separate tables for respective local calling area identifiers, for example, in order to enable a more efficient search of the access code association table for an access code that satisfies the aforementioned criteria. Therefore, the access code association table (**170**) and the process **270** illustrated in Figure **12** are examples only, and one of ordinary skill in the art will readily appreciate numerous alternative data structures and algorithms.

Gateway

Referring to Figure **13**, in this embodiment, the gateway (**18**) includes a processor circuit shown generally at **300**, which includes a microprocessor **302**. The processor circuit **300** also includes a program memory **304**, a memory **305**, and an I/O port **306**, all of which are in communication with the microprocessor **302**. The processor circuit **300** may include multiple processors etc., and the aforementioned components of the processor circuit **300** may alternatively be combined.

The I/O port **306** includes a channel interface **308**, which, in the illustrated embodiment, is in communication with the channels **20**, **22**, and **24** that were also illustrated in Figure **1**. Where, as in the illustrated embodiment, the channels **20**, **22**, and **24** are PSTN telephone lines in the PSTN network **29**, the channel interface **308** may, for example, be a T1 port for communication with one or more T1 lines (illustrated at **27** in Figure **1**) of a PSTN service provider, in a manner well-known in the art. The I/O port in the illustrated embodiment also includes an internet interface **309** for interfacing with the IP network **26** illustrated in Figure **1**. The program memory **304** stores blocks of codes for directing the microprocessor **302** to carry out the functions of the gateway (**18**). It has been found that the AS**5350** Universal Gateway available from Cisco Systems, Inc. of San Jose, California may, for example, be suitable as the gateway (**18**).

Referring back to Figure **1**, and also still to Figure **13**, when a call is received on one of the channels **20**, **22**, or **24**, the microprocessor **302** causes the I/O

port **306** to use the internet interface **309** to send a Session Initiation Protocol (SIP) Invite message to a pre-determined node with which the gateway **18** is associated, which in the illustrated embodiment is the first node **11**. Generally, the gateway **18** will be associated with a node that is geographically closest to the gateway, in order to minimize transmission times over the IP network **26**. In response to the SIP Invite message, the call controller **13** sends an RC request message to the routing controller **30** which makes various enquiries of the database **23** to produce a routing message that is sent back to the call controller **13**. The call controller **13** then communicates with the media relay **28** to cause a communications link including an audio path and a videophone (if a videopath call) to be established through the media relay to the same node, a different node, or to a communications supplier gateway as shown generally at **34** to carry audio, and where applicable, video traffic to the call recipient or callee.

Referring to Figure **14**, an exemplary SIP Invite message is shown generally at **310** and includes a caller identifier field **312**, a callee identifier field **314**, a digest parameter field **315**, a call identifier field **316**, an IP address field **317**, and a gateway UDP port field **318**. Examples of values for the fields in the SIP Invite message **310** are shown for illustration purposes only in Figure **14**. The caller identifier in the caller identifier field **312** is preferably in the form of the telephone number of the caller followed by the "@" symbol, which in turn is followed by the IP address of the gateway (**18**) in the IP network (**26**). The caller identifier may be determined by retrieving calling line identification (CLID) information from the signal provided by the PSTN network (**29**) to the gateway (**18**) for example Where the caller identification information is not available to the gateway (**18**), the caller identifier in the caller identifier field **312** preferably includes a generic number (such as **11111**, for example) indicating that the caller identification information was not available, followed by the "@" symbol and then by the IP address of the gateway (**18**).

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The callee identifier in the callee identifier field **314** is the access code identifying the channel (**20**, **22**, or **24** in the example of Figure **1**) on which the call was placed, and which was received from the access server (**14**). In the illustrated example, the access code is the PSTN telephone number **1-604-345-1212** corresponding to the channel **20** illustrated in Figure **1**, and to the access code stored in the access code field **173** of the record **172** in the exemplary access code association table **170** illustrated in Figure **10**.

The digest parameter in the digest parameter field **315** is generated by the gateway (**18**) and may uniquely identify the SIP session that is initiated with the SIP Invite message **310**.

The call identifier in the call identifier field **316** is, in the illustrated embodiment, a four-digit hexadecimal number generated by the gateway (**18**) to identify the call, followed by the "@" symbol, which in turn is followed by the IP address of the gateway.

The IP address in the IP address field **317** is the IP address of the gateway (**18**) in the IP network (**26**), and the gateway UDP port number in the gateway UDP port field **318** includes a UDP port identifier identifying a UDP port at which the audio/video path will be terminated at the gateway (**18**).

It should be noted that throughout the description of the embodiments of this invention, the IP/UDP addresses of all elements such as the gateway (**18**) will be assumed to be valid IP/UDP addresses directly accessible via the Internet or a private IP network, for example, depending on the specific implementation of the system. As such, it will be assumed, for example, that the gateway (**18**) will have an IP/UDP address directly accessible by the call controllers and the media relays on their respective nodes, and those addresses will not be obscured by Network Address Translation (NAT) or similar mechanisms. In other words, the IP/UDP information contained in SIP messages (for example the SIP Invite message or the RC Request message

which will be described below) will match the IP/UDP addresses of the IP packets carrying these SIP messages.

It will be appreciated that in many situations, the IP addresses assigned to various elements of the system may be in a private IP address space, and thus not directly accessible from other elements. Furthermore, it will also be appreciated that NAT is commonly used to share a "public" IP address between multiple devices, for example between home PCs and IP telephones sharing a single Internet connection. For example, the gateway (**18**) may be assigned an IP address such as **192.168.0.5**. This address is located in so called "non-routable" (IP) address space and cannot be accessed directly from the Internet. In order for this device to communicate with other computers located on the Internet, the IP address has to be converted into a "public" IP address, for example **24.14.102.5** assigned by the Internet Service Provider, by a device performing NAT, typically a router. In addition to translating the IP address, NAT typically also translates UDP port numbers, for example an audio path originating at the gateway (**18**) and using a UDP port **12378** at its private IP address, may have be translated to a UDP port **23465** associated with the public IP address of the NAT device. In other words, when a packet originating from the gateway (**18**) arrives at an Internet-based node, the source IP/UDP address contained in the IP packet header will be **24.14.102.5:23465**, whereas the source IP/UDP address information contained in the SIP message inside this IP packet will be **192.168.0.5:12378**. The mismatch in the IP/UDP addresses may cause a problem for SIP-based VoIP systems because, for example, a node will attempt to send messages to a private address but the messages will never get there.

Call Controller

Referring to Figure **15**, the call controller (**13**) includes a processor circuit shown generally at **320**. The processor circuit **320** includes a microprocessor **322**, program memory **324**, and an I/O port **326**. The program memory **324** and the I/O port **326** are in communication with the microprocessor **322**. The

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processor circuit **320** may include a plurality of microprocessors, a plurality of program memories, and a plurality of I/O ports to be able to handle a large volume of calls. However, for simplicity, the processor circuit **320** will be described as having only one microprocessor **322**, program memory **324**, and I/O port **326**, it being understood that there may be more.

Generally, the I/O port **326** includes an input **328** for receiving messages such as the SIP Invite message from the gateway (**18**) or from a VoIP telephone (**36** in Figure **1**, for example). The I/O port **326** also has an RC request message output **330** for transmitting an RC request message to the routing controller **30** of Figure **1**, an RC message input **332** for receiving routing messages from the routing controller **30**, a gateway output **334** for transmitting messages to the gateway **18** and/or **34** shown in Figure **1** to advise the gateway **18** and/or **34** to establish an audio path, for example, and a gateway input **336** for receiving messages from the gateway **18** and/or **34**. The I/O port **326** further includes a SIP output **338** for transmitting messages to the gateway (**18** and/or **34**) or VoIP telephone (**36**, for example) to advise the gateway **18** and/or **34** or IP telephone of the IP addresses of the gateways which will establish the audio/video path. The I/O port **326** further includes a voicemail server input and output **340** and **342** respectively for communicating with the voicemail server **19** shown in Figure **1**.

While certain inputs and outputs have been shown as separate, it will be appreciated that some may be a single IP address and IP port. For example, the messages sent to the routing controller (**30**) and received from the routing controller (**30**) may be transmitted and received on the same single IP port.

The program memory **324** includes blocks of code for directing the microprocessor **322** to carry out various functions of the call controller (**13**). For example, these blocks of code include a first block **344** for causing the processor circuit **320** to execute a SIP Invite to RC Request process to produce an RC Request Message in response to a received SIP Invite

message. In addition, there is a Routing Message to Gateway message block **346** which causes the processor circuit **320** of the call controller to produce a gateway query message in response to a received routing message from the routing controller (**30**).

Referring to Figures **15** and **16**, the SIP Invite to RC Request process is shown in more detail at **344**. On receipt of a SIP Invite message of the type shown in Figure **14**, block **350** directs the processor circuit **320** to produce an RC Request Message. Block **352** then directs the processor circuit **320** to cause the RC Request Message to be sent to the routing controller **30** illustrated in Figure **1**.

Referring to Figure **17**, an exemplary RC request message is shown generally at **360** and includes a caller identifier field **362**, a callee identifier field **364**, a digest parameters field **366**, and a call identifier field **368**. These fields may be populated with the contents of the caller identifier field **312**, callee identifier field **314**, digest parameter field **315**, and call identifier field **316** respectively of the SIP Invite message **310** illustrated in Figure **14**. In other embodiments, the RC request message may further include a type field (not shown) containing a type code to indicate whether the call is from a third party or from a system subscriber. Other variations of an RC request message are explained in PCT Publication No. WO **2008/052340**. A type field (not shown) in the RC request message **360** may be advantageous in embodiments where SIP Invite messages may also be received from an IP telephone that is using VoIP software to make a voice call. However, in the embodiments that are illustrated herein, SIP Invite messages originate from the gateway (**18**), and therefore a type designation is not necessary and may be omitted from the RC request message **360**. In embodiments where a SIP Invite message may be received from an IP telephone, the SIP invite to RC request process shown in Figure **16** may require additional steps, as illustrated in Figure **5** of PCT Publication No. WO **2008/052340**.

RC Request Message Handler

As illustrated in Figure **8**, the program memory **234** includes an RC request message handler **380** which directs the routing controller (**30**) to produce a routing message in response to a received RC request message (**360**). Referring to Figure **18**A, the RC request message handler **380** begins with a first block **382** that directs the RC processor circuit (**230**) to separately store the contents of the callee identifier field **364** and caller identifier field **362** of the RC request message (**360**) in the callee identifier store **246** and the caller identifier store **248** respectively of Figure **8**.

Block **384** then directs the RC processor circuit (**230**) to use the contents of the caller username store **249** to locate and retrieve from the database (**23**) a dialing profile **200** associated with the caller, as described above and illustrated in Figure **9**, for example. The retrieved dialing profile may then be stored in the temporary memory **240**, for example.

The RC request message handler **380** continues at block **386**, which directs the processor circuit (**230**) of the routing controller to determine whether the contents of the current number of concurrent calls field **222** of the dialing profile **200** shown in Figure **9** are less than the contents of the maximum number of concurrent calls field **220** of the dialing profile for the caller and, if so, block **388** directs the processor circuit to increment the contents of the current number of concurrent calls field **222** and the processor circuit (**230**) is directed to point A in Figure **18**B. If the contents of the current number of concurrent calls field **222** are equal to or greater than the contents of the maximum number of concurrent calls field **220**, then block **390** directs the processor circuit (**230**) to send an error message back to the call controller (**13**) to cause the call controller to notify the caller that the maximum number of concurrent calls has been reached and no further calls can exist concurrently, including the presently requested call.

Assuming that block **386** allows the call to proceed, the RC processor circuit (**230**) is directed to perform certain checks on the callee identifier in the callee identifier field **246** in Figure **8**. These checks are shown in greater detail in Figure **18**B.

Referring to Figure **18**B, the RC processor circuit (**230**) is directed to a first block **392** that causes it to determine whether a digit pattern of the callee identifier includes a pattern that matches the contents of the international dialing digits (IDD) field **208** in the dialing profile **200** (shown in Figure **9**) associated with the caller. If so, then block **394** directs the RC processor circuit (**230**) to set a call type code identifier variable maintained by the processor to indicate that the call is an international call, and block **396** directs the processor to produce a reformatted callee identifier by reformatting the callee identifier into a predefined digit format. In this embodiment, this is done by removing the pattern of digits matching the IDD field contents (**208**) of the caller dialing profile **200** to effectively shorten the callee identifier. Then, block **398** directs the RC processor circuit (**230**) to determine whether or not the callee identifier has a length which meets criteria establishing it as a number compliant with the E.**164** Standard set by the ITU. If the length does not meet these criteria, then block **400** directs RC processor circuit (**230**) to send back to the call controller (**13**) a message indicating the length is not correct. The process **380** is then ended. At the call controller **13**, routines (not shown) stored in the program memory **324** may direct the processor circuit (**320** of Figure **15**) to respond to the incorrect length message by transmitting a message back to the mobile telephone (**12** shown in Figure **1**) to indicate that an invalid number has been dialled.

If the length of the amended callee identifier meets the criteria set forth at block **398**, then block **402** directs the RC processor circuit (**230**) to make a database request to the database (**23**) to determine whether or not the amended callee identifier is found in the DID field (**373**) of a record such as shown in Figure **11** in the DID bank table. If at block **402** the RC processor

circuit (**230**) receives a response from the database (**23**) indicating that the reformatted callee identifier produced at block **396** is found in the DID field (**373**) of a record in the DID bank table, then the callee is a subscriber to the system and the call is classified as a private network call by directing the processor to block **404**, which directs the RC processor circuit (**230**) to copy the contents of the corresponding username field (**371** in Figure **11**) from the callee DID bank table record (**370** in Figure **11**) into the callee identifier store (**246** in Figure **8**). Thus, the RC processor circuit (**230**) locates a subscriber username associated with the reformatted callee identifier. The processor (**232**) is then directed to point B in Figure **18**A.

Calls to Subscribers in Different Nodes

Referring back to Figure **1**, as noted above, the gateway **18** is preferably associated with a pre-determined node, which in the illustrated embodiment is the first node **11**. Referring back to Figure **18**A, block **406** directs the processor (**232** of Figure **8**) to execute a process to determine whether or not the node associated with the reformatted callee identifier in the callee identifier store (**246** in Figure **8**, which, at block **404**, was set to be a username of the callee) is the same node that is associated with the gateway **18** illustrated in Figure **1**.

To do this, the processor (**232**) may, for example, identify a node associated with the gateway (**18**) by using an IP address associated with the gateway to determine a node identifier of the gateway. An IP address associated with the gateway (**18**) may, for example, be obtained from either the caller identifier field **362** or the call identifier field **368** of the RC request message **360** illustrated in Figure **17**, as each of these fields includes a portion following an "@" symbol that indicates an IP address of the gateway. In order to determine a node identifier associated with the gateway (**18**) using the IP address associated with gateway (**18**), the processor **232** (illustrated in Figure **8**) may access a gateway node association table stored in the database **23** (illustrated in Figure **1**).

Referring to Figure **19**, an exemplary gateway node association table is shown generally at **480**. The exemplary gateway node association table **480** includes first and second records **482** and **484**, each having a respective gateway IP address field **486** and a respective node identifier field **488**. It will be appreciated that the exemplary gateway node association table **480** is an example for illustration purposes only. The values in the gateway IP address fields **486** are preferably initialized when a gateway (such as the gateway **18** illustrated in Figure **1**) is installed as part of the system (**10**), and are preferably updated as the IP addresses of the respective gateways may change from time to time. The values in the node identifier fields **488** are also preferably initialized when a gateway (such as the gateway **18** illustrated in Figure **1**) is installed as part of the system (**10**).

As indicated above, the reformatted callee identifier in the callee identifier store (**246** in Figure **8**) was set at block **404** in Figure **18**B to be a username of the callee from the username field **371** (illustrated in Figure **11**), and in this embodiment, a prefix of the username of the callee preferably indicates a node associated with the callee. In the illustrated embodiment, the left-most digit in the username of the callee is a continent code, which is a sufficient prefix to identify a node associated with the callee. However, it will be appreciated that in other embodiments, other prefixes or other information may identify the associated node. Preferably, the values in the node identifier fields **488** correspond to the prefixes of the usernames in the username fields **371** (illustrated in Figure **11**), so that the node associated with the callee is the same node that is associated with the gateway **18** illustrated in Figure **1** if the prefix of the username of the callee matches the node identifier associated with the gateway (**18**). Therefore, in the illustrated embodiment, if the reformatted callee identifier in the callee identifier store (**246** in Figure **8**) is **2001 1050 8667**, for example, then in the example of Figure **19**, the node associated with the callee is the same node as the node identified by the continent code "**2**" that is associated with the gateway associated with the IP

address **20.14.102.5** in the record **482**, but is not the same node as the node identified by the continent code "**5**" that is associated with the gateway associated with the IP address **104.12.131.12** in the record **484**.

Referring back to Figure **18**A, if at block **406** the prefix of the username of the callee does not match the node identifier associated with the gateway (**18**), then the call is a "cross-domain" call, and block **408** in Figure **18**A directs the processor (**232** in Figure **8**) to set a call type flag in the temporary memory (**240** in Figure **8**) to indicate the call is a cross-domain call. Then, block **410** of Figure **18**A directs the processor (**232** of Figure **8**) to produce a routing message identifying an address on the private network with which the callee identified by the contents of the callee ID buffer is associated and to set a time to live for the call at a maximum value of **99999**, for example. Routing messages and time to live values, and also a method of determining the node in the system with which the callee is associated, are further described in PCT Publication No. WO **2008/052340**. Once a routing message is produced at block **410**, block **412** directs the processor (**232** in Figure **8**) to cause the routing message to be sent to the call controller **13** shown in Figure **1**, and the process ends.

Referring back to Figure **18**B, if at block **392**, the callee identifier stored in the callee identifier store (**246** in Figure **8**) does not begin with an international dialing digit, then block **414** directs the processor (**232**) to determine whether or not the callee identifier begins with the same national dial digit code as assigned to the caller. To do this, the processor (**232**) is directed to refer to the retrieved caller dialing profile as shown in Figure **9**. In Figure **9**, the national dialing digit code **206** is the number **1**. Thus, if the callee identifier begins with the number **1**, then the processor (**232**) is directed to block **416** in Figure **18**B.

Block **416** directs the processor (**232** of Figure **8**) to examine the callee identifier to determine whether or not the digits following the NDD digit identify

an area code that is the same as any of the area codes identified in the local area codes field **212** of the caller dialing profile **200** shown in Figure **9**. If not, block **418** of Figure **18B** directs the processor (**232**) to set the call type flag to indicate that the call is a national call. If the digits following the NDD digit identify an area code that is the same as a local area code associated with the caller as indicated by the caller dialing profile, block **420** directs the processor (**232**) to set the call type flag to indicate a local call, national style. After executing block **418** or **420**, block **422** directs the processor (**232**) to format the callee identifier into a pre-defined digit format to produce a re-formatted callee identifier by removing the national dialled digit and prepending a caller country code identified by the country code field **210** of the caller dialing profile shown in Figure **9**. The processor (**232**) is then directed to block **398** of Figure **18B** to perform other processing as already described above.

If at block **414**, the callee identifier does not begin with a national dialled digit, block **424** directs the processor (**232**) to determine whether the callee identifier begins with digits that identify the same area code as the caller. Again, the reference for this is the retrieved caller dialing profile shown in Figure **9**. The processor (**232**) determines whether or not the first few digits of the callee identifier identify an area code corresponding to the contents of any area code identifier stored in the local area code field **212** of the retrieved caller dialing profile **200** (illustrated in Figure **9**). If so, then block **426** directs the processor (**232**) to set the call type flag to indicate that the call is a local call. It should be noted that the call will not necessarily be a local call in every case where the first few digits of the callee identifier identify an area code corresponding to the contents of an area code identifier stored in the local area code field **212** (illustrated in Figure **9**), and other determinations of when a call is to be considered local may be appropriate. However, it has been found that the determination described above for block **424** is satisfactory for some purposes. Next, block **428** directs the processor (**232**) to format the callee identifier into a pre-defined digit format to produce a reformatted callee

identifier by prepending the caller country code to the callee identifier, the caller country code being determined from the country code field **210** of the retrieved caller dialing profile **200** shown in Figure **9**. The processor (**232**) is then directed to block **398** for further processing as described above.

