 3 procedures, i.e. Call Admission Control: a new radio link is successfully setup provided that the necessary OVSF codes are available, the estimated interference is less than a given threshold and the initial power required in Node B is available. Different classes of traffic are supported: CS AMR speech at 12.2 Kbps; PS Best Effort: three Unconstrained Delay Data bearer services at 64/64 Kbps, 64/144 Kbps and 64/384 Kbps are considered, where $x/y$ Kbps stands for $x$ Kbps for the uplink and $y$ Kbps for the downlink.

\section*{V. Traffic Settings and Merit Figures}

As previously mentioned, our investigation was focused on the opportunity to redirect on a WLAN connection those voice calls that cannot be served by the UMTS air interface; provided that the final user is equipped with a UMTS-WLAN dual mode terminal, here we assume that the voice traffic that would be blocked by the UMTS network is converted in VoIP traffic and then managed by the WLAN.

To evaluate the benefits of networks interworking, a comparison with the case of no redirection (and simple blocking) of the exceeding voice calls has been performed.

Following the investigated scenario and the assumptions made, four different traffic categories have been considered, with different burstiness and quality requirements:

- voice traffic: circuit switched bidirectional flows. This class is served by UMTS only;
- VoIP traffic: constant bit rate (CBR) bidirectional flows. This class is served by the WLAN only, hence VoIP users are located within the hot spot;
- web browsing traffic: characterized by bursts (packet calls) of small application-level packets followed by inactivity periods (reading time); users performing web browsing that are located within the hot spot coverage are served by the WLAN, otherwise by UMTS;
- FTP (file transfer protocol) traffic: bursty traffic characterized by requests of huge application-level packets followed by inactivity periods (reading time); users performing FTPs are supposed to be present only within the WLAN hot spot, taking advantage of the WLAN high speed connections.

For each kind of service (voice calls, web-browsing, FTPs), the merit figure investigated is the percentage of users experiencing a satisfactory service level, which depends on the particular kind of service:

- voice call: a voice user is satisfied if the call comes to the natural conclusion with an outage lower than 0.05% in each direction;
- VoIP: a VoIP user is satisfied if 98% of packets are received in less than 0.2 sec in each direction;
- web-browsing: the user is satisfied if 90% of packets are received in less than 5 sec;
- FTP: the user is satisfied if the average experienced throughput is higher than 800 kbit/s.

For the sake of conciseness we summarized in table III the traffic categories we considered (first column), their characteristics (second column), the related references (third column) and the satisfaction thresholds (fourth column).

The users’ satisfaction levels are evaluated as a function of the rate of arrival of new voice calls in the hot spot (voice\textsuperscript{HS}), having fixed the rates of arrival of voice and web-browsing background traffics (Voice\textsuperscript{back}, web-browsing\textsuperscript{back}) and
the rates of arrival of FTP and web-browsing traffics in the hot spot region (web-browsing\textsuperscript{HS}, FTP\textsuperscript{HS}) according to the values reported in the fifth column of table I.

It has to be pointed out that the rate of arrival of new voice calls in the hot spot, which is the variable parameter in our investigation, does not coincide with the rate of arrival of new VoIP sessions in the hot spot, since only those calls, including the background calls, that cannot be served by the UMTS air-interface are redirected to the WLAN (hence converted into VoIP flows).

In this paper, an ideal core network is assumed, with no packet loss and no delay added neither in downlink nor in uplink.

### VI. NUMERICAL RESULTS

Figures 5 to 8 show the impact on users satisfaction of the rate arrival of new voice calls in the 100x100 m\textsuperscript{2} squared area under investigation (see figure 1). The impact of redirection is compared with the case of simple call blocking, in both cases of IEEE 802.11a-UMTS and IEEE 802.11e-UMTS integrated networks.

In figure 5 the percentage of satisfied voice users is reported as a function of the rate of arrival of new voice calls. As expected, an increase of the voice service requests leads to a decrease of the percentage of satisfied voice users; it can be noted, however, that redirecting the calls rejected by UMTS to the WLAN, both in versions ‘a’ and ‘e’, allows to serve up to 20% more voice users.

It is interesting to observe that in the case of no call redirection the curve related to the satisfaction percentage is concave, whereas a convex trend is observed in the case of call redirection for both WLAN technologies. These behaviors can be easily explained: in the first case, in fact, as the number of voice service requests increases, the number of satisfied user remains almost constant, since the maximum number of voice calls that can be served by UMTS is fixed, whereas the number of blocked calls increases, thus originating the observed concave behavior of the satisfied user percentage curve.

In case of call redirection, on the contrary, as long as the WLAN capacity is not exceeded the number of satisfied users almost coincides with the number of voice service requests; afterwards, every further incoming call causes a degradation of the QoS perceived by already transmitting users, thus originating the convex behavior.

### TABLE III

**ADOPTED TRAFFIC CLASSES: PARAMETERS AND REQUIREMENTS FOR SATISFACTION.**

<table>
<thead>
<tr>
<th>Class</th>
<th>Characteristics</th>
<th>Refer to</th>
<th>Satisfaction thresholds</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice</td>
<td>Poissonian duration, 120 sec in average</td>
<td></td>
<td>Natural conclusion with outage lower than 0.05% in each direction</td>
</tr>
<tr>
<td>VoIP</td>
<td>CBR traffic, 80 bytes packets with rate 8 kbyte/s per direction</td>
<td>[14]</td>
<td>98% of packets received in less than 0.2 sec in each direction</td>
</tr>
<tr>
<td>Web B.</td>
<td>1 to 8 packet calls of 70 kbytes in average, divided into 1 to 30 packets each; 120 sec average reading time</td>
<td>[15] with $\alpha = 0.6$</td>
<td>90% of packets received in less than 5 sec</td>
</tr>
<tr>
<td>FTP</td>
<td>1 to 6 packet calls of 500 kbytes in average; 180 sec average reading time</td>
<td>Each packet call follows [16], with $\mu = 13.06$ and $\sigma = 0.12$</td>
<td>Average throughput of 800 kbit/s</td>
</tr>
</tbody>
</table>

![Fig. 5. Voice in the 500x500 squared meters area: users’ satisfaction.](image-url)
Future work should find a feasible and effective strategy to predict such a condition, thus giving optimal service to most of the users even in critical situations.

ACKNOWLEDGMENT

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REFERENCES

[4] IEEE 802.11 WG, IEEE 802.11e/D5.0