(50) Invention Title: PROVIDING VOICE AND DATA SERVICE FOR WIRELESS CELLULAR SUBSCRIBERS OPERATING IN A WIRELESS LOCAL AREA NETWORK

(57) Abstract: A system, method, and computer readable medium for providing a voice call for wireless cellular subscribers operating in a wireless local area network comprises receiving an add message (76) by a media gateway (16) from a mobile switching center (12), wherein the add message (76) informs the media gateway (16) to allocate a Real-Time Transport Protocol (RTP) Internet Protocol (IP) address and port, and receiving an acknowledge message (78) by the mobile switching center (12) from the media gateway (16), wherein the acknowledge message (78) includes the allocated RTP IP address and port.
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CROSS-REFERENCE TO RELATED APPLICATIONS

The present invention is related to and claims priority from United States provisional patent application number 60/568,891 filed on May 7, 2004, titled METHOD AND SYSTEM FOR PROVIDING VOICE SERVICE FOR WIRELESS CELLULAR SUBSCRIBERS OPERATING IN A WIRELESS LOCAL AREA NETWORK, United States provisional patent application number 60/627,932 filed on November 15, 2004, titled METHOD AND SYSTEM FOR PROVIDING VOICE SERVICE FOR WIRELESS CELLULAR SUBSCRIBERS OPERATING IN A WIRELESS LOCAL AREA NETWORK, and United States provisional patent application number 60/629,480 filed on November 19, 2004, titled METHOD AND SYSTEM FOR PROVIDING VOICE SERVICE FOR WIRELESS CELLULAR SUBSCRIBERS OPERATING IN A WIRELESS LOCAL AREA NETWORK, the entire contents of each of which are enclosed by reference herein.

FIELD OF THE INVENTION

The present disclosure relates generally to voice and data communications and, more particularly, to providing voice and data service for wireless cellular subscribers operating in a wireless local area network.

BACKGROUND OF THE INVENTION

There are various communications networks, such as mobile networks and wireless local area networks (WLANs), that provide voice and data services. A handset utilized in one of the networks typically will not operate in the other network due to spectrum and protocol incompatibilities, for example. Further, services that may be utilized in one of the networks may not be available in the other network due to the aforementioned incompatibilities and due to the inability for the systems to provide transparency between one another.

Therefore, what is needed is a system, method, and computer readable medium for providing voice and data service for wireless cellular subscribers operating in a wireless local area network that overcomes the problems and limitations described above.
The present invention provides a voice over WLAN service that will permit users to roam seamlessly between mobile networks and WLANs with a single handset for voice and data services. When served by a WLAN, subscribers experience high-performance mobile voice and data services, and a potential benefit from lower service cost. In one embodiment, a method for providing a voice call for wireless cellular subscribers operating in a wireless local area network comprises receiving an add message by a media gateway from a mobile switching center (MSC), wherein the add message informs the media gateway to allocate a Real-Time Transport Protocol (RTP) Internet Protocol (IP) address and port, and receiving an acknowledge message by the MSC from the media gateway, wherein the acknowledge message includes the allocated RTP IP address and port.

In another embodiment, a method for providing a voice call for wireless cellular subscribers operating in a wireless local area network comprises receiving a first message by a first module, allocating a RTP IP address and port, receiving a second message by a second module from the first module, wherein the second message includes the allocated RTP IP address and port, and receiving a third message by a third module from the second module, wherein the third message includes a parameter comprising the allocated RTP IP address and port from the first module for a current Voice over IP (VoIP) call.

In a further embodiment, a computer readable medium comprises instructions for: receiving a handoff request message from a MSC to a base station controller (BSC), wherein the handoff request message includes: Real-Time Transport RTP IP information from an MSC side for a VoIP call, and an International Mobile Subscriber Identity (IMSI), and receiving a handoff request acknowledge message from the BSC to the MSC, wherein the handoff request acknowledge message includes RTP IP information from a BSC side for the VoIP call.

In yet another embodiment, a system for providing a voice call for wireless cellular subscribers operating in a wireless local area network comprises a module (for example a call server or other module or node), and a BSC coupled to the module, wherein a parameter that facilitates a VoIP call is transferred between the module and BSC, wherein the parameter includes an RTP IP address and port.

In yet a further embodiment, a parameter adapted to facilitate a Voice over IP call for wireless cellular subscribers operating in a wireless local area network comprises an element identifier, a length, a most significant byte of a RTP IP address, a second byte of the RTP IP address, a third byte of the RTP IP address, and a least significant byte of the RTP IP address.
 sending a handoff request message from a mobile switching center (MSC), wherein the handoff request message includes: Real-Time Transport Protocol (RTP) Internet Protocol (IP) information from a first side for a Voice over IP (VoIP) call, and an International Mobile Subscriber Identity (IMSI).

In yet another embodiment, a method for providing a voice call for wireless subscribers operating in a wireless local area network comprises sending a message including: Real-Time Transport Protocol (RTP) Internet Protocol (IP) information for a part of a Voice over IP (VoIP) call, and an International Mobile Subscriber Identity (IMSI).