If at block **424**, the callee identifier does not start with the same area code as the caller, block **430** directs the processor (**232** of Figure **8**) to determine whether the number of digits in the callee identifier, *i.e.* the length of the callee identifier, is within the range of digits indicated by the caller minimum local number length field **214** and the caller maximum local number length field **216** of the retrieved caller dialing profile **200** shown in Figure **9**, and whether there is more than one area code identifier stored in the local area code field **212** of the retrieved caller dialing profile. If the number of digits in the callee identifier is within the aforementioned range and there is only one area code identifier stored in the local area code field (**212**), then block **432** directs the processor (**232**) to set the call type flag to indicate a local call and block **434** directs the processor (**232**) to format the callee identifier into a pre-defined digit format to produce a reformatted callee identifier by prepending to the callee identifier the caller country code (as indicated by the country code field **210** of the retrieved caller dialing profile **200** shown in Figure **9**) followed by the caller area code as indicated by any of the local area codes stored in the local area code field **212** of the caller dialing profile **200** shown in Figure **9**. The processor (**232**) is then directed to block **398** of Figure **18B** for further processing as described above.

If at block **430**, the callee identifier has a length that does not fall within the range specified by the caller minimum local number length field (**214** in Figure **9**) and the caller maximum local number length field (**216** in Figure **9**), or if there is more than one area code identifier stored in the local area code field **212** of the retrieved caller dialing profile **200** illustrated in Figure **9**, then block **436** directs the processor (**232**) to send an error message back to the call controller (**13**), and the process ends.

In alternative embodiments, such as those illustrated in PCT Publication No. WO **2008/052340**, an additional block (**402** in Figure **8**B of PCT Publication No. WO **2008/052340**) may determine whether the callee identifier is a valid username. However, in the embodiment disclosed herein, the callee identifier is assumed to be a telephone number of the callee, and not a username.

From Figure **18**B, it will be appreciated that there are certain groups of blocks of codes that direct the processor **232** in Figure **8** to determine whether the callee identifier has certain features such as an international dialing digit, a national dialing digit, an area code and a length that meet certain criteria, and cause the processor **232** to reformat the callee identifier stored in the callee identifier store **246** in Figure **8**, as necessary into a predetermined target format including only a country code, area code, and a normal telephone number, for example, to cause the callee identifier to be compatible with the E.**164** number plan standard in this embodiment. This enables block **402** in Figure **18**B to have a consistent format of callee identifiers for use in searching through the DID bank table records **370** of the type shown in Figure **11** to determine how to route calls to subscribers on the same system. Effectively, therefore blocks **392**, **414**, **424**, and **430** establish call classification criteria for classifying the call as a public network call or a private network call. Block **402** classifies the call, depending on whether or not the formatted callee identifier has a DID bank table record, and this depends on how the call classification criteria are met.

<u>Calls to Non-Subscribers</u>

Not all calls will be to subscribers, and this will be detected by the processor **232** of Figure **8** when it executes block **402** in Figure **18**B, and does not find a DID bank table record (**370** illustrated in Figure **11**) that is associated with the callee, in the DID bank table. When this occurs, the call is classified as a public network call, by directing the processor (**232**) to point C in Figure **18**C.

Referring to Figure **18C**, block **438** directs the processor (**232**) to determine whether the formatted callee identifier in the callee identifier store **246** in Figure **8** corresponds to an access code in the access code field **173** of a record in the access code association table **170** illustrated in Figure **10** that is associated with a callee identifier. Because the callee identifier in the callee identifier store **246** in Figure **8** has been formatted as described above with reference to Figure **18B**, block **438** may involve determining whether an access code in the access code field **173** of a record of the access code association table **170** (illustrated in Figure **10**) matches the formatted callee identifier in the callee identifier store **246** in Figure **8**, and also whether a callee identifier (as opposed to the "null" value assigned on initialization) is stored in the callee identifier field **177** in association with the access code. As noted above, for simplicity, this description is directed to embodiments wherein an access code association table **170** associates access codes with respective callee identifiers, caller identifiers, timeout values, and timestamp values, although it will be appreciated that the processes described herein for records in the access code association table **170** may additionally or alternatively be applied to DID bank table records **370** in an analogous manner.

If at block **438** the formatted callee identifier in the callee identifier store **246** in Figure **8** is the same as an access code in the access code field (**173**) of a record of the access code association table **170** illustrated in Figure **10** that is associated with a callee identifier, then block **440** directs the processor (**232**) to determine whether the caller identifier in the caller identifier store **248** (illustrated in Figure **8**) is the same as the caller identifier in the caller identifier field (**179**) of the record of the access code association table (**170**), and thus whether the caller identifier in the caller identifier field (**179**) of the record of the access code association table (**170**) identifies the mobile telephone identified by the caller identifier in the caller identifier store **248**. If not, then block **442** directs the processor (**232**) to send an error message to the call controller (**13**), and the process ends.

But if at block **440** the caller identifier in the caller identifier store **248** (illustrated in Figure **8**) corresponds to the caller identifier in the caller identifier field **(179)** of the record of the access code association table **(170)**, then the routing controller **(30)** will produce a routing message that will cause the call controller to establish communication through the IP network **(26)** to the callee in response to a call received at a channel **(20, 22, or 24)**. Preferably, block **444** includes codes that direct the processor **(232)** to determine whether the association of the access code with the callee identifier has expired, and thus whether the usability of the access code to initiate a call to the callee has expired, in the manner described above for block **278** in Figure **12**. If at block **444** the association of the access code with the callee identifier has expired, then block **442** directs the processor **(232)** to send an error message to the call controller **(13)**, and the process ends. Thus the routing controller produces a routing message that causes the call controller to establish the call only when the association of the access code with the callee identifier has not expired.

It will be appreciated that in alternative embodiments, one or more of the caller identifier, timeout, and timestamp fields **179**, **181**, and **182** may be omitted from the access code association table **170** illustrated in Figure **10**, and in these embodiments, one or more of the blocks **440**, **442**, and **444** may also be omitted.

If at block **444** the association of the access code with the callee identifier has not expired, or if one or both of blocks **440** and **444** is omitted, then block **446** directs the processor **(232)** to store the callee identifier from the callee identifier field **177** of the record of the access code association table **(170)** in the callee identifier store **246** illustrated in Figure **8**. The processor **(232)** is then directed to point A in Figure **18B** to repeat the steps illustrated in Figure **18B** using the callee identifier retrieved from the callee identifier field **(177)** in the record of the access code association table **(170)**.

However, if at block **438** the formatted callee identifier in the callee identifier store **246** in Figure **8** does not correspond to an access code in a record of the access code association table **170** illustrated in Figure **10** that is associated with a callee identifier, then block **448** of Figure **18**B causes the processor (**232**) to set the contents of the callee identifier store **246** of Figure **8** to be the newly formatted callee identifier, i.e., a number compatible with the E.**164** standard. Then, block **450** of Figure **18**B directs the processor (**232**) to generate a routing message identifying a gateway to the public network usable by the call controller (**13**) to establish a "public system" call. In one embodiment, block **450** includes codes that, for example, direct the processor (**232**) to search a database of route or master list records and to search a database of supplier records to identify at least one supplier operable to supply a communications link for the call, and to load a routing message buffer with supplier information, time to live values, and timeout values. An example of an implementation of these steps is described with reference to blocks **410**, **412**, **560**, **562**, **563**, **564**, **566**, and **571** in Figures **8**B and **8**D in PCT Publication No. WO **2008/052340**. Next, block **452** directs the processor **232** of Figure **10** to send the routing message to the call controller **13** in Figure **1**, and the process ends.

Calls to Subscribers Within the Same Node

Referring back to Figure **18**A, if at block **406**, the prefix of the username of the callee matches the node identifier associated with the gateway (**18**), then the call is on one domain, and block **454** directs the processor (**232**) to use the callee identifier in the callee identifier store **246** illustrated in Figure **8** (which, at block **404**, was set to be a username of the callee) to locate and retrieve a dialing profile for the callee. The dialing profile may be of the type shown in Figure **9**, for example. Block **456** of Figure **18**A then directs the processor **232** of Figure **8** to get call block, call forward, and voicemail records from the database **23** of Figure **1**, based on the username identified in the callee dialing profile retrieved by the processor at block **454**. Exemplary call block,

call forward, and voicemail records are described in PCT Publication No. WO **2008/052340**.

Then block **458** directs the processor **232** of Figure **8** to determine whether or not the caller identifier received in the RC request message matches a block pattern stored in the call block record associated with the callee and retrieved at block **454**. If the caller identifier matches a block pattern, then block **460** directs the processor to send a drop call or non-completion message to the call controller (**13**) and the process is ended. If the caller identifier does not match a block pattern associated with the callee, then block **462** directs the processor (**232**) to determine whether or not call forwarding is required, as described in PCT Publication No. WO **2008/052340**.

If at block **462**, the call forwarding record for the callee indicates that no call forwarding is required, then the processor (**232**) is directed to block **464**, which directs the processor (**232**) to generate a routing message identifying an address on the private network, associated with the callee for a "private system" call. In one embodiment, block **464** includes codes that, for example, direct the processor (**232**) to store, in a routing message buffer, a username and domain of the callee, time to live values, and an IP address of the current node, to determine whether or not the user identified by the callee identifier has paid for voicemail service and if so, to store voicemail information in the routing message buffer. An example of an implementation of these steps is described with reference to blocks **609**, **620**, **640**, **642**, and **644** in Figures **8A** and **8C** in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference. Next, block **466** directs the processor **232** of Figure **8** to cause the routing message to be sent to the call controller **13** in Figure **1**, and the process ends.

But if at block **462**, the call forwarding record for the callee indicates that call forwarding is required, then block **468** directs the processor (**232**) to search the dialing profile table to find a dialing profile record as shown in Figure **9**, for

the user identified by the destination number field of the call forward record, as illustrated in PCT Publication No. WO **2008/052340**. The processor (**232**) is further directed to store the username and domain for that user and a time to live value in a routing message buffer, an example of which is described in PCT Publication No. WO **2008/052340**. This process is repeated for each call forwarding record associated with the callee identified by the callee identifier store **246** in Figure **8** to add to the routing message buffer all call forwarding usernames and domains associated with the callee.

Referring to Figures **1**, **18A**, and **18C**, the routing message sent at one of blocks **412**, **452**, and **466** is received at the call controller **13** and the call controller interprets the receipt of the routing message as a request to establish a call. Referring to Figure **15**, the program memory **324** of the call controller **13** includes a routing to gateway routine depicted generally at **346**.

Where a routing message received at the call controller **13** is of the type produced at block **464** shown in Figure **18A**, indicating that the callee is a system subscriber on the same node as the gateway (**18**) (such as a user of the VoIP telephone **36** illustrated in Figure **1**), the routing to gateway routine **346** may direct the microprocessor **322** to cause a message to be sent back through the IP network **26** shown in Figure **1** to the VoIP telephone (**36**), using the IP address of the VoIP telephone (**36**) that is available from the callee username.

Alternatively, if the routing message received at the call controller **13** is of the type produced at block **410** shown in Figure **18A**, identifying a domain associated with another node in the system, the call controller **13** may send a SIP invite message along the high speed/high data throughput link **17** in communication with the other node. The other node may function as explained above and in PCT Publication No. WO **2008/052340**, in response to receipt of a SIP invite message.

If the routing message received at the call controller **13** is of the type produced at block **450** shown in Figure **18**C, indicating that the callee is not a subscriber to the system (such as a user of the PSTN telephone **32** that is in communication with the IP network **26** through the gateway **34** as illustrated in Figure **1**), the call controller sends one or more SIP invite messages to the suppliers identified in the routing message to identify the IP address of a supplier that is able to carry the call, such as the IP address of the gateway **34** illustrated in the example of Figure **1**. A process for identifying the IP address of a supplier that is able to carry the call is given in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference. In some cases, the gateway of the supplier that is able to carry the call will be the gateway **18** illustrated in Figure **1**, that is, the same gateway through which the caller telephone (**12**) initiated the call. For simplicity, the following description assumes that the gateways **18** and **34** are distinct gateways. It will be understood that in some cases, they may be the same gateway, but in these cases, the following steps may still be applied.

Referring to Figure **1**, the IP address of the gateway **34** is sent in a message from the call controller **13** to the media relay **28**, which responds with a message indicating an IP address to which the gateway **18** should send its audio/video traffic, and an IP address to which the gateway **34** should send its audio/video for the call. The call controller conveys the IP address at which the media relay **28** expects to receive audio/video from the gateways **18** and **34**, to the gateways **18** and **34** in one or more messages. The gateway **18** replies to the call controller **13** with an IP address at which it would like to receive audio/video, and the call controller conveys that IP address to the media relay **28**. The call may then be conducted between the caller and callee through the media relay **28** and the gateways **18** and **34**.

If the call controller **13** receives a routing message of the type produced at block **464** shown in Figure **18**A, indicating that the callee is a system subscriber on the same node as the gateway (**18**) (such as a user of the VoIP

telephone **36** illustrated in Figure **1**), and which has at least one call forwarding number and/or a voicemail number, the call controller attempts to establish a call to the callee VoIP telephone **36** by seeking from the callee telephone a message indicating an IP address to which the media relay **28** should send audio/video. If no such message is received from the callee telephone, no call is established. If no call is established within a pre-determined time, the call controller **13** attempts to establish a call with the next user identified in the call routing message in the same manner. This process is repeated until all call forwarding possibilities have been exhausted, in which case the call controller communicates with the voicemail server **19** identified in the routing message to obtain an IP address to which the media relay **28** should send audio/video and the remainder of the process mentioned above for establishing IP addresses at the media relay and the caller telephone is carried out to establish audio/video paths to allowing the caller to leave a voicemail message with the voicemail server.

When an audio/video path through the media relay **28** is established, a call timer maintained by the call controller **13** preferably logs the start date and time of the call and logs the call ID and an identification of the route (i.e., audio/video path IP address) for later use in billing.

Terminating the Call

Referring back to Figure **1**, in the event that the caller terminates a call, the gateway **18** sends a SIP bye message to the call controller **13**. Similarly, in the event that the callee terminates the call, the gateway **34** or the VoIP telephone **36** of the callee sends a SIP bye message to the call controller **13**. Exemplary SIP bye messages are described in PCT Publication No. WO **2008/052340**. The SIP bye message is received at the call controller **13**, and the call controller executes a process that involves decrementing the contents of the current number of concurrent calls field **222** dialing profile **200** of the caller as illustrated in Figure **9**, generating an RC call stop message (not shown), sending the RC call stop message to the routing controller **30**, and

sending a "bye" message to the party that did not terminate the call. An exemplary RC call stop message, and an example of how these steps may be implemented, are described in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference.

When the routing controller **30** receives the RC call stop message from the call controller **13**, the routing controller executes an RC call stop message process that involves making various updates to subscriber, reseller, and supplier account records (not shown) following the call. Examples of subscriber, reseller, and supplier account records, and of updates to subscriber, reseller, and supplier account records, are described in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference.

While specific embodiments of the invention have been described and illustrated, such embodiments should be considered illustrative of the invention only and not as limiting the invention.

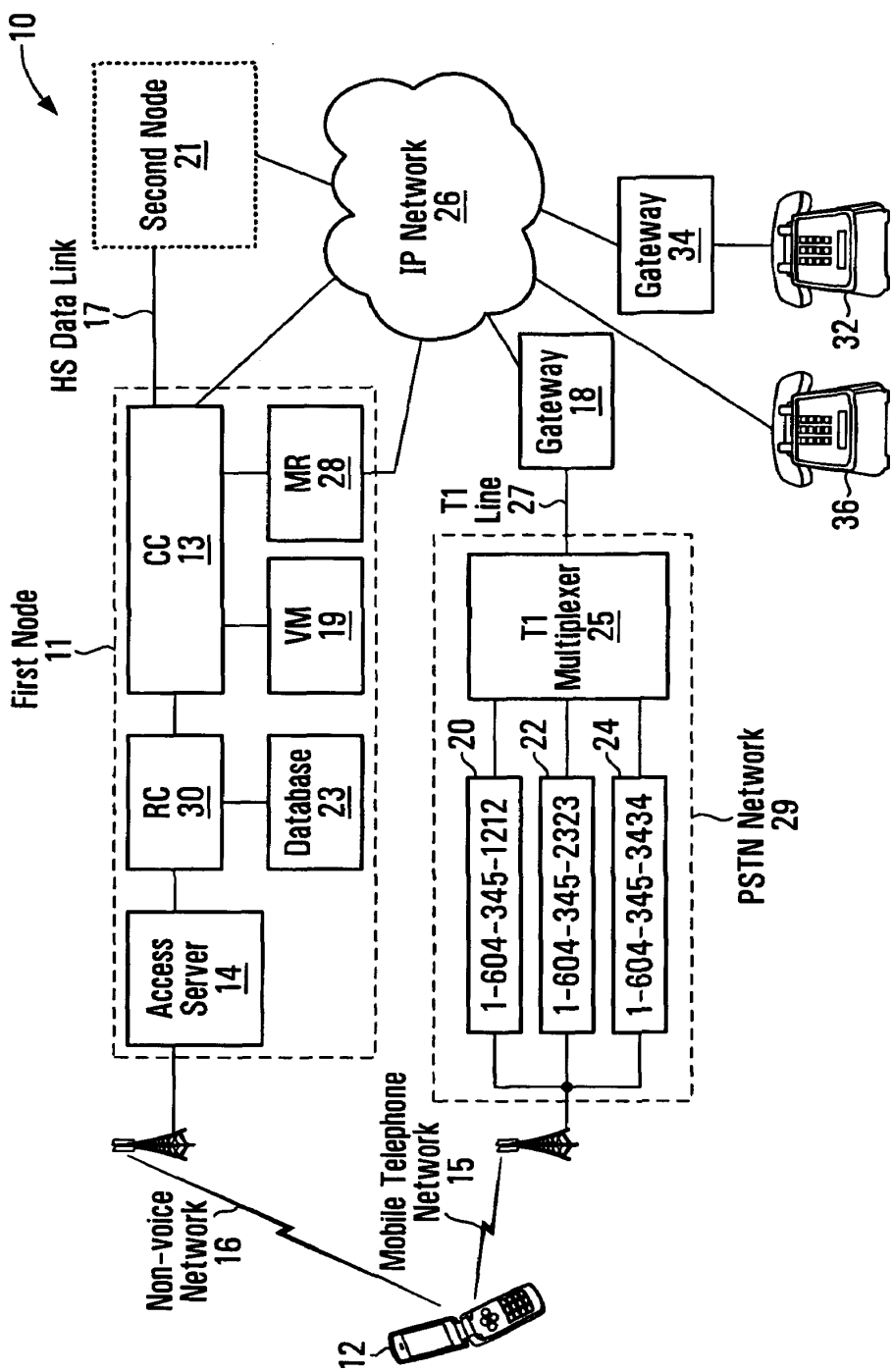

10
Second Node 21
IP Network 26
Gateway 34
32
HS Data Link 17
Gateway 18
36
T1 Line 27
First Node 11
CC 13
MR 28
VM 19
RC 30
Database 23
Access Server 14
T1 Multiplexer 25
20
22
24
1-604-345-1212
1-604-345-2323
1-604-345-3434
PSTN Network 29
Non-voice Network 16
Mobile Telephone Network 15
12