10 BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 depicts a voice over WLAN system in accordance with a preferred embodiment of the present invention;

Figure 2 depicts a WLAN to GSM call flow in accordance with a preferred embodiment of the present invention;

Figure 3 depicts a WLAN to WLAN call flow in accordance with a preferred embodiment of the present invention;

Figure 4 depicts a GSM to WLAN call flow in accordance with a preferred embodiment of the present invention;

Figure 5 depicts a GSM handover to WLAN call flow in accordance with a preferred embodiment of the present invention;

Figure 6 depicts a WLAN handover to GSM call flow in accordance with a preferred embodiment of the present invention;

Figure 7 depicts a WLAN to PSTN call flow in accordance with a preferred embodiment of the present invention; and

Figure 8 depicts a PSTN to WLAN call flow in accordance with a preferred embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

The present invention provides a voice over WLAN service that permits users to roam seamlessly between mobile networks and WLANs with a single handset for voice and data services. Important components of the system of the present invention include a call server, and a media gateway that includes the Real-Time Transport Protocol (RTP), the Voice over Internet Protocol (VoIP) interface. More specifically, RTP is the Internet-standard protocol for the
data, including audio and video. It can be used in PCT/US2005/015984 and as well as interactive services such as Internet telephony.

In the present invention, utilizing RTP allows media streams to be directly connected between the handset/access point and the Media Gateway through the Internet. The present invention further provides a Voice over WLAN (VoWLAN) feature that is implemented by utilizing a handset adapted to connect to a WLAN using an 802.x (802.11, 802.16, etc.), Bluetooth or Ultra Wide Band (UWB) standard, for example. Through the WLAN, the handset is in communication with a Base Station Controller (BSC) that connects to the call server of the present invention via the A-interface that provides signaling functionality. The bearer connection is carried over VoIP. In other embodiments of the present invention, the A-interface functionality is transferred via VoIP.

Referring now to Figure 1, a VoWLAN system 10 of the present invention is depicted. The system 10 includes a call server 14 (which includes a MSC and a Visitor Location Register (VLR)) and a media gateway 16, and a BSC 18 that is connected to the call server 14 via the A-interface 20. A handset 24 is adapted to connect to the media gateway 16 and the BSC 18 via an IP network 22. More specifically, the handset 24 utilizes Direct Transfer Application Part over IP (DTAPoIP) 26 to connect to the BSC 18 via the IP network 22, and utilizes VoIP 28 to connect to the media gateway 16 via the IP network 22. Further, the call server 14 and the media gateway 16 may communicate via the IP network 22.

The call server 14 and the media gateway 16 may further be coupled to various components and networks 30-34 via various interfaces 36-40, respectively to provide further services and functionality. More specifically, the call server 14 may be coupled to components 30 via mobility application part (MAP)/Customized Application for Mobile network Enhanced Logic (CAMEL) in order to provide SS7 services, while the media gateway 16 may be coupled to a network 32 (such as a traditional wireless network) via an A/Time Division Multiplex (TDM) interface 38 that enables many devices to communicate over the interface, and to a TDM network 34 via an Integrated Services Digital Network User Part (ISUP)/TDM interface 40.

The VoWLAN feature of the present invention utilizes various supplementary services which are value-added services, such as call forwarding, call waiting, and follow me, for example. The supplementary services preferably utilized by the present invention are depicted below in Table 1.
<table>
<thead>
<tr>
<th>Calling Line Identification Presentation (CLIP)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Line Identification Restriction (CLIR)</td>
</tr>
<tr>
<td>Connected Line Identification Presentation (CoLP)</td>
</tr>
<tr>
<td>Call Forwarding Unconditional</td>
</tr>
<tr>
<td>Call Forwarding on Mobile Subscriber Busy</td>
</tr>
<tr>
<td>Call Forwarding on No Reply</td>
</tr>
<tr>
<td>Call Forwarding on Mobile Subscriber Not Reachable</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
</tr>
<tr>
<td>Call Waiting</td>
</tr>
<tr>
<td>Call Hold</td>
</tr>
<tr>
<td>Multi Party Service</td>
</tr>
<tr>
<td>Barring of All Outgoing Calls</td>
</tr>
<tr>
<td>Barring of Outgoing International Calls</td>
</tr>
<tr>
<td>Barring of Outgoing International Calls except those directed to the Home PLMN Country</td>
</tr>
<tr>
<td>Barring of All Incoming Calls</td>
</tr>
<tr>
<td>Barring of Incoming Calls when Roaming Outside the Home PLMN Country</td>
</tr>
<tr>
<td>Explicit Call Transfer</td>
</tr>
<tr>
<td>Enhanced Multi-Level Precedence and Pre-emption</td>
</tr>
<tr>
<td>Completion of calls to busy subscribers</td>
</tr>
</tbody>
</table>

**Table 1**

These supplemental services are preferably required. However, in an embodiment of the present invention, a greater or lesser number of the services, and potentially differing services than those depicted in Table 1, may be utilized without departing from the scope of the present invention.

The present invention further discloses a new parameter named “RTP Information.” This name is used herein as an example and may be changed without departing from the scope of the present invention. The “RTP Information” parameter facilitates a VoIP call between the BSC 18 and the call server 14 and is thus passed over the A-interface 20. This new parameter is preferably an optional parameter for the following messages which will be described in further detail below: Assignment Request, Assignment Complete, Handover Request and Handover Request Acknowledge. Other messages may also utilize the parameter without departing from the scope of the present invention.

Referring now to Table 2 below, the encoding of the new parameter “RTP IPv4 Information” is depicted. This name is used herein as an example and may be changed without departing from the scope of the present invention. IPv4 stands for Internet Protocol version 4 and is the original standard set up for handling IP addresses when the Internet was initial developed by the Defense Advanced Research Projects Agency (DARPA) in the early 1970s. IPv4 uses a 32 bit address field which provides for 4,294,967,296 unique Internet addresses. In a
In another embodiment of the present invention, the encoding of the new parameter “RTP IPv6 Information” is supported and presented in Table 3 below.

<table>
<thead>
<tr>
<th>Element Identifier</th>
<th>Octet 1</th>
<th>Octet 2</th>
<th>Octet 3</th>
<th>Octet 4</th>
<th>Octet 5</th>
<th>Octet 6</th>
<th>Octet 7</th>
<th>Octet 8</th>
<th>Octet 9</th>
</tr>
</thead>
<tbody>
<tr>
<td>Length</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTP IP Address (the most significant Byte)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTP IP Address (Second Byte)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
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<tr>
<td>RTP IP Address (Third Byte)</td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTP IP Address (the least significant Byte)</td>
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<td></td>
<td></td>
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<td></td>
<td></td>
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<tr>
<td>RTP Port Address (First Part)</td>
<td></td>
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<td></td>
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<td></td>
<td></td>
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<tr>
<td>RTP Port Address (Second Part)</td>
<td></td>
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<td></td>
<td></td>
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<td></td>
<td></td>
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<td></td>
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<tr>
<td>Codec 1</td>
<td></td>
<td></td>
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<td></td>
<td></td>
</tr>
<tr>
<td>Codec 2</td>
<td></td>
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</tr>
</tbody>
</table>

Table 2

Since IPv4 addresses are nearly consumed, IPv6 was developed in response to this situation. IPv6 allocates 128 bits to map the Internet address space. The number of bits were quadrupled from IPv4’s 32 bits to insure that this address space would not run out in the foreseeable future. In addition to the large number of IP addresses, IPv6 provides for better handling of voice than IPv4 which was not initially set up to handle it. As such, phone conversations over the Internet will increase in clarity.
The present invention supports various codecs and payload which are depicted in Table 4 below.

<table>
<thead>
<tr>
<th>Bit</th>
<th>Decimal</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0 0</td>
<td>0</td>
<td>u-law PCM</td>
</tr>
<tr>
<td>0 0 0</td>
<td>8</td>
<td>A-law PCM</td>
</tr>
<tr>
<td>0 0 0 1 0 0 1 0</td>
<td>18</td>
<td>G.729</td>
</tr>
<tr>
<td>0 1 1 0 0 0 0 1</td>
<td>97</td>
<td>12.2 kbps AMR</td>
</tr>
<tr>
<td>1 1 1 1 1 1 1 1</td>
<td>255</td>
<td>Null. Include in second Payload Type field if only one payload type is offered for the connection.</td>
</tr>
</tbody>
</table>

The VoWLAN feature is a system wide mobile feature. As such, the following call types are affected: mobile-mobile (if one party is connected through the BSC 18), landline-mobile (if the mobile is connected through the BSC), and mobile-landline (if the mobile is connected through the BSC). It should be noted that maintenance related messages (such as reset, block, unlock, etc.) are not supported over the base station subsystem management application part (BSSMAP) which is a protocol that conveys general base station subsystem (BSS) control information between the mobile switching center (MSC) and the BSS. An example is the allocation of traffic channels between the MSC and the BSS. These maintenance related messages are not needed because the voice bearer channel is no longer being used as voice is provided via VoIP in the present invention.

Referring now to Figure 2, a WLAN to GSM call flow 50 is depicted. The call flow 50 utilizes the following elements: a mobile station (for example, an originating mobile station) 24, an access point 52, a BSC 18, a call server 14, a media gateway (MGW) 16, a Home Location
WO 2005/112410 A1, a BSS 50, and a mobile station (for example, a VoPCT/US2005/015984 station) 58. The VoWLAN feature of the present invention is centered around the CALL SERVER 14 and the BSC 18, and more specifically around the WSS’s MSC. This feature will apply in situations where a Gateway MSC (GMSC) and a Visiting MSC (VMSC) is the same node. The 5 call flow 50 utilizes a combined GMSC/VMSC node. In situations where the GMSC and the VMSC are separated, external ISUP messages (towards the VMSC) are used to provide the functionality of the present invention.

The messaging between the MS 24 and the MGW 16, and more specifically, the messaging between the Call server 14 and the BSC 18, occurs via the IP network 22. 10 Messaging between the MS 24 and the BSC 18 is further performed utilizing DTAPoIP 26, while messaging between the MS 24 and the AP (such as a WiFi, Bluetooth, WiMax, etc. access point) 24 is further performed utilizing VoWiFi, VoBluetooth, VoWiMax, etc. In a further embodiment, the messaging can utilize other protocols than those described above without departing from the scope of the present invention.

Referring again to Figure 2, a CM Service Request message 60 is received by the call server 14 which responds to the BSC 18 with a Connection Confirm message 62 and an Authenticate message 64. The BSC 18 then sends an Authenticate Response message 66. The call server 14 may, at this point, optionally send a Start Security message 68 to the BSC 18 which would then respond with a Security Complete message 70. A Setup message 72 is sent 20 from the BSC 18 to the CALL SERVER 14 which responds with a Call Proceeding message 74.