FIG. 1

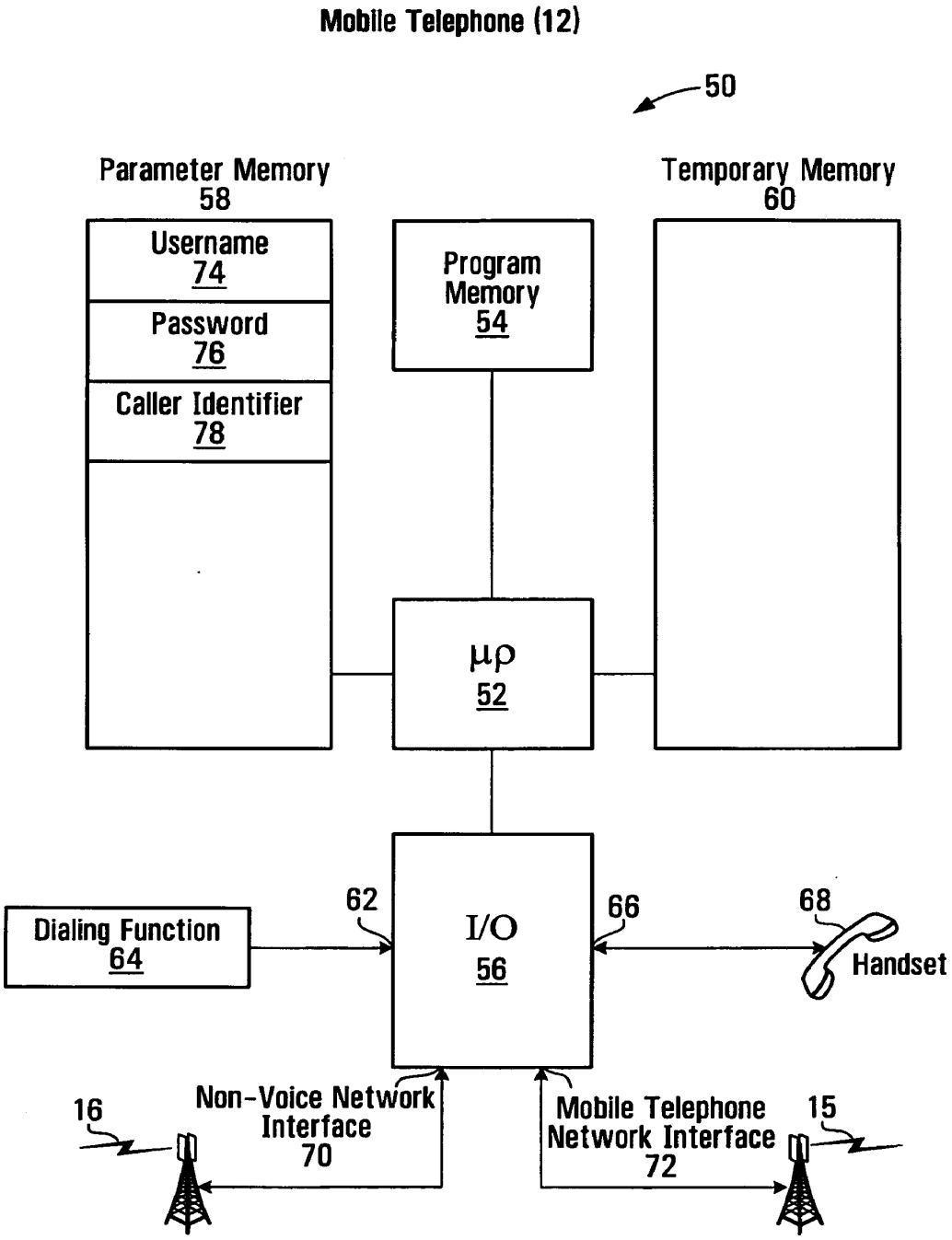

Mobile Telephone (12)
50
Parameter Memory
58
Username
74
Password
76
Caller Identifier
78
Program
Memory
54
Temporary Memory
60
μρ
52
I/O
56
62
Dialing Function
64
66
68
Handset
16
Non-Voice Network
Interface
70
Mobile Telephone
Network Interface
72
15

FIG. 2

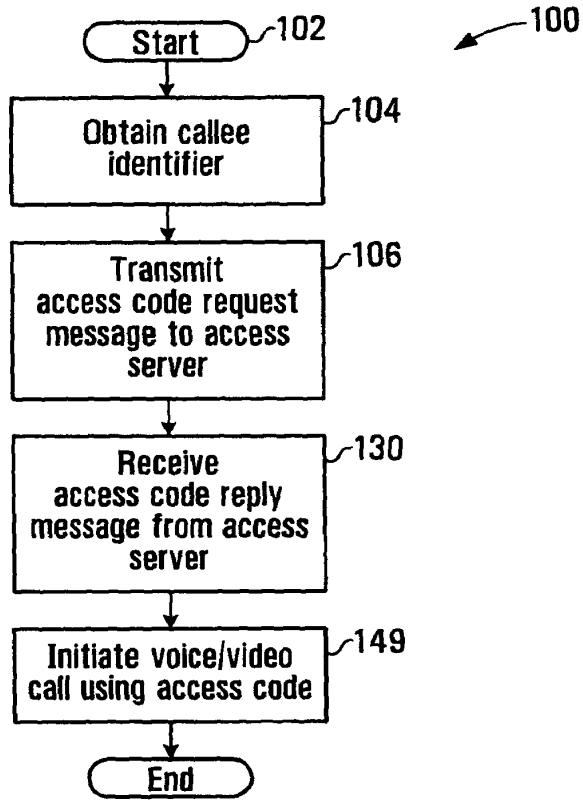


FIG. 3

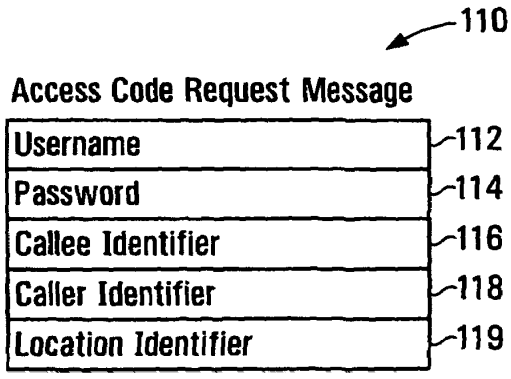


FIG. 4

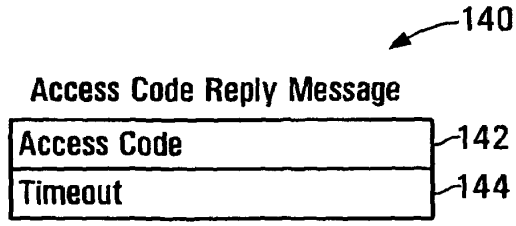


FIG. 5

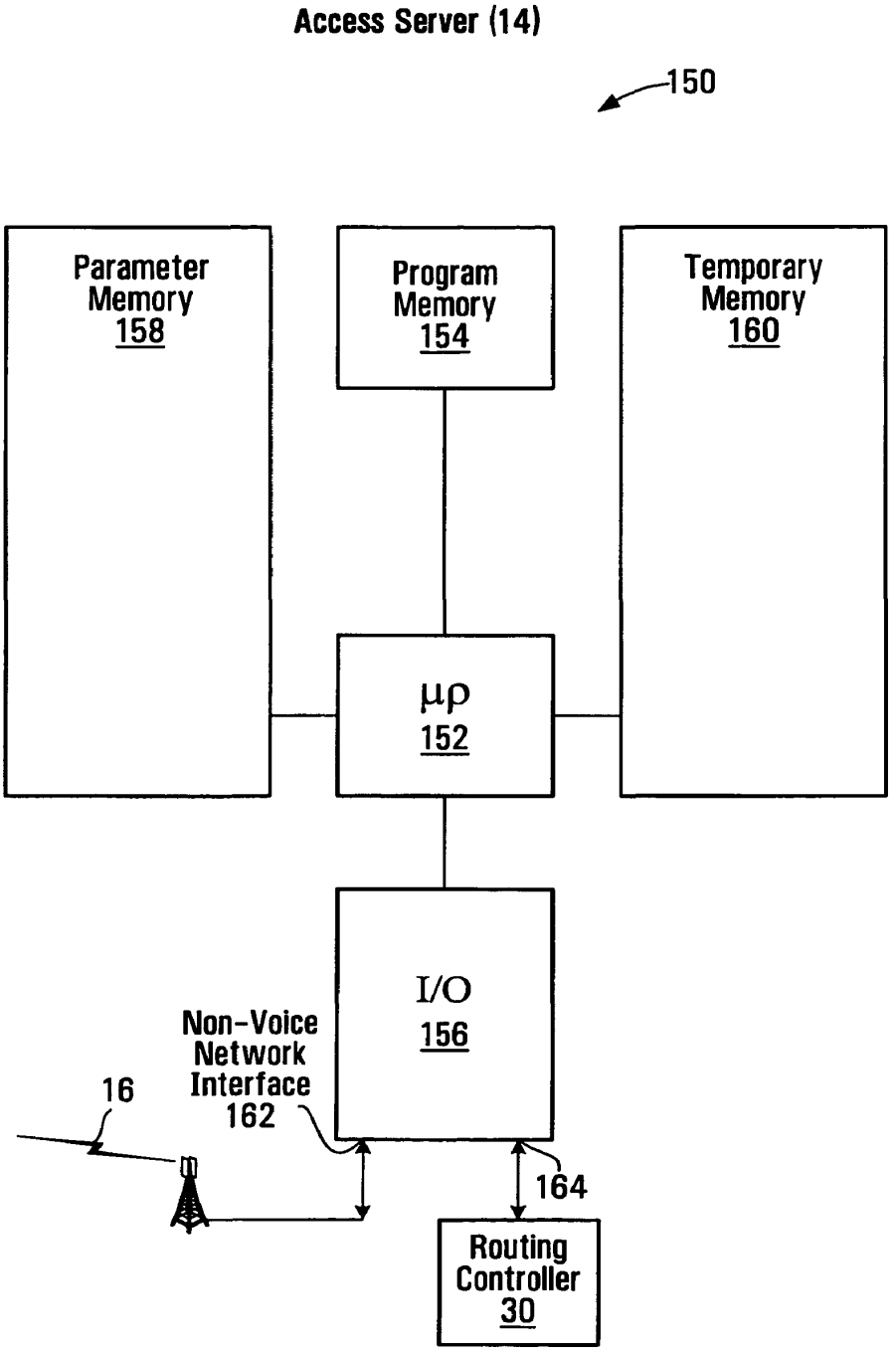

Access Server (14)
150
Parameter Memory 158
Program Memory 154
Temporary Memory 160
μρ 152
I/O 156
Non-Voice Network Interface 162
16
164
Routing Controller 30

FIG. 6

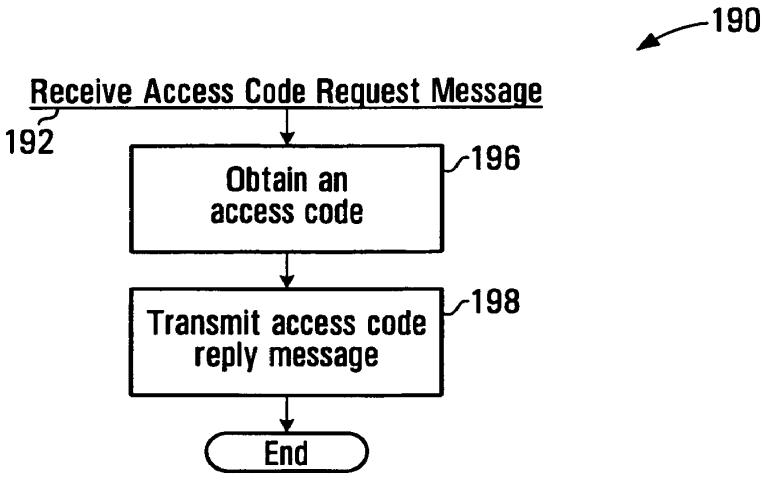

190
Receive Access Code Request Message
192
Obtain an access code
196
Transmit access code reply message
198
End

FIG. 7

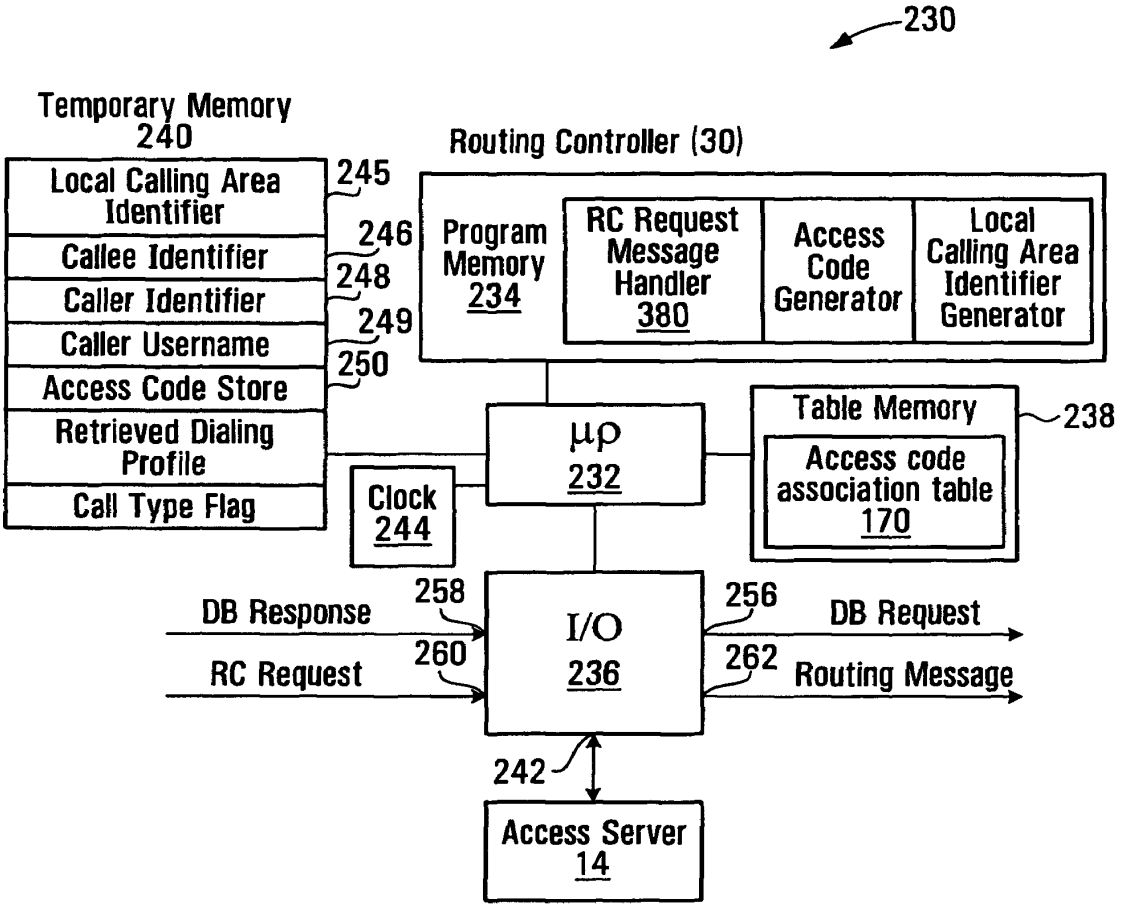

230
Temporary Memory
240
Local Calling Area Identifier
Callee Identifier
Caller Identifier
Caller Username
Access Code Store
Retrieved Dialing Profile
Call Type Flag
245
246
248
249
250
Routing Controller (30)
Program Memory 234
RC Request Message Handler 380
Access Code Generator
Local Calling Area Identifier Generator
μρ 232
Clock 244
Table Memory
238
Access code association table 170
DB Response
258
RC Request
260
I/O 236
256
DB Request
262
Routing Message
242
Access Server 14

FIG. 8

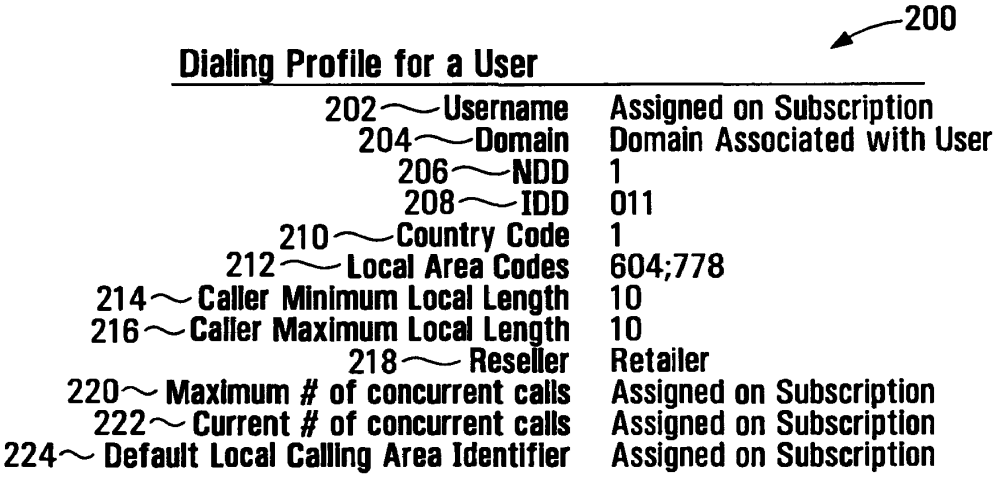

200
Dialing Profile for a User
202 Username
Assigned on Subscription
204 Domain
Domain Associated with User
206 NDD
1
208 IDD
011
210 Country Code
1
212 Local Area Codes
604;778
214 Caller Minimum Local Length
10
216 Caller Maximum Local Length
10
218 Reseller
Retailer
220 Maximum # of concurrent calls
Assigned on Subscription
222 Current # of concurrent calls
Assigned on Subscription
224 Default Local Calling Area Identifier
Assigned on Subscription

FIG. 9

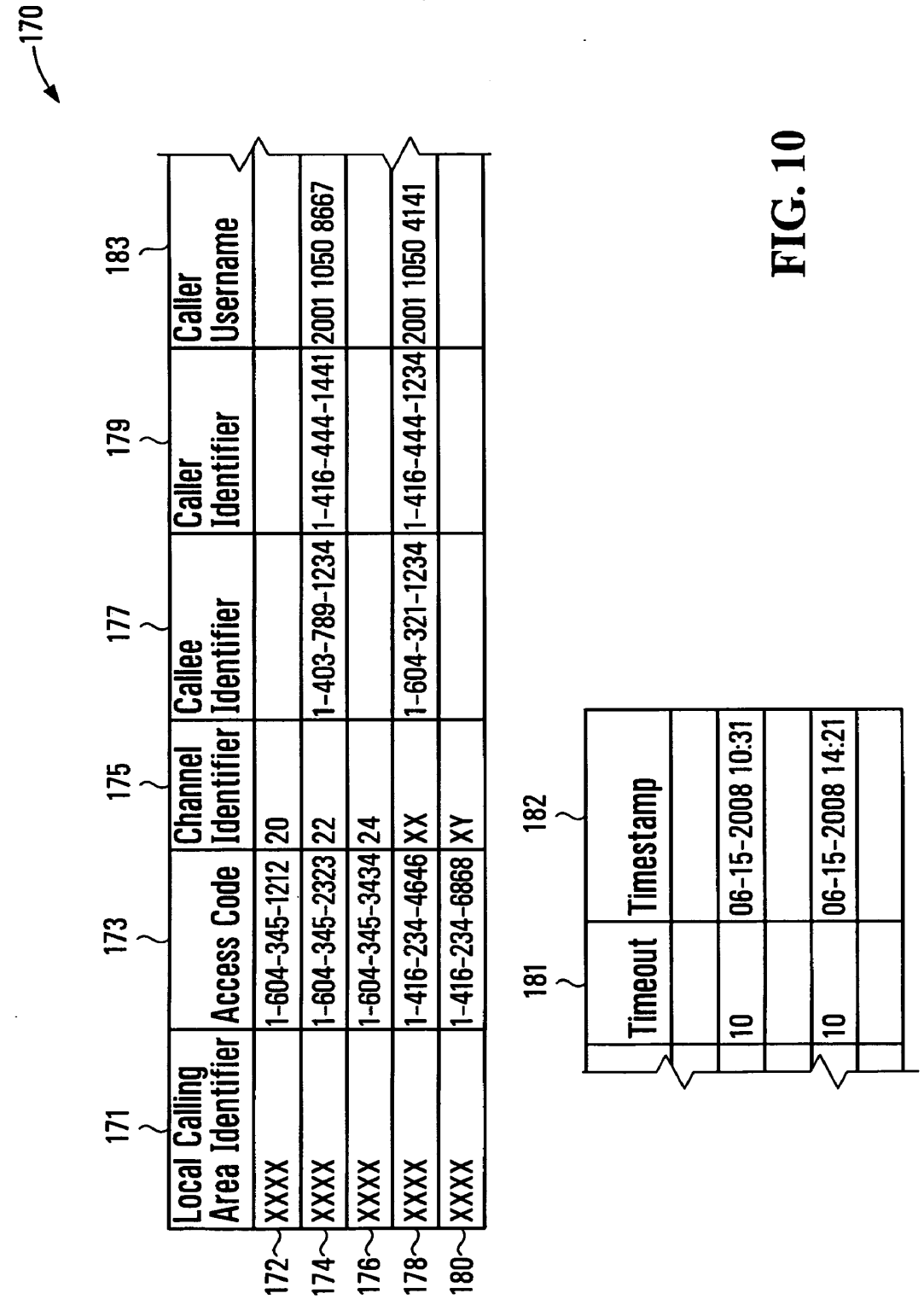

170
171
173
175
177
179
183
Local Calling Area Identifier
Access Code
Channel Identifier
Callee Identifier
Caller Identifier
Caller Username
172
174
176
178
180
XXXX
1-604-345-1212
20
1-604-345-2323
22
1-403-789-1234
1-416-444-1441
2001 1050 8667
1-604-345-3434
24
1-416-234-4646
XX
1-604-321-1234
1-416-444-1234
2001 1050 4141
1-416-234-6868
XY
181
182
Timeout
Timestamp
10
06-15-2008 10:31
10
06-15-2008 14:21