The call server 14 sends an Add message (or first message) 76 to the MGW (or first module) 16 informing the MGW to allocate an RTP IP address and port. An Acknowledge message (or second message) 78 to the call server (or second module) 14 includes the allocated RTP IP address and port. An Assignment Request message (or third message) 80 is sent from the call server 14 to the BSC (or third module) 18 including the new “RTP Information” parameter which contains the allocated RTP IP address and port from the WSS12/MGW16 for the current VoIP call. When the BSC 18 receives the Assignment Request message 80 an Assignment Complete message (or fourth message) 82 is sent to the call server 14. A new “RTP Information” parameter is included in this message and contains an RTP IP address and port 30 from the BSC18/AP (or fifth module) 52 for the VoIP call. In an alternate embodiment, the second module may be a module or node independent of the call server 14 that is able to receive and process all messages (including the third message, for example). This node may be one of the nodes depicted in Figure 1 or another node that can communicate with at least one of the nodes in Figure 1.
A Send Routing Information (SRI) message 84 is sent from the PCT/US2005/015984 to the HLR 54 in order to query the HLR 54 for the location of the terminating mobile 58, and a Provide Roaming Number (PRN) message 86 is returned to the call server 14, followed by a PRN Acknowledge message 88 to the HLR 54 and a SRI Acknowledge message 90 to the call server 14. A Page message 92 is then sent from the call server 14 to the BSS 56 and then, via Page message 94, to the terminating MS 58. The MS 58 sends a Channel Request message 96 to the BSS 56 which responds with an Immediate Assignment message 98 to the MS 58. A Page Response message 100 is sent to the BSS 56 which then sends a MS Connect Established message 102 to the CALL SERVER 14. The call server 14 then sends a Start Security message 104 to the BSS 56 which sends a Security Control Command message 110 to the MS 58 and receives a Security Control Response message 112. While this messaging is occurring, an Add message 106 is sent from the call server 14 to the MGW 16 which responds with an Acknowledge message 108.

A Setup message 114 is then sent to the MS 58 from the call server 14 which responds with a Call Confirm message 116. The call server 14 then sends an Allocate Channel message 118 to the BSS 56 which sends an Assignment Command message 120 to the MS 58 and receives an Assignment Complete message 122. The BSS 56 then sends an Allocation Complete message 124 to the call server 14 which further receives an Alerting message 126 from the MS 58. The call server 14 then sends an Alerting message 128 to the BSC 18, receives a Connect message 130 from the MS 58, and sends a Connect message 132 to the BSC 18. A Mod (or Modification) message 134 is sent from the call server 14 to the MGW 16 which responds with an Acknowledge message 136. A Connect Acknowledge message 138 is sent from the BSC 18 to the call server 14 and further a Connect Acknowledge message 140 is sent to the MS 58. With the call now established, the voice path between the MS 24 and the AP 52 is providing voice over 802.x, Bluetooth, or UWB, for example, as indicated by “network” 28, while the voice path between the AP 52 and the MGW 16 is providing voice via RTP, as indicated by the “network” 140.

Referring now to Figure 3 a WLAN to WLAN call flow 150 is depicted. The call flow 150 utilizes the following elements: a mobile station (for example, an originating mobile station) 24, an access point 52, a BSC 18, a call server 14, a media gateway (MGW) 16, a Home Location Register (HLR) 54, a BSC 152, an access point 154, and a mobile station (for example, a terminating mobile station) 58. The VoWLAN feature of the present invention is centered around the call server 14 and the BSCs 18 and 152, and more specifically around the call server 14. This feature will apply in situations where a Gateway MSC (GMSC) and a Visiting MSC
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(VMSC) is the same node. The call flow 150 utilizes a combined PCT/US2005/015984 file. In situations where the GMSC and the VMSC are separated, external ISUP messages (towards the VMSC) are used to provide the functionality of the present invention.

The messaging between the MS 24 and the MS 58 occurs via the IP network 22. Messaging between the MS 24 and the BSC 18 is further performed utilizing DTAPoIP 26, while messaging between the MS 24 and the AP (such as a WiFi, Bluetooth, WiMax, etc. access point) 52 and between the MS 58 and the AP 154 is further performed utilizing VoWiFi, VoBluetooth, VoWiMax, etc. In a further embodiment, the messaging can utilize other protocols than those described above without departing from the scope of the present invention.

Referring again to Figure 3, a CM Service Request message 156 is received by the CALL SERVER 14 which responds to the BSC 18 with a Connection Confirm message 158 and an Authenticate message 160. The BSC 18 then sends an Authenticate Response message 162. The call server 14 may, at this point, optionally send a Start Security message 164 to the BSC 18 which would then respond with a Security Complete message 166. A Setup message 168 is sent from the BSC 18 to the CALL SERVER 14 which responds with a Call Proceeding message 174.

The CALL SERVER 14 sends an Add message 170 to the MGW 16 informing the MGW to allocate a RTP IP address and port. An Acknowledge message 172 to the call server 14 includes the allocated RTP IP address and port. An Assignment Request message 176 is sent from the call server 14 to the BSC 18 including the new “RTP Information” parameter which contains the allocated RTP IP address and port from the call server 14/MGW16 for the current VoIP call. When the BSC 18 receives the Assignment Request message 176 an Assignment Complete message 178 is sent to the call server 14. A new “RTP Information” parameter is included in this message and contains an RTP IP address and port from the BSC18/AP 52 for the VoIP call.

An SRI message 180 is sent from the call server 14 to the HLR 54 in order to query the HLR 54 for the location of the terminating mobile 58, and a PRN message 182 is returned to the call server 14, followed by a PRN Acknowledge message 184 to the HLR 54 and a SRI Acknowledge message 186 to the call server 14. A Page message 188 is then sent from the call server 14 to the BSC 152, which then sends a MS Connect Established message 190 to the call server 14. The CALL SERVER 14 then sends a Start Security message 192 to the BSC 152, and an Add message 194 to the MGW 16 informing the MGW to allocate an RTP IP address and port. An Acknowledge message 196 to the call server 14 includes the allocated RTP IP address

10
An Assignment Request message 202 is sent from the call server 14 to the BSC 152 including the new “RTP Information” parameter which contains the allocated RTP IP address and port from the call server 14/MGW16 for the current VoIP call. When the BSC 152 receives the Assignment Request message 202 an Assignment Complete message 204 is sent to the call server 14. A new “RTP Information” parameter is included in this message and contains an RTP IP address and port from the BSC 152/AP 154 for the VoIP call.

The call server 14 then receives an Alerting message 206 from the BSC 152, and sends an Alerting message 208 to the BSC 18. The call server 14 further receives a Connect message 210 from the BSC 152 and sends a Connect message 212 to the BSC 18. A Mod message 214 is sent from the call server 14 to the MGW 16 which responds with an Acknowledge message 216. A Connect Acknowledge message 218 is sent from the BSC 18 to the call server 14 and further, a Connect Acknowledge message 220 is sent to the BSC 152. With the call now established, the voice path between the MS 24 and the AP 52 is providing voice over 802.11 or 802.16 or Bluetooth, for example, as indicated by “network” 28, while the voice path between the AP 52 and the AP 154 is providing voice via RTP, as indicated by the “network” 140.

Referring now to Figure 4 a GSM to WLAN call flow 230 is depicted. The call flow 230 utilizes the following elements: a mobile station (for example, an originating mobile station) 24, a BSS 232, a call server 14 a media gateway (MGW) 16, a Home Location Register (HLR) 54, a BSC 152, an access point 154, and a mobile station (for example, a terminating mobile station) 58. The VoWLAN feature of the present invention is centered around the call server 14 and the BSC 152, and more specifically around the WSS’s 12 MSC. This feature will apply in situations where a Gateway MSC (GMSC) and a Visiting MSC (VMSC) is the same node. The call flow 230 utilizes a combined GMSC/VMSC node. In situations where the GMSC and the VMSC are separated, external ISUP messages (towards the VMSC) are used to provide the functionality of the present invention. The messaging between the call server 14 and the BSC 152 occurs via the IP network 22.

Referring again to Figure 4, a CM Service Request message 234 is received by the BSS 232 from the MS 24 and a CM Service Request message 236 is received by the call server 14 which responds to the BSS 232 with a Connection Confirm message 238, and to the MS with a Connection Confirm message 240. An Authenticate message 242 is sent from the call server 14 to the BSS 232 which sends an Authenticate message 244 to the MS 24. The MS 24 sends an Authenticate Response message 246 from the MS 24 to the BSS 232 which sends an
At time 

optionally send a Start Security message 250 to the BSS 232 which sends a Security Control message 252 to the MS 24. A Security Control Response message 254 is sent to the BSS 232, which then sends a Security Complete message 256 to the call server 14. A Setup message 258 is then sent from the MS 18 to the call server 14 which responds with a Call Proceeding message 264 to the MS 24.

The call server 14 sends an Add message 260 to the MGW 16 informing the MGW to allocate an RTP IP address and port. An Acknowledge message 262 to the call server 14 includes the allocated RTP IP address and port. An Allocate Channel message 266 is sent from the call server 14 to the BSS 232 and an Allocate Assignment message 268 is sent to the MS 24. The MS 24 sends an Assignment Complete message 270 to the BSS 232 which then sends an Allocation Complete message 272 to the call server 14. An SRI message 274 is sent from the call server 14 to the HLR 54 in order to query the HLR 54 for the location of the terminating mobile 58, and a PRN message 276 is returned to the call server 14, followed by a PRN Acknowledge message 278 to the HLR 54 and a SRI Acknowledge message 280 to the call server 14. A Page message 282 is then sent from the call server 14 to the BSC 152, which then sends a MS Connect Established message 284 to the call server 14. The call server 14 then sends a Start Security message 286, a Setup message 288 to the BSC 152, and an Add message 290 to the MGW 16 informing the MGW to allocate a RTP IP address and port. An Acknowledge message 292 sent to the call server 14 includes the allocated RTP IP address and port. A Call Confirm message 294 is then sent to the call server 14.

An Assignment Request message 296 is sent from the call server 14 to the BSC 152 including the new “RTP Information” parameter which contains the allocated RTP IP address and port from the call server 14/MGW16 for the current VoIP call. When the BSC 152 receives the Assignment Request message 296 an Assignment Complete message 298 is sent to the call server 14. A new “RTP Information” parameter is included in this message and contains an RTP IP address and port from the BSS 232 for the VoIP call.

The call server 14 then receives an Alerting message 300 from the BSC 152, and sends an Alerting message 302 to the MS 24. The call server 14 further receives a Connect message 304 from the BSC 152 and sends a Connect message 306 to the MS 24. A Mod message 308 is sent from the call server 14 to the MGW 16 which responds with an Acknowledge message 310. A Connect Acknowledge message 312 is sent from the BSC MS 24 to the call server 14 and further, a Connect Acknowledge message 314 is sent to the BSC 152. With the call now
established, the voice path between the call server 14 and the BSC 18 is via RTP, as indicated by the "network" 140.

Referring now to Figure 5 a GSM handover to WLAN call flow 320 is depicted. The call flow 320 utilizes the following elements: a mobile station (for example, an originating mobile station) 24, an Access Point (AP) 52, a BSC 18, a WSS (which includes an MSC and a Visitor Location Register (VLR)) 12, a media gateway (MGW) 16, a Home Location Register (HLR) 54, and a serving BSS 322. The VoWLAN feature of the present invention is centered around the call server 14 and the BSC 18, and more specifically around the WSS's 12 MSC. This feature will apply in situations where a Gateway MSC (GMSC) and a Visiting MSC (VMSC) is the same node. The call flow 320 utilizes a combined GMSC/VMSC node. In situations where the GMSC and the VMSC are separated, external ISUP messages (towards the VMSC) are used to provide the functionality of the present invention. The messaging between the MS 24 and the MGW 16 occurs via the IP network 22.