FIG. 10

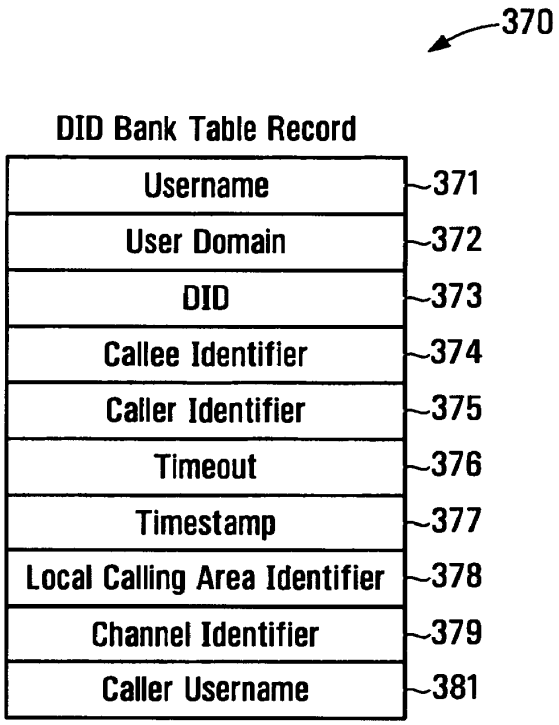

370
DID Bank Table Record
Username
371
User Domain
372
DID
373
Callee Identifier
374
Caller Identifier
375
Timeout
376
Timestamp
377
Local Calling Area Identifier
378
Channel Identifier
379
Caller Username
381

FIG. 11

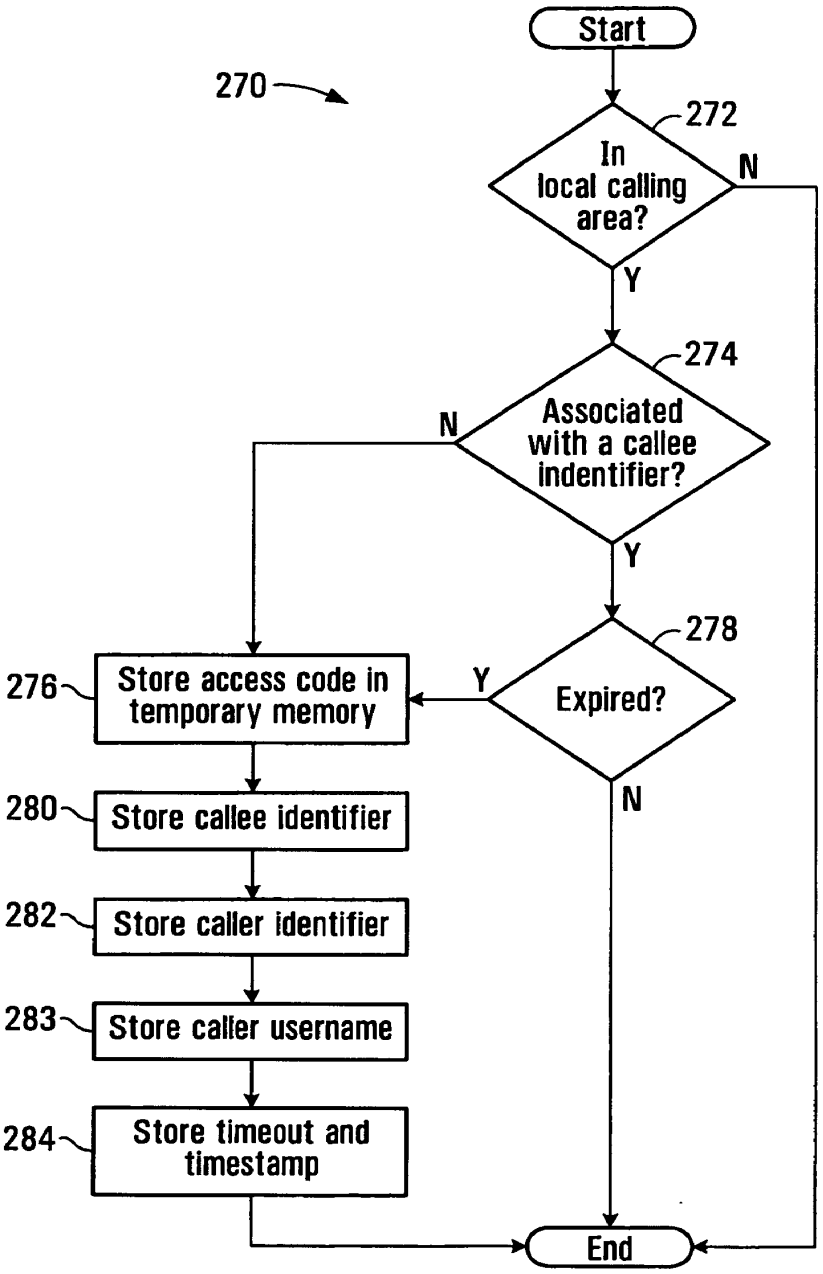

Start
270
272
In local calling area?
N
Y
274
Associated with a callee indentifier?
N
Y
278
Expired?
Y
N
276
Store access code in temporary memory
280
Store callee identifier
282
Store caller identifier
283
Store caller username
284
Store timeout and timestamp
End

FIG. 12

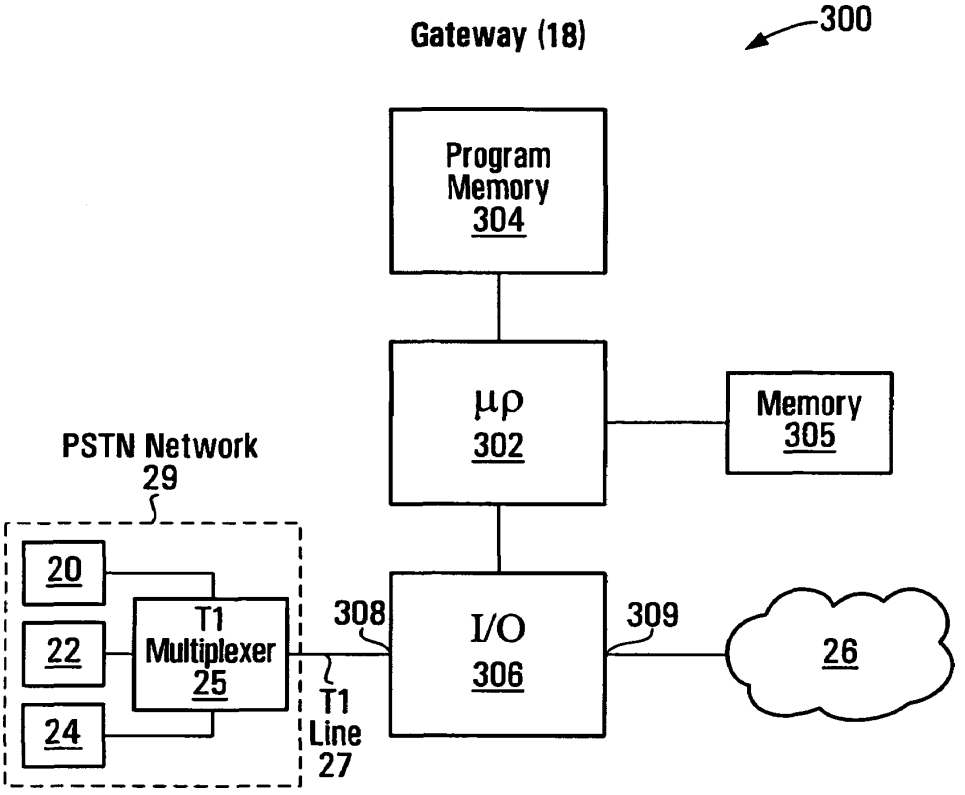


FIG. 13

310

SIP Invite Message

312	Caller Identifier	1-604-678-1234@20.14.102.5
314	Callee Identifier	1-604-345-1212
315	Digest Parameter	XXXXXXX
316	Call Identifer	FF10@20.14.102.5
317	IP Address	20.14.102.5
318	Gateway UDP Port	12378

FIG. 14

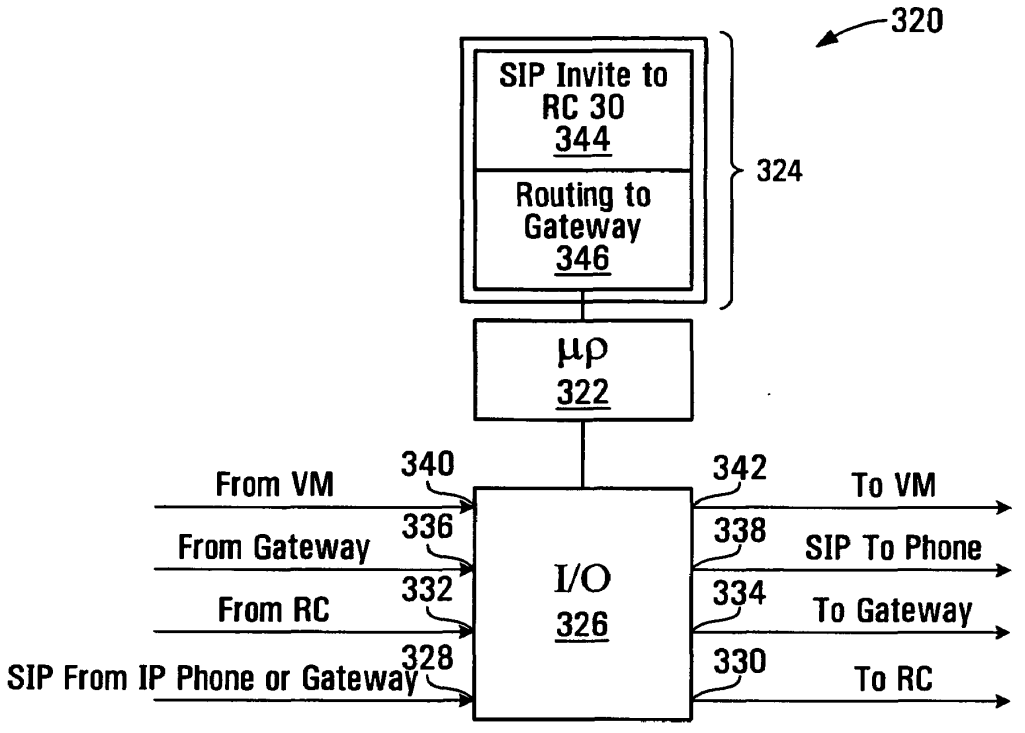

Call Controller (13)
320
SIP Invite to RC 30
344
Routing to Gateway
346
324
μρ
322
From VM
340
From Gateway
336
From RC
332
SIP From IP Phone or Gateway
328
I/O
326
342
To VM
338
SIP To Phone
334
To Gateway
330
To RC

FIG. 15

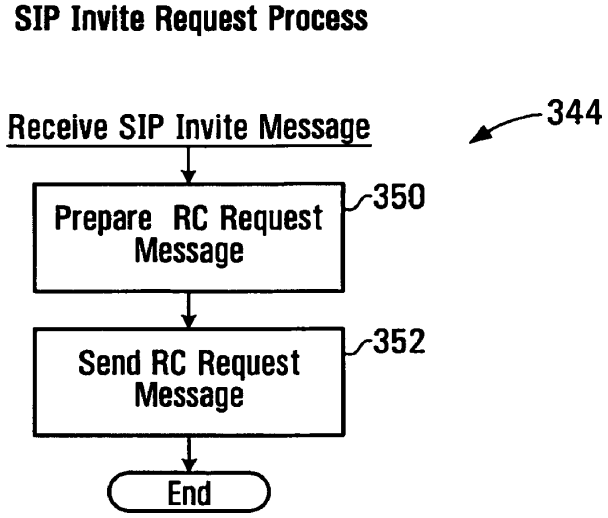

SIP Invite Request Process
Receive SIP Invite Message
344
Prepare RC Request Message
350
Send RC Request Message
352
End

FIG. 16

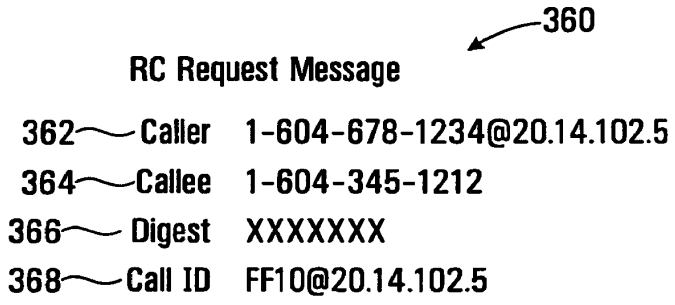

360
RC Request Message
362 Caller 1-604-678-1234@20.14.102.5
364 Callee 1-604-345-1212
366 Digest XXXXXXX
368 Call ID FF10@20.14.102.5

FIG. 17

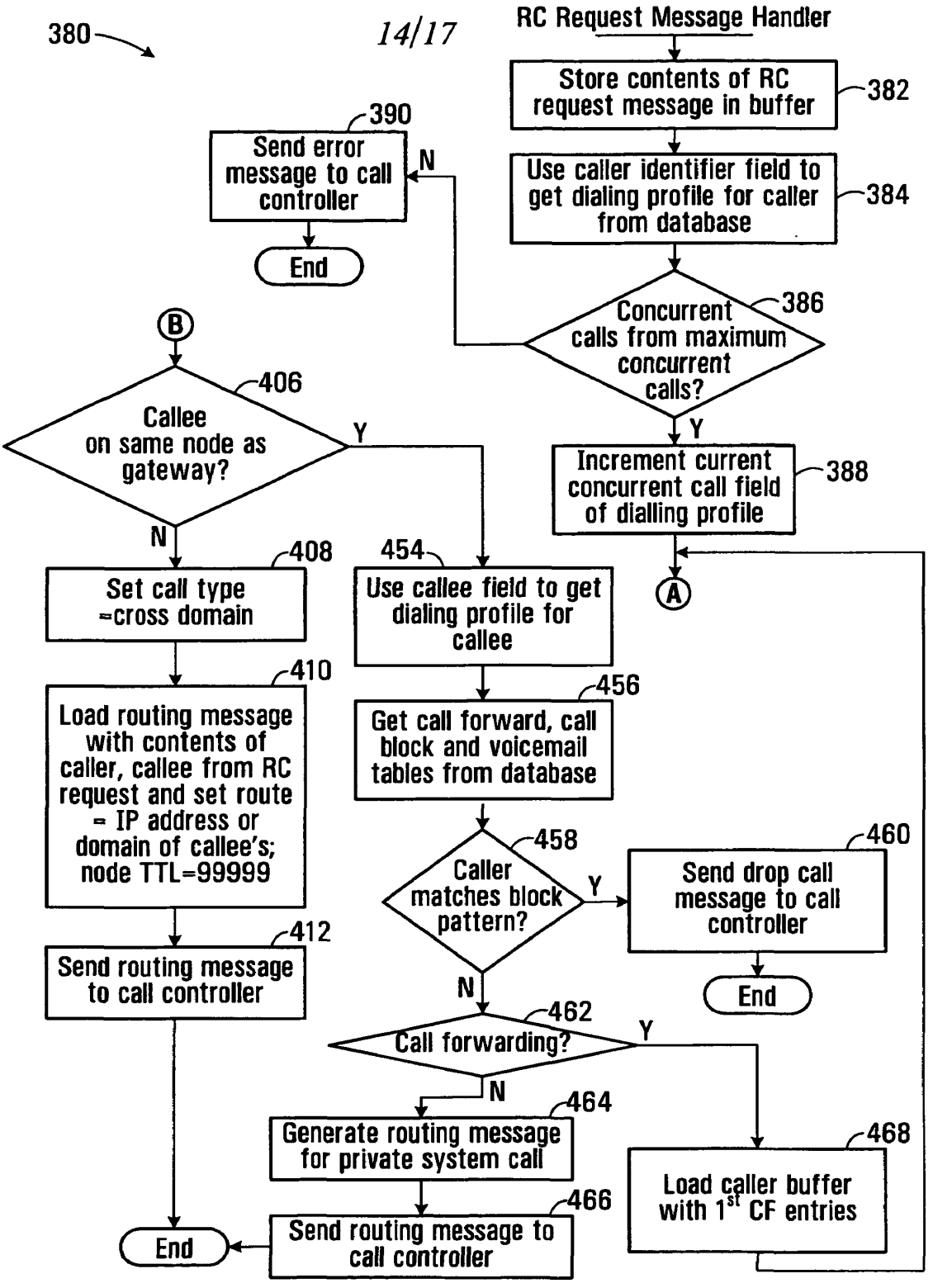

380
14/17
RC Request Message Handler
Store contents of RC request message in buffer
382
Use caller identifier field to get dialing profile for caller from database
384
Concurrent calls from maximum concurrent calls?
386
N
Y
Send error message to call controller
390
End
Increment current concurrent call field of dialling profile
388
A
B
Callee on same node as gateway?
406
Y
N
Set call type =cross domain
408
Load routing message with contents of caller, callee from RC request and set route = IP address or domain of callee's; node TTL=99999
410
Send routing message to call controller
412
End
454
Use callee field to get dialing profile for callee
456
Get call forward, call block and voicemail tables from database
458
Caller matches block pattern?
Y
N
Send drop call message to call controller
460
End
462
Call forwarding?
Y
N
464
Generate routing message for private system call
466
Send routing message to call controller
468
Load caller buffer with 1st CF entries