Referring again to Figure 5, a Handoff (HO) Required message 324 is sent from the serving BSS 322 to the call server 14. An Add message 326 and Mod messages 328 and 330 are sent from the call server 14 to the MGW 16. A HO Request message, which is sent from the call server 14 to the BSC 18, includes the new "RTP Information" parameter that contains the RTP IP address and port (or information) from the call server 14 side for this VoIP call. The subscriber's International Mobile Subscriber Identity (IMSI) is required to be included in the message. The BSC 18 then sends back a HO Request Acknowledge message 334 which includes the new "RTP Information" parameter that contains the RTP IP address and port from the BSC 18 side for this VoIP call.

A HO Command message 336 is sent from the call server 14 to the serving BSS 322 which then sends a HO Command message 338 to the MS 24. A HO Detect message 340 is sent from the BSC 18 to the call server 14 which sends Mod messages 342 and 346 to the MGW 16. An HO Complete message 344 is then sent to the call server 14 which sends a Clear Command message 348 to the serving BSS 322. The BSS 322 sends a Clear Command message 350 to the call server 14 which then sends a Sub message 352 to the MGW 16.

In one embodiment of the present invention, a method for providing a voice call for wireless cellular subscribers operating in a wireless local area network comprises sending a message including: RTP IP information for a part of a VoIP call and an IMSI. The method further comprises receiving a message including RTP IP information for another part of the VoIP call. The part and another part of the VoIP call refer to at least one of a call center 14 perspective and a BSC 18 perspective.
Now to Figure 6 a WLAN handover to GSM call flow depicted the call flow 360 utilizes the following elements: a BSC 18, a call server 14, a media gateway (MGW) 16, a Home Location Register (HLR) 54, a serving BSS 322, and a mobile station (for example, a terminating mobile station) 58. The VoWLAN feature of the present invention is centered around the call server 14 and the MGW 16, and more specifically around the call server 14. This feature will apply in situations where a Gateway MSC (GMSC) and a Visiting MSC (VMSC) is the same node. The call flow 320 utilizes a combined GMSC/VMSC node. In situations where the GMSC and the VMSC are separated, external ISUP messages (towards the VMSC) are used to provide the functionality of the present invention. The messaging between the BSC 18 and the MGW 16 occurs via the IP network 22.

Referring again to Figure 6, a Handoff (HO) Required message 362 is sent from the BSC 18 to the call server 14 which sends an Add message 364 to the MGW 16. The Add message 364 is an indication to the MGW 16 to allocate a RTP IP address and port. An Acknowledge message 366 returned to the call server 14 from the MGW 16 includes the RTP IP Address and port. Mod messages 368 and 370 are sent from the call server 14 to the MGW 16 and a HO Request message is sent from the call server 14 to the serving BSS 322. The serving BSS 322 returns a HO Request Acknowledge message 374 to the WSS which sends a HO Command message 376 to the BSC 18. The BSC 18 then sends a HO Command message 378 to the MS 58.

A HO Detect message 380 is then sent from the serving BSS 322 to the call server 14 which sends Mod messages 382 and 384 to the MGW 16. An HO Complete message 386 is then sent from the serving BSS 322 to the call server 14 which sends a Clear Command message 348 to the BSC 18. The BSC 18 sends a Clear Command message 390 to the call server 14 which then sends a Sub message 392 to the MGW 16.

Referring now to Figure 7, a WLAN to PSTN call flow 400 is depicted. The call flow 400 utilizes the following elements: a BSC 18, a WSS (which includes an MSC and a Visitor Location Register (VLR)) 12, a media gateway (MGW) 16, a Home Location Register (HLR) 54, and a Public Switch Telephone Network (PSTN) 402. The VoWLAN feature of the present invention is centered around the call server 14 and the BSCs 18 and 152, and more specifically around the call server 14. This feature will apply in situations where a Gateway MSC (GMSC) and a Visiting MSC (VMSC) is the same node. The call flow 150 utilizes a combined GMSC/VMSC node. In situations where the GMSC and the VMSC are separated, external ISUP messages (towards the VMSC) are used to provide the functionality of the present invention. The messaging between the BSC 18 and the MGW 16 occurs via the IP network.
WO 2005/112410 again to Figure 7, a CM Service Request message isPCT/US2005/015984 the call server 14 which responds to the BSC 18 with a Connection Confirm message 406 and an Authenticate message 408. The BSC 18 then sends an Authenticate Response message 410. The call server 14 may, at this point, optionally send a Start Security message 412 to the BSC 18 which would then respond with a Security Complete message 414. A Setup message 416 is sent from the BSC 18 to the call server 14 which responds with a Call Proceeding message 422.

The call server 14 sends an Add message 418 to the MGW 16 informing the MGW to allocate a RTP IP address and port. An Acknowledge message 420 to the call server 14 includes the allocated RTP IP address and port. An Assignment Request message 424 is sent from the call server 14 to the BSC 18 including the new “RTP Information” parameter which contains the allocated RTP IP address and port from the call server 14/MGW16 for the current VoIP call. When the BSC 18 receives the Assignment Request message 424 an Assignment Complete message 426 is sent to the call server 14. A new “RTP Information” parameter is included in this message and contains an RTP IP address and port from the BSC18/AP 52 for the VoIP call.