FIG. 18A

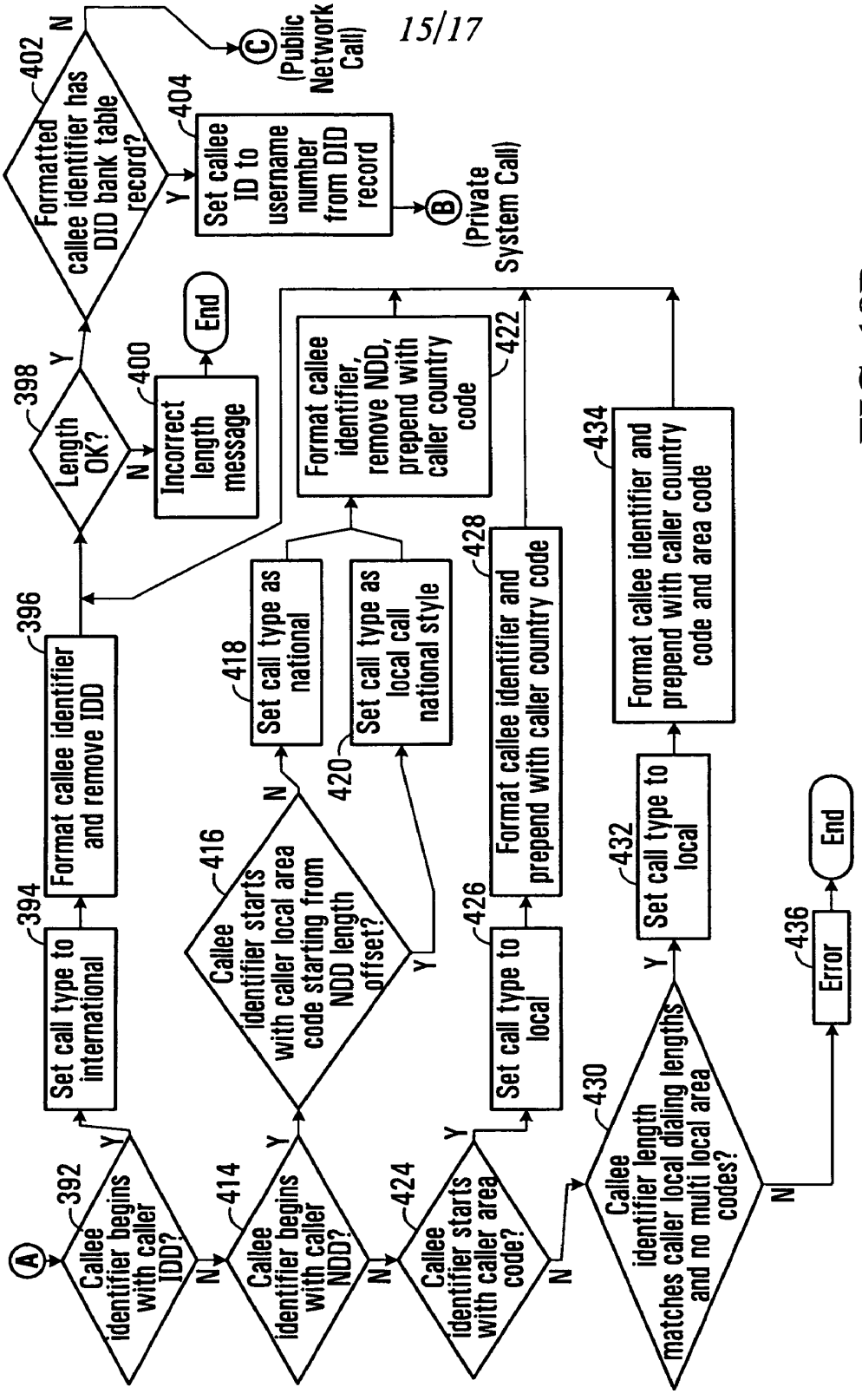

A
392
Callee identifier begins with caller IDD?
Y
394
Set call type to international
396
Format callee identifier and remove IDD
N
414
Callee identifier begins with caller NDD?
Y
416
Callee identifier starts with caller local area code starting from NDD length offset?
N
418
Set call type as national
420
Set call type as local call national style
Y
422
Format callee identifier, remove NDD, prepend with caller country code
N
424
Callee identifier starts with caller area code?
Y
426
Set call type to local
428
Format callee identifier and prepend with caller country code
N
430
Callee identifier length matches caller local dialing lengths and no multi local area codes?
Y
432
Set call type to local
434
Format callee identifier and prepend with caller country code and area code
N
436
Error
End
398
Length OK?
Y
N
400
Incorrect length message
End
402
Formatted callee identifier has DID bank table record?
Y
404
Set callee ID to username number from DID record
B
(Private System Call)
N
C
(Public Network Call)

FIG. 18B

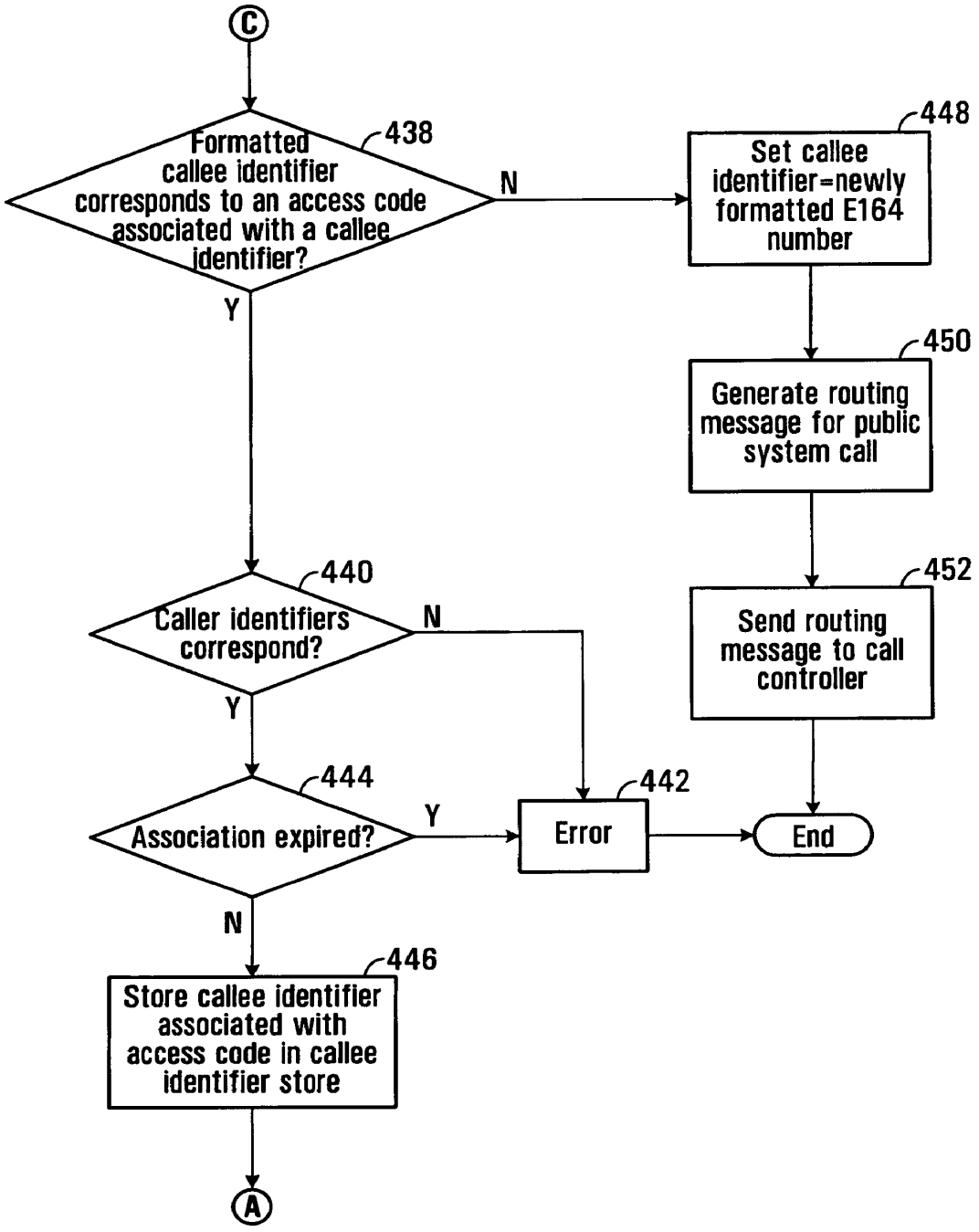

C
Formatted callee identifier corresponds to an access code associated with a callee identifier?
438
N
Y
448
Set callee identifier=newly formatted E164 number
450
Generate routing message for public system call
452
Send routing message to call controller
440
Caller identifiers correspond?
N
Y
444
Association expired?
Y
442
Error
End
N
446
Store callee identifier associated with access code in callee identifier store
A

FIG. 18C

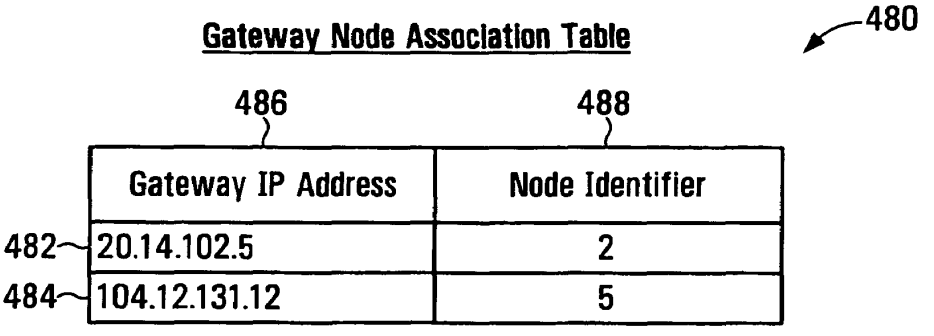

Gateway Node Association Table
480
486
488
Gateway IP Address
Node Identifier
482
20.14.102.5
2
484
104.12.131.12
5

FIG. 19

Document made available under the Patent Cooperation Treaty (PCT)

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Number: 61/129,898
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Remark: Priority document submitted or transmitted to the International Bureau in compliance with Rule 17.1(a) or (b)



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Organisation Mondiale de la Propriété Intellectuelle (OMPI) - Genève, Suisse

PCT

REQUEST

The undersigned requests that the present international application be processed according to the Patent Cooperation Treaty.

For receiving Office use only

PCT/CA2009/001062
International Application No.

28 JULY 2009 28.07.2009
International Filing Date

RO/CA
Name of receiving Office and "PCT International Application"

Applicant's or agent's file reference *(if desired) (12 characters maximum)* 83636-55

Box No. I TITLE OF INVENTION

MOBILE GATEWAY

Box No. II APPLICANT ☐ This person is also inventor

Name and address: ***(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country. The country of the address indicated in this Box is the applicant's State (that is, country) of residence if no State of residence is indicated below.)***

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State *(that is, country)* of nationality:
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CA

This person is applicant for the purposes of: ☐ all designated States ☒ all designated States except the United States of America ☐ the United States of America only ☐ the States indicated in the Supplemental Box

Box No. III FURTHER APPLICANT(S) AND/OR (FURTHER) INVENTOR(S)

☒ Further applicants and/or (further) inventors are indicated on a continuation sheet.

Box No. IV AGENT OR COMMON REPRESENTATIVE; OR ADDRESS FOR CORRESPONDENCE

The person identified below is hereby/has been appointed to act on behalf of the applicant(s) before the competent International Authorities as: ☒ agent ☐ common representative

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country.)*

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☐ **Address for correspondence:** Mark this check-box where no agent or common representative is/has been appointed and the space above is used instead to indicate a special address to which correspondence should be sent.

Sheet No. . . . 2 . . . .

Box No. III FURTHER APPLICANT(S) AND/OR (FURTHER) INVENTOR(S)

If none of the following sub-boxes is used, this sheet should not be included in the request.

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country. The country of the address indicated in this Box is the applicant's State (that is, country) of residence if no State of residence is indicated below.)*

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RPO South Granville Post
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This person is:
- ☐ applicant only
- ☒ applicant and inventor
- ☐ inventor only *(If this check-box is marked, do not fill in below.)*

Applicant's registration No. with the Office

State *(that is, country)* of nationality: SE

State *(that is, country)* of residence: CA

This person is applicant: ☐ all designated States ☐ all designated States except the United States of America ☒ the United States of America only ☐ the States indicated in the Supplemental

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country. The country of the address indicated in this Box is the applicant's State (that is, country) of residence if no State of residence is indicated below.)*

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New Westminster, British Columbia
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- ☒ applicant and inventor
- ☐ inventor only *(If this check-box is marked, do not fill in below.)*

Applicant's registration No. with the Office

State *(that is, country)* of nationality: UA

State *(that is, country)* of residence: CA

This person is applicant for the: ☐ all designated States ☐ all designated States except the United States of America ☒ the United States of America only ☐ the States indicated in the Supplemental

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country. The country of the address indicated in this Box is the applicant's State (that is, country) of residence if no State of residence is indicated below.)*

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This person is:
- ☐ applicant only
- ☒ applicant and inventor
- ☐ inventor only *(If this check-box is marked, do not fill in below.)*

Applicant's registration No. with the Office

State *(that is, country)* of nationality: CA

State *(that is, country)* of residence: CA

This person is applicant for the: ☐ all designated States ☐ all designated States except the United States of America ☒ the United States of America only ☐ the States indicated in the Supplemental

Name and address: *(Family name followed by given name; for a legal entity, full official designation. The address must include postal code and name of country. The country of the address indicated in this Box is the applicant's State (that is, country) of residence if no State of residence is indicated below.)*

This person is:
- ☐ applicant only
- ☐ applicant and inventor
- ☐ inventor only *(If this check-box is marked, do not fill in below.)*

Applicant's registration No. with the Office

State *(that is, country)* of nationality:

State *(that is, country)* of residence:

This person is applicant for the: ☐ all designated States ☐ all designated States except the United States of America ☐ the United States of America only ☐ the States indicated in the Supplemental

☐ Further applicants and/or (further) inventors are indicated on another continuation sheet.

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Sheet No. ...3...

Box No.V DESIGNATIONS

The filing of this request **constitutes under Rule 4.9(a), the designation** of all Contracting States bound by the PCT on the international filing date, for the grant of every kind of protection available and, where applicable, for the grant of both regional and national patents.

However,

☐ DE Germany **is not designated** for any kind of national protection

☐ JP Japan **is not designated** for any kind of national protection

☐ KR Republic of Korea **is not designated** for any kind of national protection

☐ RU Russian Federation **is not designated** for any kind of national protection

(The check-boxes above may only be used to exclude (irrevocably) the designations concerned if, at the time of filing or subsequently under Rule 26bis.1, the international application contains in Box No. VI a priority claim to an earlier national application filed in the particular State concerned, in order to avoid the ceasing of the effect, under the national law, of this earlier national application.)

Box No. VI PRIORITY CLAIM

The priority of the following earlier application(s) is hereby claimed:

Filing date of earlier application *(day/month/year)*	Number of earlier application	Where earlier application is:		
		national application: country or Member	regional application: regional Office	international application: receiving Office
item (1) 28/07/2008 28-July-2008	61/129,898	US		
item (2)				
item (3)				

☐ Further priority claims are indicated in the Supplemental Box.

Transmit certified copy: the receiving Office is requested to prepare and transmit to the International Bureau a certified copy of the earlier application(s) *(only if the earlier application was filed with the Office which for the purposes of this international application is the receiving Office)* identified above as:

☐ all items ☐ item (1) ☐ item (2) ☐ item (3) ☐ other, see Supplemental Box

Restore the right of priority: the receiving Office is requested to restore the right of priority for the earlier application(s) identified above or in the Supplemental Box as item(s) (_____). *(See also the Notes to Box No. VI; further information **must** be provided to support a request to restore the right of priority.)*

Incorporation by reference: where an element of the international application referred to in Article 11(1)(iii)(d) or (e) or a part of the description, claims or drawings referred to in Rule 20.5(a) is not otherwise contained in this international application but is completely contained in an earlier application whose priority is claimed on the date on which one or more elements referred to in Article 11(1)(iii) were first received by the receiving Office, that element or part is, subject to confirmation under Rule 20.6, incorporated by reference in this international application for the purposes of Rule 20.6.

Box No. VII INTERNATIONAL SEARCHING AUTHORITY

Choice of International Searching Authority (ISA) *(if two or more International Searching Authorities are competent to carry out the international search, indicate the Authority chosen; the two-letter code may be used):*

ISA/ CA .....

Request to use results of earlier search; reference to that search *(if an earlier search has been carried out by or requested from the International Searching Authority):*

Date *(day/month/year)* Number Country *(or regional Office)*

Box No. VIII DECLARATIONS

The following **declarations** are contained in Boxes Nos. VIII (i) to (v) *(mark the applicable check-boxes below and indicate in the right column the number of each type of declaration)*:			Number of declarations
☒ Box No. VIII (i)	Declaration as to the identity of the inventor	:	3
☒ Box No. VIII (ii)	Declaration as to the applicant's entitlement, as at the international filing date, to apply for and be granted a patent	:	1
☐ Box No. VIII (iii)	Declaration as to the applicant's entitlement, as at the international filing date, to claim the priority of the earlier application	:	
☒ Box No. VIII (iv)	Declaration of inventorship (only for the purposes of the designation of the United States of America)	:	3
☐ Box No. VIII (v)	Declaration as to non-prejudicial disclosures or exceptions to lack of novelty	:	

Sheet No. 8 11

Box No. IX CHECK LIST; LANGUAGE OF FILING

This international application **contains:**

(a) **on paper**, the following number of sheets:

request (including declaration and supplemental sheets)	8	13
description (excluding sequence listing and/or tables related thereto)	62	
claims	18	
abstract	1	
drawings	17	
Sub-total number of sheets	106	
sequence listing		111
tables related thereto		
(for both, actual number of sheets if filed on paper, whether or not also filed in electronic form; see (c) below)		
Total number of sheets	106	

(b) ☐ **only in electronic form** (Section 801(a)(i)) 111
- (i) ☐ sequence listing
- (ii) ☐ tables related thereto

(c) ☐ **also in electronic form** (Section 801(a)(ii))
- (i) ☐ sequence listing
- (ii) ☐ tables related thereto

Type and number of carriers (diskette, CD-ROM, CD-R or other) on which are contained the
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This international application is **accompanied by** the following item(s) *(mark the applicable check-boxes below and indicate in right column the number of each item)*: — Number of items

1. ☐ fee calculation sheet :
2. ☐ original separate power of attorney :
3. ☐ original general power of attorney :
4. ☐ copy of general power of attorney; reference number, if any: ..... :
5. ☐ statement explaining lack of signature :
6. ☐ priority document(s) identified in Box No. VI as item(s): ..... :
7. ☐ translation of international application into *(language)*: ..... :
8. ☐ separate indications concerning deposited microorganism or other biological material :
9. ☐ sequence listing in electronic form *(indicate type and number of carriers)*
   - (i) ☐ copy submitted for the purposes of international search under Rule 13*ter* only (and not as part of the international application) :
   - (ii) ☐ *(only where check-box (b)(i) or (c)(i) is marked in left column)* additional copies including, where applicable, the copy for the purposes of international search under Rule 13*ter* :
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11. ☐ other *(specify)*: :

Figure of the drawings which should accompany the abstract: 1

Language of filing of the international application: English

Box No. X SIGNATURE OF APPLICANT, AGENT OR COMMON REPRESENTATIVE

Next to each signature, indicate the name of the person signing and the capacity in which the person signs (if such capacity is not obvious from reading the request).

X

BJORSELL, Johan Emil Viktor | SOBOLYEV, Maksym | HUTTUNEN, Pentti Kalevi for Digifonica (International) Limited

Date: 28th July 2009

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1. Date of actual receipt of the purported international application: 28 JULY 2009 28.07.2009	2. Drawings: ☒ received: ☐ not received:
3. Corrected date of actual receipt due to later but timely received papers or drawings completing the purported international application:	
4. Date of timely receipt of the required corrections under PCT Article 11(2):	
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META 1002
META V. VOIP-PAL

Sheet No. 8 12

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BJORSELL, Johan Emil Viktor SOBOLYEV, Maksym HUTTUNEN, Pentti Kalevi for Digifonica (International) Limited

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BJORSELL, Johan Emil Viktor; SOBOLYEV, Maksym; HUTTUNEN, Pentti Kalevi for Digifonica (International) Limited

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META 1002
META V. VOIP-PAL

(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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61/129,898 28 July 2008 (28.07.2008) US

(71) Applicant *(for all designated States except US)*: **DIGIFONICA (INTERNATIONAL) LIMITED** [CA/CA]; Suite #890, 999 West Hastings Street, Vancouver, British Columbia V6C 2W2 (CA).

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(75) Inventors/Applicants *(for US only)*: **BJORSELL, Johan, Emil, Viktor** [SE/CA]; PO Box 29164, RPO South Granville Post, Vancouver, British Columbia V6J 0A6 (CA). **SOBOLYEV, Maksym** [UA/CA]; 506-210 Eleventh Street, New Westminster, British Columbia V3M 4C9 (CA). **HUTTUNEN, Pentti, Kalevi** [CA/CA]; 76-3180 East 58th Avenue, Vancouver, British Columbia V5S 3S8 (CA).

(74) Agent: **KNOX, John, W.**; SMART & BIGGAR, 2200-650 West Georgia Street, Vancouver, British Columbia V6B 4N8 (CA).

(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AO, AT, AU, AZ, BA, BB, BG, BH, BR, BW, BY, BZ, CA, CH, CL, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PE, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, ST, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

(84) Designated States *(unless otherwise indicated, for every kind of regional protection available)*: ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HR, HU, IE, IS, IT, LT, LU, LV, MC, MK, MT, NL, NO, PL, PT, RO, SE, SI, SK, SM, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Declarations under Rule 4.17:

— *as to the identity of the inventor (Rule 4.17(i))*

— *as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii))*

— *of inventorship (Rule 4.17(iv))*

[Continued on next page]

(54) **Title**: MOBILE GATEWAY

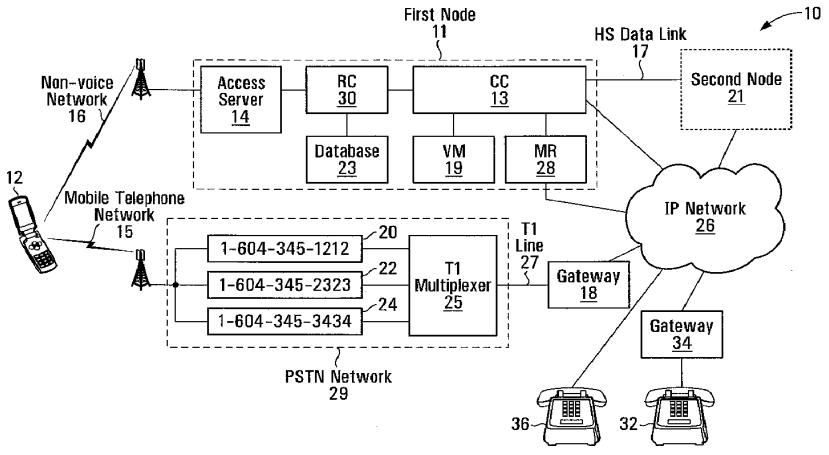


FIG. 1

(57) **Abstract**: A method of initiating a call to a callee using a mobile telephone involves: receiving, from a user of the mobile telephone, a callee identifier associated with the callee; transmitting an access code request message to an access server, said access code request message including said callee identifier; receiving an access code reply message from the access server in response to said access code request message, said access code reply message including an access code different from said callee identifier and associated with said callee identifier; and initiating a call with the mobile telephone using said access code to identify the callee.

WO 2010/012090 A2

Published:

— *without international search report and to be republished upon receipt of that report (Rule 48.2(g))*

MOBILE GATEWAY

BACKGROUND OF THE INVENTION

1. Field of Invention

This invention relates generally to telecommunication, and more particularly to methods, systems, apparatuses, and computer readable media for initiating or enabling a call with a mobile telephone to a callee.

2. Description of Related Art

Mobile telephone service providers often charge significant fees for long distance telephone calls, particularly when the mobile telephone is roaming in another mobile telephone service provider's network.

One known technique for avoiding the long distance charges of mobile telephone service providers is to use a "calling card". A "calling card" may permit the user of the mobile telephone to place a call to a local telephone number or to a less-expensive telephone number (such as a toll-free number, for example) instead of placing the call directly to the callee. The user may thus avoid the long distance charges of the mobile telephone service provider, which may be higher than the charges for using the "calling card". However, this technique can be cumbersome and undesirable, because it may require the user of the mobile telephone to follow a number of complicated or cumbersome steps in order to initiate a call to the callee, for example.

SUMMARY OF THE INVENTION

In accordance with one aspect of the invention there is provided a method of initiating a call to a callee using a mobile telephone. The method involves receiving, from a user of the mobile telephone, a callee identifier associated with the callee; transmitting an access code request message to an access server, the access code request message including the callee identifier; receiving an access code reply message from the access server in response to the access code request message, the access code reply message including an access code different from the callee identifier and associated

with the callee identifier; and initiating a call with the mobile telephone using the access code to identify the callee.

Transmitting may involve transmitting the access code request message to the access server on a non-voice network.

Transmitting may involve transmitting a location identifier of a location associated with the mobile telephone to the access server.

Transmitting the location identifier may involve transmitting an IP address of the mobile telephone in a wireless IP network.

Transmitting the location identifier may involve transmitting an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

Transmitting the location identifier may involve transmitting a user-configured identifier of a location associated with the mobile telephone.

Receiving the access code reply message may involve receiving the access code reply message from the access server on a non-voice network.

Receiving the access code reply message may involve receiving, in the access code reply message, an access code temporarily associated with the callee identifier.

Receiving the access code reply message may involve receiving, in the access code reply message, a telephone number identifying a channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

Initiating the call may involve engaging a routing controller to route the call on the IP network to the callee.

The method may further involve: receiving from the mobile telephone the access code request message; communicating with a routing controller to obtain from the routing controller the access code wherein the access code identifies a channel and is useable by the mobile telephone to cause the routing controller to establish a call to the callee using the channel; and transmitting the access code reply message to the mobile telephone.

In accordance with another aspect of the invention, there is provided a mobile telephone. The mobile telephone includes: provisions for receiving, from a user of the mobile telephone, a callee identifier associated with the callee; transmitting provisions for transmitting an access code request message to an access server, the access code request message including the callee identifier; provisions for receiving an access code reply message from the access server in response to the access code request message, the access code reply message including an access code different from the callee identifier and associated with the callee identifier; and provisions for initiating a call using the access code to identify the callee.

The transmitting provisions may include a non-voice network interface for transmitting the access code request message to the access server on a non-voice network.

The access code request message may further include a location identifier of a location associated with the mobile telephone.

The location identifier may include an IP address of the mobile telephone in a wireless IP network.

The location identifier may include an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

The location identifier may include a user-configured identifier of a location associated with the mobile telephone.

The provisions for receiving an access code reply message may include a non-voice network interface for receiving the access code reply message on a non-voice network.

The access code may include a telephone number.

The means for initiating may involve a mobile telephone network interface.

In accordance with another aspect of the invention, there is provided a system for initiating a call to a callee. The system includes the mobile telephone, a routing controller, and an access server. The access server includes: provisions for receiving from the mobile telephone the access code request message; provisions for communicating with the routing controller to obtain from the routing controller the access code wherein the access code identifies a channel and is useable by the mobile telephone to cause the routing controller to establish a call to the callee using the channel; and provisions for transmitting the access code reply message including the access code to the mobile telephone.

In accordance with another aspect of the invention, there is provided a mobile telephone. The mobile telephone includes a processor circuit, a network interface in communication with the processor circuit, and a computer readable medium in communication with the processor circuit and encoded with codes for directing the processor circuit to: receive, from a user of the mobile telephone, a callee identifier associated with the callee; cause an access code request message to be transmitted to an access server, the access code request message including the callee identifier; receive an access code reply message from the access server in response to the access code request message, the access code reply message including an access

code different from the callee identifier and associated with the callee identifier; and initiate a call using the access code to identify the callee.

The network interface may include a non-voice network interface, and the codes for directing the processor circuit to cause the access code request message to be transmitted may include codes for directing the processor circuit to cause the access code request message to be transmitted to the access server using the non-voice network interface on a non-voice network.

The access code request message may further include a location identifier of a location associated with the mobile telephone.

The location identifier may include an IP address of the mobile telephone in a wireless IP network.

The location identifier may include an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

The location identifier may include a user-configured identifier of a location associated with the mobile telephone.

The network interface may include a non-voice network interface, and the codes for directing the processor circuit to receive an access code reply message may include codes for directing the processor circuit to cause the access code reply message to be received from the access server using the non-voice network interface on a non-voice network.

The access code may include a telephone number identifying a channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

The network interface may include a mobile telephone network interface, and the codes for directing the processor circuit to initiate may include codes for

directing the processor circuit to cause a call to be initiated using the mobile telephone network interface on a mobile telephone network.

In accordance with another aspect of the invention, there is provided a system for initiating a call to a callee. The system includes: the mobile telephone; a routing controller; and an access server comprising a processor circuit and a computer readable medium in communication with the processor circuit. The computer readable medium is encoded with codes for directing the processor circuit to: receive from the mobile telephone the access code request message; communicate with the routing controller to obtain from the routing controller the access code wherein the access code identifies a channel and is useable by the mobile telephone to cause the routing controller to establish a call to the callee using the channel; and transmit the access code reply message to the mobile telephone.

In accordance with another aspect of the invention, there is provided a computer readable medium encoded with codes for directing a processor circuit to: receive, from a user of a mobile telephone, a callee identifier associated with a callee; transmit an access code request message to an access server, the access code request message including the callee identifier; receive an access code reply message from the access server in response to the access code request message, the access code reply message including an access code different from the callee identifier and associated with the callee identifier; and initiate a call using the access code to identify the callee.

In accordance with another aspect of the invention, there is provided a method for enabling a mobile telephone to initiate a call to a callee through a channel. The method involves: receiving from the mobile telephone an access code request message including a callee identifier associated with the callee; communicating with a routing controller to obtain from the routing controller an access code identifying the channel, the access code being different from the callee identifier and useable by the mobile telephone to

initiate a call to the callee using the channel; and transmitting an access code reply message including the access code to the mobile telephone.

Receiving may involve receiving the access code request message on a non-voice network.

The method may further involve causing the routing controller to produce the access code.

Producing may involve selecting the access code from a pool of access codes, where each access code in the pool of access codes identifies a respective telephone number.

The method may further involve determining a local calling area associated with the mobile telephone.

Determining may involve accessing a dialing profile associated with the caller, the dialing profile including a location field having contents identifying at least a default location of the caller.

Determining may involve receiving an IP address of the mobile telephone in a wireless IP network.

Determining may involve receiving an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

Determining may involve receiving a user-configured identifier of a location associated with the mobile telephone.

Selecting may involve selecting an access code in the local calling area associated with the mobile telephone.

Each access code in the pool of access codes may further identify a respective channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

The method may further involve causing the routing controller to establish communication through the IP network in response to a call received on the channel.

Producing may further involve storing a caller identifier associated with the mobile telephone in association with the access code.

Causing the routing controller to establish communication may involve causing the routing controller to establish communication only if the caller identifier associated with the access code identifies the mobile telephone.

Producing may further involve storing the callee identifier in association with the access code.

Producing may further involve searching the pool of access codes for an access code associated with the callee identifier to identify the channel usable by the mobile telephone to initiate a call to the callee.

Producing may further involve storing, in association with the access code, a timestamp for use in determining when the usability of the access code to initiate a call to the callee will expire.

Causing the routing controller to establish communication may involve causing the routing controller to establish communication only if the usability of the access code to initiate a call to the callee has not expired.

Transmitting may involve transmitting the access code reply message on a non-voice network.

In accordance with another aspect of the invention, there is provided a system for enabling a mobile telephone to initiate a call to a callee through a channel. The system includes: provisions for receiving from the mobile telephone an access code request message including a callee identifier associated with the callee; provisions for communicating with the routing controller to obtain from the routing controller an access code identifying the channel, the access code being different from the callee identifier and useable by the mobile telephone to initiate a call to the callee using the channel; and provisions for transmitting an access code reply message including the access code to the mobile telephone.

The provisions for receiving may include a non-voice network interface for receiving the access code request message on a non-voice network.

The system may further include provisions for producing the access code.

The provisions for producing may include a processor circuit operably configured to select the access code from a pool of access codes, where each access code in the pool of access codes identifies a respective telephone number.

The processor circuit may be operably configured to determine a local calling area associated with the mobile telephone.

The processor circuit may be operably configured to determine a local calling area associated with the mobile telephone using a dialing profile associated with the caller, the dialing profile including a location field having contents identifying at least a default location of the caller.

The processor circuit may be operably configured to determine a local calling area associated with the mobile telephone using an IP address of the mobile telephone in a wireless IP network.

The processor circuit may be operably configured to determine a local calling area associated with the mobile telephone using an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

The processor circuit may be operably configured to determine a local calling area associated with the mobile telephone using a user-configured identifier of a location associated with the mobile telephone.

The processor circuit may be operably configured to select an access code in the local calling area associated with the mobile telephone.

Each access code in the pool of access codes may further identify a respective channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

The processor circuit may be operably configured to establish communication through the IP network in response to a call received on the channel.

The processor circuit may be operably configured to store a caller identifier associated with the mobile telephone in association with the access code.

The processor circuit may be operably configured to cause the routing controller to establish communication only if the caller identifier associated with the access code identifies the mobile telephone.

The processor circuit may be operably configured to store the callee identifier in association with the access code.

The processor circuit may be operably configured to search the pool of access codes for an access code associated with the callee identifier to

identify the channel usable by the mobile telephone to initiate a call to the callee.

The processor circuit may be operably configured to store, in association with the access code, a timestamp for use in determining when the usability of the access code to initiate a call to the callee will expire.

The processor circuit may operably configured to establish communication only if the usability of the access code to initiate a call to the callee has not expired.

The provisions for transmitting may include a non-voice network interface for transmitting the access code reply message on a non-voice network.

In accordance with another aspect of the invention, there is provided a system for enabling a mobile telephone to initiate a call to a callee through a channel. The system includes a processor circuit, a network interface in communication with the processor circuit, and a computer readable medium in communication with the processor circuit and encoded with codes for directing the processor circuit to: receive from the mobile telephone an access code request message including a callee identifier associated with the callee; communicate with the routing controller to obtain from the routing controller an access code identifying the channel, the access code being different from the callee identifier and useable by the mobile telephone to initiate a call to the callee using the channel; and cause an access code reply message including the access code to be transmitted to the mobile telephone.

The network interface may include a non-voice network interface, and the codes for directing the processor circuit to receive may include codes for directing the processor circuit to cause the access code request message to be received using the non-voice network interface on a non-voice network.

The computer readable medium may be further encoded with codes for directing the processor circuit to cause the access code to be produced.

The codes for directing the processor circuit to cause the access code to be produced may cause the access code to be selected from a pool of access codes, where each access code in the pool of access codes identifies a respective telephone number.

The computer readable medium may be further encoded with codes for directing the processor circuit to cause to be determined a local calling area associated with the mobile telephone.

The codes for directing the processor circuit to cause to be determined may cause a dialing profile associated with the caller to be accessed, the dialing profile including a location field having contents identifying at least a default location of the caller.

The codes for directing the processor circuit to cause to be determined may cause to be received an IP address of the mobile telephone in a wireless IP network.

The codes for directing the processor circuit to cause to be determined may cause to be received an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

The codes for directing the processor circuit to cause to be determined may cause to be received a user-configured identifier of a location associated with the mobile telephone.

The codes for directing the processor circuit to cause the access code to be produced may further cause to be selected an access code in the local calling area associated with the mobile telephone.

Each access code in the pool of access codes may further identify a respective channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

The computer readable medium may be further encoded with codes for directing the processor circuit to cause communication through the IP network to be established in response to a call received on the channel.

The codes for directing the processor circuit to cause the access code to be produced may cause a caller identifier associated with the mobile telephone to be stored in association with the access code.

The codes for directing the processor circuit to cause communication to be established may cause communication to be established only if the caller identifier associated with the access code identifies the mobile telephone.

The codes for directing the processor circuit to cause the access code to be produced may cause the callee identifier to be stored in association with the access code.

The codes for directing the processor circuit to cause the access code to be produced may cause the pool of access codes to be searched for an access code associated with the callee identifier to identify the channel usable by the mobile telephone to initiate a call to the callee.

The codes for directing the processor circuit to cause the access code to be produced may cause a timestamp for use in determining when the usability of the access code to initiate a call to the callee will expire, to be stored in association with the access code..

The codes for directing the processor circuit to cause communication to be established may cause communication to be established only if the usability of the access code to initiate a call to the callee has not expired.

The network interface may include a non-voice network interface, and the codes for directing the processor circuit to transmit include codes for directing the processor circuit to cause the access code reply message to be transmitted using the non-voice network interface on a non-voice network.

In accordance with another aspect of the invention, there is provided a computer readable medium encoded with codes for directing a processor circuit to: receive from the mobile telephone an access code request message including a callee identifier associated with the callee; communicate with the routing controller to obtain from the routing controller an access code identifying the channel, the access code being different from the callee identifier and useable by the mobile telephone to initiate a call to the callee using the channel; and cause an access code reply message including the access code to be transmitted to the mobile telephone.

Other aspects and features of the present invention will become apparent to those ordinarily skilled in the art upon review of the following description of specific embodiments of the invention in conjunction with the accompanying figures.

BRIEF DESCRIPTION OF THE DRAWINGS

In drawings which illustrate embodiments of the invention,

Figure **1** is a block diagram of a system for enabling a mobile telephone to initiate a call through a channel to a callee in accordance with a first embodiment in the invention;

Figure **2** is a block diagram of mobile telephone shown in Figure **1**;

Figure **3** is a flow chart of a process executed by the mobile telephone shown in Figure **1**;

Figure **4** is a schematic representation of an access code request message transmitted between the mobile telephone and an access server shown in Figure **1**;

Figure **5** is a schematic representation of an access code reply message transmitted between the mobile telephone and the access server shown in Figure **1**;

Figure **6** is a block diagram of the access server shown in Figure **1**;

Figure **7** is a flow chart of a process executed by the access server shown in Figure **1**;

Figure **8** is a block diagram of a routing controller shown in Figure **1**;

Figure **9** is a tabular representation of a dialing profile stored in a database accessible by the routing controller illustrated in Figure **1**;

Figure **10** is a tabular representation of an access code association table stored in memory accessible by the routing controller shown in Figure **1**;

Figure **11** is a schematic representation of a DID bank table record stored in a database shown in Figure **1**;

Figure **12** is a flow chart of a process executed by the routing controller illustrated in Figure **1**;

Figure **13** is a block diagram of a gateway shown in Figure **1**;

Figure **14** is a tabular representation of an SIP invite message transmitted between the gateway and a call controller illustrated in Figure **1**;

Figure **15** is a block diagram of the call controller illustrated in Figure **1**;

Figure **16** is a flow chart of a process executed by the call controller illustrated in Figure **1**;

Figure **17** is a tabular representation of an RC request message transmitted between the call controller and the routing controller illustrated in Figure **1**

Figures **18**A–**18**C are a flow chart of a process executed by the routing controller illustrated in Figure **1**; and

Figure **19** is a tabular representation of a gateway node association table stored in the database illustrated in Figure **1**.

DETAILED DESCRIPTION

Referring to Figure **1**, a system for enabling a mobile telephone to initiate a call to a callee is shown generally at **10**. The system **10** includes a first node **11**, a second node **21**, and a mobile telephone **12**.

The first and second nodes **11** and **21** in the illustrated embodiment may support "voice-over-IP" (VoIP) calls between telephones and/or videophones using the internet protocol (IP), as described in PCT Publication No. WO **2008/052340**, which is hereby incorporated by reference in its entirety herein. In the embodiment shown, the first node **11** is located in a geographical area, such as Vancouver, British Columbia, Canada, for example, and the second node **21** is located in London, England, for example. Different nodes may be located in different geographical regions throughout the world to provide telephone/videophone service to subscribers in respective regions. These nodes may be in communication with each other by high speed/high data throughput links including optical fiber, satellite, and/or cable links illustrated generally at **17**, forming a backbone to the system. These nodes may alternatively, or in addition, be in communication with each other through conventional internet services.

In the embodiment shown, the first node **11** provides telephone/videophone service to western Canadian customers from Vancouver Island to Ontario. Another node (not shown) may be located in Eastern Canada to provide services to subscribers in that area, for example.

Other nodes of the type shown may also be employed within the geographical area serviced by a node to provide for call load sharing, for example, within a region of the geographical area serviced by the node. However, in general, all nodes may be similar and have the properties described in connection with the first node **11**.

In this embodiment, the first node **11** includes a call controller (CC) **13**, an access server **14**, a routing controller (RC) **30,** a database **23**, a voicemail server **19,** and a media relay **28**. Each of these may be implemented as separate modules on a common computer system or by separate computers, for example. The voicemail server **19** need not be included in the node and can be provided by a third party service provider. Although the access server **14** is illustrated as being part of the first node **11**, access servers in alternative embodiments may be separate from the node and may be in communication with one or more nodes, for example.