The call server 14 sends an Add message 428 to the MGW 16 which returns an Acknowledge message 430. An Initial Address Message (IAM) 432 is then sent from the call server 14 to the PSTN 402 which responds with an Address Complete Message (ACM) 434. An alerting message 436 is then sent form the call server 14 to the BSC 18. The PSTN further sends an Answer Message (ANM) 438 to the call server 14 which sends a Connect message to the BSC 440. A Mod message 442 is sent from the call server 14 to the MGW 16 which responds with an Acknowledge message 444. A Connect Acknowledge message 446 is then sent from the BSC 18 to the call server 14 and further.

Referring now to Figure 8 a PSTN to WLAN call flow 450 is depicted. The call flow 450 utilizes the following elements: a PSTN 402, a call server 14, a media gateway (MGW) 16, a Home Location Register (HLR) 54, a BSC 18, an access point 52, and a mobile station (for example, a terminating mobile station) 24. The VoWLAN feature of the present invention is centered around the call server 14 and the MGW 16, and more specifically around the call server 14. This feature will apply in situations where a Gateway MSC (GMSC) and a Visiting MSC (VMSC) is the same node. The call flow 230 utilizes a combined GMSC/VMSC node. In situations where the GMSC and the VMSC are separated, external ISUP messages (towards the VMSC) are used to provide the functionality of the present invention. The messaging between the call server 14 and the BSC 18 occurs via the IP network 22.

Referring again to Figure 8, an IAM 452 is received from the PSTN 402 to the call server 14. The call server 14 sends an Add message 454 to the MGW 16 which responds with an
An SMS message 450 is sent from the call server 14 in order to query the HLR 54 for the location of the MS 24, and a PRN message 460 is returned to the call server 14, followed by a PRN Acknowledge message 462 to the HLR 54 and a SRI Acknowledge message 464 to the call server 14. A Page message 466 is then sent from the call server 14 to the BSC 18, which then sends a MS Connect Established message 468 to the call server 14. The call server 14 then sends a Start Security message 470 and an Add message 472 to the MGW 16 informing the MGW to allocate a RTP IP address and port. An Acknowledge message 474 sent to the call server 14 includes the allocated RTP IP address and port. A Call Confirm message 478 is then sent to the call server 14.

An Assignment Request message 480 is sent from the call server 14 to the BSC 18 including the new “RTP Information” parameter which contains the allocated RTP IP address and port from the call server 14/MGW16 for the current VoIP call. When the BSC 18 receives the Assignment Request message 480 an Assignment Complete message 482 is sent to the call server 14. A new “RTP Information” parameter is included in this message and contains an RTP IP address and port from the BSC 18 for the VoIP call.

The call server 14 then receives an Alerting message 484 from the BSC 18, and sends an ACM 486 to the PSTN 402. The call server 14 further receives a Connect message 488 from the BSC 18 and sends an ANM 490 to the PSTN 402. A Mod message 492 is sent from the call server 14 to the MGW 16 which responds with an Acknowledge message 494. A Connect Acknowledge message 496 is sent from the call server 14 to the BSC 18.

In one embodiment of the present invention, a computer readable medium (or software) comprises instructions for sending a handoff request message from a MSC, wherein the handoff request message includes: RTP IP information from a first side (associated with the MSC) for a VoIP call, and an IMSI. The computer readable medium of the present invention is preferably stored on the call server 14 but may be stored in one or more other modules or nodes depicted or not depicted in Figure 1. The computer readable medium further comprises instructions for: receiving a handoff request acknowledge message at the MSC, wherein the handoff request acknowledge message includes RTP IP information from a second side (associated with at least one of: a base station controller and a base station subsystem) for the VoIP call.

Although an exemplary embodiment of the system and method of the present invention has been illustrated in the accompanied drawings and described in the foregoing detailed description, it will be understood that the invention is not limited to the embodiments disclosed, but is capable of numerous rearrangements, modifications, and substitutions without departing from the spirit of the invention as set forth and defined by the following claims. For example,
the system 10 can be performed by one or more of a PCT/US2005/015984 protocols described or not described herein and/or in a distributed architecture. For example, all or part of the functionality associated with the call server 14 may be included within or co-located with the media gateway 16 or the BSC 18, or in a new node that can be introduced in the network and can communicate with any other node and support all messages and appropriate protocols for example, or within a module or node not depicted in Figures 1-8. Also, the call server 14 and the media gateway 16 can be combined into one module and perform all or part of the functionality described herein. Further, the functionality described herein may be performed at various times and in relation to various events, internal or external to the modules or components. Also, the information sent between various modules, can be sent between the modules via at least one of a data network, the Internet, a voice network, an Internet Protocol network, a wireless source, a wired source and/or via plurality of protocols.
WHAT IS CLAIMED IS:

1. A method for providing a voice call for wireless cellular subscribers operating in a wireless local area network, comprising:
   receiving an add message by a media gateway from a mobile switching center (MSC), wherein the add message informs the media gateway to allocate a Real-Time Transport Protocol (RTP) Internet Protocol (IP) address and port; and
   receiving an acknowledge message by the MSC from the media gateway, wherein the acknowledge message includes the allocated RTP IP address and port.

2. The method of claim 1 comprising receiving an assignment request message by a base station controller (BSC) from the MSC, wherein the assignment request message includes an RTP information parameter.

3. The method of claim 2, wherein the RTP information parameter includes the allocated RTP IP address and port from the media gateway for a current Voice over IP (VoIP) call.