The mobile telephone **12** is configured to place calls over a mobile telephone network, illustrated generally at **15**, in a manner well-known in the art. Furthermore, the mobile telephone **12** and the access server **14** are configured to communicate with each other, preferably on a non-voice network illustrated generally at **16**, such as a "WiFi" wireless IP network or a General Packet Radio Service (GPRS) network, for example. However, in alternative embodiments, the mobile telephone **12** and the access server **14** may communicate with each other over other networks, such as a mobile telephone network using Short Message Service (SMS) messages, for example.

The system **10** further includes a gateway **18** in communication with at least one, and preferably a plurality of, channels, which are illustrated schematically at **20**, **22**, and **24**, to which the mobile telephone **12** may initiate a call over the mobile telephone network **15**. The channels **20**, **22**, and **24** maybe telephone lines in a Public Switched Telephone Network (PSTN) **29**. The channels **20**, **22**, and **24** maybe associated with PSTN telephone numbers in a local calling area associated with the mobile telephone **12**, and thus these channels preferably depend on a geographical location of the mobile telephone. The expression "local calling area" herein refers generally to a set of telephone numbers, typically defined by a geographical region, to which telephone calls may be placed by callers within the local calling area at either no additional charge or at a lower additional charge than would be required for calls to

numbers that are outside of the local calling area. However, it will be appreciated that in other embodiments, the gateway **18** may be in communication with any number of channels, which need not be PSTN telephone lines. Also, in the illustrated embodiment, the channels **20**, **22**, and **24** are associated with telephone numbers for Vancouver, British Columbia, Canada and the surrounding area, although it will be appreciated that these channels may include PSTN telephone lines associated with other areas, for example, which may not necessarily be in a local calling area associated with the mobile telephone **12**.

In the illustrated embodiment, each of the channels **20**, **22**, and **24** is configured by a PSTN service provider (which, in Canada, may be Bell Canada or Telus, for example) to direct calls that are received on the channels to the gateway **18**. In the illustrated embodiment, the PSTN service provider has configured the channels **20**, **22**, and **24** to communicate with a T1 multiplexer **25**, which multiplexes the channels **20**, **22**, and **24** in a manner known in the art onto one or more T1 lines **27** that are in communication with the gateway **18**. The gateway **18** is in communication with an IP network shown generally at **26**. The channels **20**, **22**, and **24** are thus configured to cooperate with the IP network **26** (via the gateway **18** in the illustrated embodiment) to cause a call involving the mobile telephone **12** and the callee to be routed through the IP network in response to a call received at one of the channels.

Also, in the illustrated embodiment, the access server **14** is in communication with the routing controller **30** of the first node **11**, and the routing controller **30** is configurable to associate a callee identifier with one of the channels **20**, **22**, and **24**, as described below. A callee identifier associated with one of the channels **20**, **22**, and **24** may be a telephone number of a PSTN telephone **32** that is in communication with the IP network **26** through a gateway **34**, or it may be a telephone number of a VoIP telephone **36** that is directly in communication with the IP network **26**, for example. Other routing controllers **30** of other nodes, such as the second node **21**, for example, may also

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associate callee identifiers with other channels that are in communication with other gateways (not shown).

Mobile Telephone

Referring to Figure **2**, in this embodiment, the mobile telephone (**12**) includes a processor circuit shown generally at **50**. The processor circuit **50** includes a microprocessor **52**, a program memory **54**, an input/output (I/O) port **56**, parameter memory **58**, and temporary memory **60**. The program memory **54**, I/O port **56**, parameter memory **58**, and temporary memory **60** are all in communication with the microprocessor **52**. The processor circuit **50** may alternatively include a plurality of processors, a plurality of program memories, a plurality of temporary memories, and/or a plurality of I/O ports, or these components may alternatively be combined into a single device. However, for simplicity, the components of the processor circuit **50** are illustrated as shown in the example of Figure **2**.

In the illustrated embodiment, the I/O port **56** includes a dialing input **62** for receiving a callee identifier from a key pad, for example, or from a voice recognition unit, or from pre-stored callee identifiers stored in the parameter memory **58**, for example. For illustration purposes only, a myriad of possible dialing functions for providing a callee identifier are represented by the block entitled dialing function **64**. A callee identifier may be a telephone number of a callee, for example.

The I/O port **56** also includes a handset interface **66** for receiving and producing signals to and from a handset **68** that may be placed close to the user's ear and mouth, for producing and receiving audible signals for and from the user. It will be appreciated that alternatively, the handset **68** may include a camera and video screen, for example, and that video or other types of signals may be transmitted additionally or alternatively to audible signals.

The I/O port **56** also includes a non-voice network interface **70** for transmitting information to, and receiving information from, the non-voice network **16**

illustrated in Figure **1**, for example, and preferably interfaces with a high-speed internet connection.

The I/O port **56** in the illustrated embodiment further includes a mobile telephone network interface **72** for transmitting signals to and receiving signals from a mobile telephone service provider over a network such as a Global System for Mobile communications (GSM) or a Code Division Multiple Access (CDMA) network, such as the mobile telephone network **15** illustrated in Figure **1**, for example. Again, for simplicity, a mobile telephone network interface is illustrated, although it will be appreciated that video signals or other signals may be handled similarly when the mobile telephone (**12**) is facilitating communication of one or more of these types of signals. It will also be appreciated that alternatively, the non-voice network interface **70** and mobile telephone network interface **72** need not be distinct, but may be a single interface for communication over a single network, for example, or may be configured to communicate over a plurality of different networks, for example.

In the illustrated embodiment, the parameter memory **58** includes a username field **74** and a password field **76**, although it will be appreciated that the username and password may not be necessary, or may be input by the user as required, for example. The parameter memory **58** in the illustrated embodiment also includes a caller identifier field **78** for storing a caller identifier, which may be a telephone number associated with the mobile telephone (**12**) for identifying a "channel" such as a telephone line assigned to the mobile telephone that may be used to call back to the mobile telephone, for example. Generally, the contents of the username field **74**, the password field **76**, and the caller identifier field **78** are set once when the user first subscribes to the system.

The usernames referred to herein, such as the username in the username field **74**, preferably include a twelve digit number such as **2001 1050 8667**, for example, wherein the left-most digit is a continent code (such as "**2**" to

indicate North America, for example), followed by a three-digit country code (such as "**001**" to indicate Canada and the United States, for example), a four-digit dealer code (such as "**1050**", for example), and a unique four-digit number code (such as "**8667**", for example), as discussed more generally in PCT Publication No. **2008/052340**. Therefore, a prefix of a username referred to herein preferably indicates a geographical region associated with the user, or with the access code, and more preferably indicates a node associated with the user or access code.

The program memory **54** stores blocks of codes for directing the microprocessor **52** to carry out the functions of the mobile telephone (**12**), which are illustrated by example below.

Referring to Figures **2** and **3**, a flow chart representing functions performed by blocks of code that direct the microprocessor **52** to initiate a call with the mobile telephone **12** to a callee is shown generally at **100**. The blocks shown in Figure **3** generally represent codes that may be stored in the program memory **54** for example, for directing the microprocessor **52** to perform various functions relating to initiating a call with the mobile telephone (**12**) to a callee. The actual code to implement each block may be written in any suitable programming language, such as Java, C, and/or C++, for example.

The process **100** begins at **102**, in response to an interrupt produced at or for the microprocessor **52** by the dialing function **64**. Upon initiation of the process **100**, block **104** directs the microprocessor **52** to obtain a callee identifier from the dialing function **64** at the dialing input **62** of the I/O port **56** in the illustrated embodiment. The callee identifier is associated with a desired callee, and may be a telephone number of the callee, for example. The microprocessor **52** thus receives, from a user of the mobile telephone (**12**), a callee identifier associated with a callee.

Block **106** directs the microprocessor **52** to transmit, using the non-voice network interface **70** in the illustrated embodiment, an access code request

message, the access code request message including the callee identifier obtained at block **104**, to the access server **14** (illustrated in Figure **1**). In general, preferably block **106** directs the microprocessor **52** to cause an access code request message to be transmitted to the access server **14** over a non-voice network, such as an internet, using WiFi or GPRS technology for example. However, it will be appreciated that block **106** may direct the microprocessor **52** to transmit an access code request message to the access server **14** using any suitable technique, which may alternatively include a voice network, for example.

Referring to Figure **4**, an exemplary access code request message is shown generally at **110**. The access code request message **110** includes a username field **112**, a password field **114**, a callee identifier field **116**, and a caller identifier field **118**. In the illustrated embodiment, values for the username, password, and caller identifier fields **112**, **114**, and **118** are retrieved from the username, password, and caller identifier fields **74**, **76**, and **78** respectively in the parameter memory **58** of the processor circuit **50** (illustrated in Figure **2**), and a value for the callee identifier field **116** is obtained from the dialing function **64** in block **104**, and may be stored in the temporary memory (**60**), for example. It will be appreciated that the username field **112**, password field **114**, and caller identifier field **118** are not essential, although these fields are preferable in order to identify the user of the mobile telephone for billing purposes, for example.

Referring to Figures **1** and **4**, it will be appreciated that in order to minimize charges from the mobile telephone service provider of the mobile telephone **12**, the channels **20**, **22**, **24** will preferably be local or relatively inexpensive telephone lines associated with a geographical location, more particularly a pre-defined local calling area, associated with the mobile telephone **12**. Therefore, the exemplary access code request message **110** further includes a location identifier field **119**. The location identifier stored in the location identifier field **119** preferably identifies a location of the mobile telephone **12**

for use in determining a local calling area associated with the mobile telephone **12**.

For example, the location identifier in the location identifier field **119** may include an IP address of the mobile telephone **12** in a wireless IP network, such as the non-voice network **16** to which the non-voice network interface **70** shown in Figure **2** is connected, because this IP address may be an indicator of a geographical location of the mobile telephone **12**. The location identifier may also or alternatively include an identifier of a wireless voice signal station in wireless communication with the mobile telephone. In the illustrated embodiment, the wireless voice signal station is part of the mobile telephone network **15** that is in communication with the mobile telephone **12** through the mobile telephone network interface **72** illustrated in Figure **2**. In still other embodiments, the location identifier may include a user-configured identifier of a geographical location or local calling area where the mobile telephone **12** is or may be situated. The location identifier may thus be pre-determined and stored in the parameter memory **58** shown in Figure **2** or may be acquired from non-voice network or wireless voice signal station or from user input, for example. Therefore, in summary, the location identifier in the location identifier field **119** may include one or more of an IP address of the mobile telephone **12** in a wireless IP network, an identifier of a wireless voice signal station in wireless communication with the mobile telephone, and a user-configured identifier.

As described below, the location identifier in the location identifier field **119** may be used to determine a local calling area associated with the mobile telephone **12**, within which local calling area channels (illustrated as **20**, **22**, and **24** in Figure **1**) are available to the mobile telephone **12** for the lowest cost to the user. However, it will be appreciated that the location identifier may only approximately identify a local calling area, and may not necessarily identify the lowest cost channel (illustrated as **20**, **22**, and **24** in Figure **1**) for the mobile telephone **12**. It will also be appreciated that in other embodiments, the location identifier field **119** may be omitted.

Referring back to Figure **3**, the process **100** continues at block **130**, which directs the microprocessor (**52**) to receive an access code reply message from the access server (**14**) in response to the access code request message that was transmitted at block **106**.

Referring to Figure **5**, an exemplary access code reply message is shown generally at **140**. The access code reply message **140** includes an access code field **142** and a timeout field **144**. In the illustrated embodiment, the access code field **142** stores an access code which is a telephone number associated with a telephone line associated with one of the channels **20**, **22**, or **24** in Figure **1**. It will be appreciated that the access code is different from the callee identifier in the callee identifier field **116** shown in Figure **4**, in that the access code identifies a channel, other than that provided by the callee identifier provided by the dialing function **64** in Figure **2**, that the mobile telephone (**12**) can use to initiate a call to the callee. It will be appreciated that use of the access code facilitates avoidance of long distance or roaming charges that a mobile telephone service provider would charge for a call placed directly using the callee identifier using conventional calling processes, for example.

Still referring to Figure **5**, the timeout field **144** in the illustrated embodiment stores a value that indicates a period of time, for example a number of minutes, during which the access code in the access code field **142** is associated with the callee identifier in the callee identifier field **116** of the exemplary access code request message **110** illustrated in Figure **4**, such that the access code is only temporarily associated with the callee identifier. In one embodiment, the value stored in the timeout field **144** indicates **10** minutes, for example. It will be appreciated that in other embodiments, the timeout field **144** may not be necessary, but preferably it is included.

In the illustrated embodiment, the program codes in block **130** direct the microprocessor **52** to receive the access code reply message over a non-

voice network, such as a WiFi or GPRS network (illustrated at **16** in Figure **1**) via the non-voice network interface **70** shown in Figure **2**. However, it will be appreciated that the access code reply message may be received on any suitable network, even a voice network, for example.

Referring back to Figures **2** and **3**, block **149** directs the microprocessor **52** to initiate a call with the mobile telephone (**12**) on the mobile telephone network **15** (illustrated in Figure **1**) using the access code received in the access code field **142** of the access code reply message **140** (shown in Figure **5**) to identify the callee. In the illustrated embodiment, the codes in block **149** direct the microprocessor **52** to initiate a call to the channel (**20**, **22,** or **24**) identified by the access code, using the mobile telephone network interface **72** of the I/O port **56** of the mobile telephone (**12**), to engage the mobile telephone network (**15**).

Referring to Figure **1**, in the embodiment shown, the access code in the access code field (**142**) is a telephone number identifying a channel **20**, **22**, or **24** that is in communication with the gateway **18** to the IP network **26**. Through the gateway **18**, the channel **20**, **22**, or **24** is thus operably configured to cooperate with the IP network **26** to cause a call from the mobile telephone **12** to the callee to be routed through the IP network. Routing the call through the IP network may involve engaging the routing controller **30** to route the call on the IP network **26** to the callee, as described below. However, it will be appreciated that in other embodiments, the access code need not be a telephone number, but may be any code identifying a channel through which the mobile telephone **12** can initiate a call. Alternatively, if the mobile telephone is capable of voice over IP communications, the access code may be used to identify an IP address in the IP network to which the call is routed. In this embodiment, the IP address may act as the access code. The process **100** shown in Figure **3** is then ended.

Access Server

Referring to Figure **6**, the access server (**14**) includes a processor circuit shown generally at **150**. The processor circuit **150** includes a microprocessor **152**, program memory **154**, an I/O port **156**, parameter memory **158**, and temporary memory **160**. The program memory **154**, I/O port **156**, parameter memory **158**, and temporary memory **160** are all in communication with the microprocessor **152**. The processor circuit **150** may alternatively include a plurality of microprocessors or I/O ports, for example, and the components of the illustrated processor circuit **150** may also alternatively be combined into a single device.

The program memory **154** stores blocks of codes for directing the microprocessor **152** to carry out the functions of the access server **14**. The I/O port **156** includes a non-voice network interface **162** for communicating with the non-voice network **16** illustrated in Figure **1**. The I/O port **156** also includes a routing controller interface **164** for interfacing with the routing controller **30** illustrated in Figure **1**.

Referring to Figures **6** and **7**, a flow chart of blocks of code for directing the microprocessor **152** of the access server (**14**) to provide an access code to the mobile telephone (**12**) is shown generally at **190**. The blocks **190** in Figure **7** generally represent codes that may be stored in the program memory **154** for directing the microprocessor **152** to perform various functions to provide the access to the mobile telephone (**12**) to enable the mobile telephone to place a call through a channel (**20**, **22**, or **24**).

The process **190** begins at **192**, in response to an interrupt created by or for the microprocessor **152** when it receives an access code request message **110** (as illustrated in Figure **4**) from the mobile telephone (**12**). In the illustrated embodiment, the access code request message (**110**) is received via the non-voice network interface **162** through a non-voice network (**16**) such as a WiFi or GPRS network, for example. However, it will be appreciated that other embodiments may use different techniques for receiving the access code request message (**110**) from the mobile telephone (**12**).

The process **190** continues at block **196**, which directs the microprocessor **152** to communicate with the routing controller **30** to obtain from the routing controller an access code identifying a channel (illustrated as **20**, **22**, or **24** in Figure **1**) in communication with the gateway (**18**), wherein the access code is different from the callee identifier in the callee identifier field **116** (shown in Figure **4**) and is usable by the mobile telephone (**12**) to initiate a call to the callee using the channel, as further described below. Therefore, block **196** preferably causes an access code to be produced by retransmitting the access code request message **110** illustrated in Figure **4** that was received at **192** from the mobile telephone (**12**), to the routing controller **30** through the routing controller interface **164** of the I/O port **156**.

The process **190** continues at block **198**, which directs the microprocessor **152** to transmit an access code reply message (**140**), including the access code obtained by block **196**, to the mobile telephone (**12**). An exemplary access code reply message is shown in Figure **5**. In the illustrated embodiment, an access code reply message (**140**) is produced by the routing controller **30** in a manner described below in response to the access code request message (**110**) that was transmitted to the routing controller at block **196**, and the access code reply message (**140**) is received from the routing controller through the routing controller interface **164** of the I/O port **156**. Block **198** then causes the access code reply message that was received from the routing controller to be retransmitted to the mobile telephone (**12**). In the illustrated embodiment, the codes in block **198** direct the microprocessor **152** to transmit the access code reply message (**140**) using the non-voice network interface **162** to the non-voice network **16**, which may be a WiFi or GPRS network, for example. However, it will be appreciated that other embodiments may employ other types of networks for communicating the access code reply message (**140**) to the mobile telephone (**12**). The process **190** is then ended.

In summary, referring to Figure **1**, the access server **14** generally acts as an interface to the routing controller **30** for relaying access code request

messages and access code reply messages between the mobile telephone **12** and the routing controller. Therefore, it will be appreciated that in alternative embodiments, the access server **14** and the routing controller **30** need not be separate, but may, for example, be combined in a single component.

Routing Controller (RC)

Referring to Figure **1**, generally, the routing controller **30** executes a process to facilitate communication between callers and callees. The function of a routing controller generally in a VoIP system is described in PCT Publication No. WO **2008/052340**.

Referring to Figure **8**, the routing controller (**30**) includes a processor circuit shown generally at **230**. The processor circuit **230** includes a microprocessor (or more generally a processor) **232**, program memory **234**, an I/O port **236**, table memory **238**, temporary memory **240**, and a clock **244**. The program memory **234**, I/O port **236**, table memory **238**, temporary memory **240**, and clock **244** are all in communication with the processor **232**. The processor circuit **230** may include a plurality of microprocessors, for example, and the aforementioned components of the processor circuit **230** may be combined, for example. The program memory **234** includes blocks of code for directing the processor **232** to carry out the functions of the routing controller (**30**), and the I/O port **236** includes an access server interface **242** for communicating with the access server **14**.

In the illustrated embodiment as described above, the access server (**14**) transmits (at block **196** illustrated in Figure **7**) an access code request message (**110**) to the routing controller (**30**) in order to obtain from the routing controller (**30**) an access code. When an access code request message (**110**) is received at the access server interface **242**, the processor **232** preferably stores certain values from the access code request message in stores in the temporary memory **240** for ease of retrieval. In particular, the temporary memory **240** includes a callee identifier store **246** for storing the callee

identifier from the callee identifier field **116** in the access code request message **110** illustrated in Figure **4**, a caller identifier store **248** for storing the caller identifier that was stored in the caller identifier field **118** of the access code request message **110** illustrated in Figure **4**, a caller username store **249** for storing the caller username that was stored in the caller username field **112** of the access code request message **110** illustrated in Figure **4**, and an access code store **250** for storing an access code that is selected when the routing controller (**30**) receives an access code request message (**110**). The temporary memory **240** also includes a local calling area identifier store **245** for storing an identifier of a local calling area associated with the mobile telephone (**12**). The clock **244** generally maintains and stores a representation of a current date and time.

The I/O port **236** further includes a database request port **256** through which a request to the database (**23** in Figure **1**) can be made, and also includes a database response port **258** for receiving a reply from the database (**23**). The I/O port **236** further includes a routing controller (RC) request message input **260** for receiving an RC request message (illustrated in Figure **17**) from the call controller (**13** in Figure **1**) and includes a routing message output **262** for sending a routing message back to the call controller **13**. The I/O port **236** thus acts to receive a caller identifier and a callee identifier contained in an RC request message from the call controller, the RC request message being received in response to initiation of a call by a subscriber of the system, as described below.