4. The method of claim 2 comprising receiving an assignment complete message by the MSC from the BSC, wherein the assignment complete message includes an RTP information parameter.

5. The method of claim 4, wherein the RTP information parameter includes the allocated RTP IP address and port from an access point for the VoIP call, wherein the access point is coupled to the BSC.

6. The method of claim 1, wherein the messages are transferred over an IP network.

7. The method of claim 5 comprising establishing a voice call between a first mobile station coupled to the access point and between a second mobile station coupled to a base station subsystem (BSS).

8. The method of claim 7, wherein the (BSS) is coupled to the MSC.
The method of claim 7, wherein a voice path is established between the mobile station and the access point via at least one of following protocol:

802.11;
802.16; and
Bluetooth.

10. The method of claim 7, wherein a voice path is established between the access point and the media gateway via RTP.

11. A method for providing a voice call for wireless cellular subscribers operating in a wireless local area network, comprising:

receiving a first message by a first module;
allocating a Real-Time Transport Protocol (RTP) Internet Protocol (IP) address and port;
receiving a second message by a second module from the first module, wherein the second message includes the allocated RTP IP address and port; and
receiving a third message by a third module from the second module, wherein the third message includes a parameter comprising the allocated RTP IP address and port from the first module for a current Voice over IP (VoIP) call.

12. The method of claim 11 comprising receiving a fourth message by the second module from the third module, wherein the fourth message includes a parameter comprising the allocated RTP IP address and port from a fifth module for the VoIP call, wherein the fifth module is coupled to the third module.

13. A computer readable medium comprising instructions for:
receiving a handoff request message from a mobile switching center (MSC) to a base station controller (BSC), wherein the handoff request message includes:
Real-Time Transport Protocol (RTP) Internet Protocol (IP) information from an MSC side for of a Voice over IP (VoIP) call; and
an International Mobile Subscriber Identity (IMSI); and
receiving a handoff request acknowledge message from the BSC to the MSC, wherein the handoff request acknowledge message includes RTP IP information from a BSC side for the VoIP call.
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System for providing a voice call for wireless cellular phones in a wireless local area network, comprising:

- a module; and
- a base station controller (BSC) coupled to the module;

wherein a parameter that facilitates a Voice over Internet Protocol (VoIP) call is transferred between the module and BSC;

wherein the parameter includes a Real-Time Transport Protocol (RTP) Internet Protocol (IP) address and port.

15. The system of claim 14, wherein the parameter is transferred in at least one of a following message:

- an assignment request message;
- an assignment complete message;
- a handover request message; and
- a handover request acknowledge message.

16. The system of claim 14, wherein the parameter is preferably an optional parameter.

17. The system of claim 14, wherein the BSC is coupled to the module via an A-interface.

18. The system of claim 14, wherein the BSC is coupled to the module via an IP network.

19. The system of claim 14, wherein a mobile station is adapted to connect to the module and the BSC via an IP network.

20. The system of claim 19, wherein the mobile station utilizes Direct Transfer Application Part over IP to connect to the BSC via the IP network.

21. The system of claim 20, wherein the mobile station utilizes Voice over IP to connect to the media gateway via the IP network.

22. The system of claim 21, wherein the parameter facilitates a Voice over IP call and is passed call between the BSC and the module over the A-interface.
parameter adapted to facilitate a Voice over IP cellular
subscribers operating in a wireless local area network, comprising:
an element identifier;
a length;
a most significant byte of a Real-Time Transport Protocol (RTP) Internet Protocol
(IP) address;
a second byte of the RTP IP address;
a third byte of the RTP IP address; and
a least significant byte of the RTP IP address.

24. The parameter of claim 23, comprising a first part of the RTP port address.

25. The parameter of claim 23, comprising a second part of the RTP port address.

26. The parameter of claim 23, comprising a first audio codec.

27. The parameter of claim 23, comprising a second audio codec.

28. A computer readable medium comprising instructions for:

sending a handoff request message from a mobile switching center (MSC),

wherein the handoff request message includes:

  Real-Time Transport Protocol (RTP) Internet Protocol (IP) information
from a first side for a Voice over IP (VoIP) call; and

  an International Mobile Subscriber Identity (IMSI).

29. The computer readable medium of claim 28 comprising instructions for: receiving
a handoff request acknowledge message at the MSC, wherein the handoff request acknowledge
message includes RTP IP information from a second side for the VoIP call.

30. The computer readable medium of claim 28, wherein the first side is associated
with the MSC.

31. The computer readable medium of claim 29, wherein the second side is associated
with at least one of: a base station controller and a base station subsystem.
A method for providing a voice call for wireless subscribers operating in a wireless local area network, comprising:

sending a message including:

Real-Time Transport Protocol (RTP) Internet Protocol (IP) information for a part of a Voice over IP (VoIP) call; and

an International Mobile Subscriber Identity (IMSI).

33. The method of claim 32 comprising receiving a message including RTP IP information for another part of the VoIP call.
FIG. 7

MS 24
AP 52
BSC 18
MSC/HLR 14
MGW 16
HLR 54
PSTN 400

DTAP OVER IP

VOICE OVER BLUE TOOTH/802.x

IP NETWORK

RTP PACKAGE EXCHANGE

CM SERVICE REQ
CONNECTION CONFIRM
AUTHENTICATE
AUTHENTICATE RSP
START SECURITY
SECURITY COMPLETE
SETUP
CALL PROCEEDING
ASSIGNMENT REQUEST
ASSIGNMENT COMPLETE

ADD
ACK

ALERTING
CONNECT
MOD
CONNECT ACK

IAM
ACM
ANM