The program memory **234** includes blocks of codes for directing the processor **232** to carry out various functions of the routing controller (**30**). One of these blocks includes an RC request message handler **380** which directs the routing controller (**30**) to produce a routing message in response to a received RC request message, an example of which is illustrated in Figure **17**. The RC request message handler process is shown in greater detail at **380** in Figures **18**A through **18**C. Another of these blocks in the program memory **234** includes an access code generator, which is described at **270** in Figure **12**,

and which directs the routing controller (**30**) to produce an access code as directed by the program codes in block **196** shown in Figure **7**. Yet another of these blocks in the program memory **234** includes a local calling area identifier generator, which directs the routing controller (**30**) to produce a local calling area identifier using the location identifier from the location identifier field **119** of the access code request message **110** illustrated in Figure **4**.

Local Calling Area Identifier Generator

Referring to Figure **1**, it will be appreciated that preferably, a call made by the mobile telephone **12** using the access code obtained from the access server **14** will be a local call for the mobile telephone **12**, based on a geographical location of the mobile telephone. Therefore, blocks in the program memory **234** include a local calling area identifier generator, which directs the routing controller **30** to produce a local calling area identifier.

For example, the local calling area identifier generator may direct the microprocessor **152** to access a dialing profile associated with the caller. The dialing profile may be identified using the username in the username field **112** in the access code request message **110** illustrated in Figure **4**, and to store in the local calling area identifier field **245** a default location of the caller retrieved from the dialing profile associated with the caller.

Referring to Figure **9**, an exemplary dialing profile is illustrated generally at **200** and includes a username field **202**, a domain field **204**, and calling attributes comprising a national dialing digits (NDD) field **206**, an international dialing digits (IDD) field **208**, a country code field **210**, a local area codes field **212**, a caller minimum local number length field **214**, a caller maximum local number length field **216**, a reseller field **218**, a maximum number of concurrent calls field **220**, a current number of concurrent calls field **222**, and a default local calling area identifier field **224**. Therefore, in some embodiments, the local calling area identifier generator directs the

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microprocessor **152** to determine a local calling area associated with the mobile telephone (**12**) by retrieving the default local calling area identifier from the default local calling area identifier field **224** of the dialing profile **200**.

Effectively, the dialing profile **200** is a record identifying calling attributes of the caller identified by the username in the username field **202**. More generally, dialing profiles **200** represent calling attributes of respective users, and are discussed in more detail in PCT publication No. WO **2008/052340**. As described in PCT publication No. WO **2008/052340**, a dialing profile of the type shown in Figure **9**, and also other records such as direct-in-dial (DID) records, call blocking records, call forwarding records, and voicemail records, may be created whenever a user registers with the system or agrees to become a subscriber to the system.

Alternatively, the local calling area identifier generator may generate a local calling area identifier to be stored in the local calling area identifier store **245** using the location identifier from the location identifier field **119** of the access code request message **110** illustrated in Figure **4**. As described above, the location identifier field (**119**) may store one or more of an IP address of the mobile telephone (**12**) in a wireless IP network, an identifier of a wireless voice signal station in wireless communication with the mobile telephone, and a user-configured identifier. One or more of these values may be used to identify a local calling area that is or is likely to be associated with the mobile telephone (**12**) in order to generate a local calling area identifier to be stored in the local calling area identifier store **245**.

For example, it has been found that services available from web sites such as http://www.ip**2**location.com/ and http://www.serviceobjects.com/products/dots_ipgeo.asp, for example, can produce a name of a location, and also latitude and longitude values, associated with an IP address. Using this information derived from an IP address, or other information from the location identifier field (**119**), a local calling area may be identified by hierarchical jurisdictional designations (such as country, province, and city in Canada or

country, state, and city in the United States) and encoded as codes identifying the local calling area. These codes may then be stored in the local calling area identifier store **245**.

Access Code Association Table

In the illustrated embodiment, the table memory **238** (shown in Figure **8**) includes an access code association table **170**, an example of which is illustrated in Figure **10**, for associating access codes with callee identifiers, caller identifiers, caller usernames, timeouts, and timestamps. Although the routing controller (**30**) is illustrated in this embodiment as a separate component from the access server (**14**), it will be appreciated that in other embodiments, the routing controller (**30**) may be part of or integrated with the access server (**14**), and in these other embodiments, the access code association table **170** may be part of or integrated with the access server.

Referring to Figures **1** and **10**, the access code association table **170** generally includes a plurality of records, each having an access code field **173** storing an access code. The access codes in the access code association table **170** may thus form a pool of access codes, where each access code may identify a respective telephone number. In the illustrated embodiment, the access codes in the access code fields **173** of records of the access code association table **170** identify respective channels (illustrated by example only as **20**, **22**, and **24**) that are operably configured to cooperate with the IP network **26** via the gateway **18** to cause a call involving the mobile telephone **12** to be routed through the IP network.

Referring to Figure **10**, the exemplary access code association table **170** includes records **172**, **174**, **176**, **178**, and **180**, each having respective fields for storing a local calling area identifier **171**, an access code **173**, a channel identifier **175**, a callee identifier **177**, a caller identifier **179**, a caller username **183**, a timeout **181**, and a timestamp **182**. Generally, a record in the access code association table **170** will be created for each access code that identifies a channel (such as the channels **20**, **22**, and **24** illustrated in Figure **1**) that is

configured or configurable to establish communication through a gateway (such as the gateway **18** illustrated in Figure **1**) to an IP network (**26** in Figure **1**) in response to a call received at the channel. When a record is created in respect of a channel, the local calling area identifier field **171** is preferably initialized with an identifier of a local calling area associated with the channel, the access code field **173** is preferably initialized with an access code associated with the channel, and the channel identifier field **175** is preferably initialized with an identifier of the channel. The remaining fields (for storing a callee identifier **177**, a caller identifier **179**, a caller username **183**, a timeout **181**, and a timestamp **182**) are preferably initialized with default "null" values when a record is created. The fields for storing a local calling area identifier **171**, an access code **173**, a channel identifier **175** preferably remain generally constant during ordinary operation of the access code association table **170**, although the values stored in the fields for storing a callee identifier **177**, a caller identifier **179**, a caller username **183**, a timeout **181**, and a timestamp **182** may vary as described below. It will be appreciated that in some embodiments, one or more of the fields for storing a local calling area identifier **171**, a channel identifier **175**, a caller identifier **179**, a caller username **183**, a timeout **181**, and a timestamp **182** may not be required and be omitted.

As noted above, the local calling area identifier field **171** is preferably initialized with an identifier of a local calling area associated with the channel. The local calling area identifier field **171** preferably stores codes that are encoded in the same manner as the codes in the local calling area identifier store **245**, as described above, so that an access code in the local calling area identified by the codes in the local calling area identifier store **245** may be identified by searching the access code association table **170** for an access code associated with a local calling area identifier in the associated local calling area identifier field **171** that matches the local calling area identifier in the local calling area identifier store **245**. It has been found that information available from web sites such as http://en.wikipedia.org/wiki/List_of_NANP_area_codes, and services available from web sites such as

http://www.serviceobjects.com/demos/PhoneExchangeDemo.asp, for example, may be used to determine a local calling area identifier associated with a given access code where, for example, the access code is a PSTN telephone number.

In the exemplary access code association table **170**, the access codes in the access code fields **173** are telephone numbers for PSTN lines, three of which are in the **604** area code in Vancouver, British Columbia, Canada, and two of which are in the **416** area code in Toronto, Ontario, Canada. It will be appreciated that the access code association table **170** is an example only, and other access code association tables may include any number of access codes, which need not be PSTN telephone numbers, and which need not be limited to particular geographical areas.

In the exemplary access code association table **170**, the access code field **173** in the record **174** stores an access code **1-604-345-2323**, which may be a local telephone number for Vancouver, British Columbia, Canada, and the callee identifier field **177** of the record **174** stores a callee identifier **1-403-789-1234**, which may be a telephone number for a callee in Calgary, Alberta, Canada for example, thereby associating the callee identifier **1-403-789-1234** with the access code **1-604-345-2323**. Furthermore, the caller identifier field **179** of the record **174** stores a caller identifier **1-416-444-1441** and the caller username field **183** stores a caller username **2001 1050 8667**, thereby associating the caller identifier **1-416-444-1441** and caller username **2001 1050 8667** with the aforementioned access code and callee identifier. The caller identifier **1-416-444-1441** may be associated with a mobile telephone normally geographically located in Toronto, Ontario, Canada, but which may be in Vancouver and is therefore using a Vancouver-based access code to place a call to a Calgary-based number, for example. In the example record **174**, the timestamp field **182** indicates that the callee identifier **1-403-789-1234**, the caller identifier **1-416-444-1441**, and the caller username **2001 1050 8667** were associated with the access code **1-604-345-2323** on June **15**,

2008 at **10:31** am, and the timeout field **181** indicates that this association is to expire **10** minutes after the time indicated in the timestamp field.

Likewise, the exemplary record **178** indicates that the callee identifier **1-604-321-1234**, the caller identifier **1-416-444-1234**, and the caller username **2001 1050 4141** were associated with the access code **1-416-234-4646** on June **15**, **2008** at **2:21** pm, and the timeout field **181** of the record **178** indicates that this association is to expire within **10** minutes of the time in the timestamp field **182**.

It will also be appreciated that the access code association table **170** may, in other embodiments, be substituted with other data structures or storage media. For example, in alternative embodiments, as described below, a DID record of the type shown at **370** in Figure **11** may associate an access code with a callee identifier and with other information such as a caller identifier, a timeout value, and a timestamp value, additionally or alternatively to the access code association table **170**.

DID Bank Table Records

As described in PCT Publication No. **2008/052340**, a DID bank table record may be created and stored in a DID bank table in the database (**23** in Figure **1**) when a user registers with the system, to associate the username of the user and a host name of the node with which the user is associated, with a number on the PSTN network formatted in compliance with the E.**164** standard set by the International Telecommunication Union (ITU). However, as explained below, DID records may, in some embodiments, also associate usernames and host names with respective access codes, and may also associate access codes with respective callee identifiers and with other information such as caller identifiers, timeout values, and timestamp values.

Referring to Figure **11**, an exemplary DID bank table record is shown generally at **370**, and includes a username field **371**, a user domain field **372**, and a DID field **373**. The username field **371** may store a username of a user

of the system, in which case the user domain field **372** stores a host name of the node with which the user is associated, and the DID field **373** stores an E.**164** number on the PSTN network associated with the user. Exemplary host names stored in the user domain field **372** include sp.yvr.digifonica.com for Vancouver, British Columbia, Canada and sp.lhr.digifonica.com for London England, for example, as described in PCT Publication No. **2008/052340**. If the user has multiple telephone numbers, then multiple records of the type shown at **370** would be included in the DID bank table, each having the same username and user domain, but different DID field **373** contents reflecting the different telephone numbers associated with that user.

However, DID fields **373** of DID bank table records **370** may also store access codes, in which case the username field **371** may store a username associated with the access code. In these DID bank table records **370**, the user domain field **372** stores a host name of the node with which the access code is associated. Therefore, DID bank table records **370** may, in some embodiments, associate usernames and host names with respective access codes.

The exemplary DID bank table record **370** further includes a callee identifier field **374**, a caller identifier field **375**, a timeout field **376**, a timestamp field **377**, a local calling area identifier field **378**, a channel identifier field **379**, and a caller username field **381**, which may be used in an analogous manner to the callee identifier field **177**, the caller identifier field **179**, the timeout field **181**, the timestamp field **182**, the local calling area identifier field **171**, the channel identifier field **175**, and the caller username field **183** respectively of the access code association table **170** illustrated in Figure **10**. The DID bank table records **370** may thus associate access codes with respective local calling area identifiers, callee identifiers, caller identifiers, caller usernames, timeouts, and timestamps, although the caller identifier field **375**, timeout field **376**, timestamp field **377**, local calling area identifier field **378**, channel identifier field **379**, and caller username field **381** may not be necessary, and one or more of these fields may be omitted in some embodiments.

Furthermore, it will be appreciated that the callee identifier field **374**, caller identifier field **375**, timeout field **376**, and timestamp field **377** of the DID bank table record **370** may be omitted for DID table records that are not in respect of access codes, but rather are in respect of telephone numbers of users of the system, for example, as described in PCT Publication No. **2008/052340**. The callee identifier field **374**, caller identifier field **375**, timeout field **376**, and timestamp field **377** of the DID bank table record **370** may also be omitted in embodiments where the access code association table **170** includes records with these types of fields.

For simplicity, the following description is directed to embodiments wherein an access code association table **170** associates access codes with respective callee identifiers, caller identifiers, timeout values, and timestamp values. However, it will be appreciated that the processes described herein for records in the access code association table **170** may additionally or alternatively be applied to DID bank table records **370** in an analogous manner.

Access Code Generator

Referring back to Figures **1**, **4**, and **8** in the illustrated embodiment as described above, the access server **14** transmits (at block **196** illustrated in Figure **7**) an access code request message **110** to the routing controller **30** in order to obtain from the routing controller **30** an access code. When an access code request message **110** is received at the access server interface **242**, the processor **232** preferably authenticates the user by making various enquiries of databases to which it has access, to determine whether or not the password in the password field **114** of the access code request message **110** matches a password stored in the database in association with the username in the username field **112**. Various functions may be used to pass encryption keys or hash codes back and forth to ensure that the transmittal of passwords is secure. If the user is successfully authenticated, the processor **232** then preferably produces an access code.

Referring to Figures **8** and **12**, a process for producing an access code is shown generally at **270**. Essentially the process **270** determines whether the access code in a given record (referred to below as the "currently addressed record") in the access code association table shown at **170** in Figure **10** is within the local calling area identified by the local calling area identifier store **245**, and whether the access code is currently available for association with a callee identifier. In order to produce an access code in response to receiving an access code request message (**110**) from the access server (**14**), the processor **232** of the routing controller (**30**) preferably searches the pool of access codes in the access code association table (**170**) to identify an access code identifying a channel usable by the mobile telephone (**12**) to initiate a call to the callee, using the process **270** until an available access code in the local calling area identified by the local calling area identifier store **245** is identified. The access code generator thus preferably selects an access code from the pool of access codes in the access code association table (**170**), and preferably selects an access code in a local calling area associated with the mobile telephone (**12**).

Starting with the first record in the access code association table, the process **270** begins at block **272**, which directs the processor **232** of the routing controller (**30**) to determine whether the access code in the currently addressed record of the access code association table **170** is associated with the same local calling area as the mobile telephone (**12**) as identified by the contents of the local calling area identifier store **245**. If at block **272** the access code of the currently addressed record is not associated with the same local calling area as the mobile telephone (**12**), the process **270** ends, the next record in the access code association table **170** is addressed, and the process is repeated for the next record in the access code association table.

However, if at block **272** the access code of the currently addressed record is associated with the same local calling area as the mobile telephone (**12**), or if the access code request message **110** (illustrated in Figure **4**) did not include

a local calling area identifier, then the process **270** continues at block **274**, which directs the processor **232** to determine whether the access code of the currently addressed record is associated with a callee identifier. To do this, the processor **232** determines whether the callee identifier field (**177**) of the currently addressed record stores a "null" value that was assigned to it on initialization, or whether the callee identifier field instead stores a callee identifier. In other words the processor checks to see whether the currently addressed record has already been in use.

If at block **274** the callee identifier field (**177**) of the currently addressed record in the access code association table (**170**) does store a callee identifier and not the "null" value that was assigned to the callee identifier field on initialization (for example, records **174** and **178** in Figure **10**), then the access code of that record is associated with a callee identifier, and the process **270** continues at block **278**, which directs the processor **232** to determine whether the association of the callee identifier with the access code has expired. In the illustrated embodiment, the codes at block **278** direct the processor **232** to determine whether the sum of the contents of the timestamp field (**182**) and of the timeout field (**181**) in the currently addressed record of the access code association table **170** (shown in Figure **10**) is less than the current time represented by the clock **244**. If at block **278** the sum of the timeout and timestamp fields in the currently addressed record of the access code association table **170** is less than the time represented by the clock **244**, then the association of the callee identifier with the access code is not expired and the process **270** ends, the next record in the access code association table (**170**) is addressed, and the process **270** is repeated for the next record in the access code association table.

However, if at block **278** the sum of the contents of the timeout and timestamp fields (**181** and **182**) in the currently addressed record of the access code association table (**170**) is not less than the time represented by the clock **244**, then the association of the callee identifier with the access code has expired, and the process **270** continues at block **276** which directs the processor **232**

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to store the contents of the access code field **173** of the currently addressed record in the access code store **250** of the temporary memory **240** of the routing controller **30**.

Referring to Figures **8, 10**, and **12**, if at block **274** the callee identifier field in the currently addressed record does not store a callee identifier but stores instead the "null" value that was assigned to the callee identifier field on initialization (for example, records **172**, **176**, and **180**), then the access code of that record is not associated with a callee identifier, and the process **270** continues at block **276**, which directs the processor **232** to store the access code from the access code field **173** of the currently addressed record, in the access code store **250** in the temporary memory **240**.

After the selected access code is stored in the access code store **250** at block **276**, the process **270** continues at block **280**, which directs the processor **232** to store the callee identifier from the callee identifier store **246** in the callee identifier field **177** of the currently addressed record, thereby creating an association of the callee identifier with the selected access code.

The process **270** then continues at block **282**, which directs the processor **232** to store the caller identifier from the caller identifier store **248** (which identifies the mobile telephone **12** shown in Figure **1**) in the caller identifier field **179** of the currently addressed record of the access code association table **170**, thereby also storing the caller identifier in association with the selected access code.

The process **270** then continues at block **283**, which directs the processor **232** to store the caller username from the caller username store **249** in the caller username field **183** of the currently addressed record of the access code association table **170**, thereby also storing the caller username in association with the selected access code.

The process **270** then continues at block **284**, which directs the processor **232** to store timeout and timestamp values in the timeout and timestamp fields **181** and **182** of the currently addressed record of the access code association table **170**, thus further storing, in association with the selected access code, a timestamp for use in determining when the usability of the access code to initiate a call to the callee will expire. A default value, such as **10** minutes, for example may be stored in the timeout field **181** of the currently addressed record. Also, the current time indicated by the clock **244** is preferably stored in the timestamp field **182** of the currently addressed record.

In alternative embodiments, the access code association table (**170**) might not include fields for a caller identifier, caller username, a timeout, or a timestamp. In these embodiments, one or more of blocks **282**, **283**, and **284** described above are not necessary, and one or more of the caller identifier store **248** and the caller username store **249** may be omitted.

In summary, the access code generator in the illustrated embodiment responds to receiving an access code request message **110** illustrated in Figure **4** from the access server (**14**) by first authenticating the user, and then by searching through a pool of access codes, using the process **270** shown in Figure **12**, to identify an access code that is associated with the local calling area identified by the local calling area identifier store (**245**) and that is not previously and validly associated with another callee identifier. It will be appreciated that in alternative embodiments, different data structures and algorithms may be preferable for identifying an access code that meets the aforementioned criteria. For example, in accordance with conventional database design that is well-known in the art, the records illustrated in the access code association table **170** illustrated in Figure **10** may alternatively be organized in a binary tree according to the value in the local calling area identifier field **171**, or in separate tables for respective local calling area identifiers, for example, in order to enable a more efficient search of the access code association table for an access code that satisfies the aforementioned criteria. Therefore, the access code association table (**170**)

and the process **270** illustrated in Figure **12** are examples only, and one of ordinary skill in the art will readily appreciate numerous alternative data structures and algorithms.

Gateway

Referring to Figure **13**, in this embodiment, the gateway (**18**) includes a processor circuit shown generally at **300**, which includes a microprocessor **302**. The processor circuit **300** also includes a program memory **304**, a memory **305**, and an I/O port **306**, all of which are in communication with the microprocessor **302**. The processor circuit **300** may include multiple processors etc., and the aforementioned components of the processor circuit **300** may alternatively be combined.

The I/O port **306** includes a channel interface **308**, which, in the illustrated embodiment, is in communication with the channels **20**, **22**, and **24** that were also illustrated in Figure **1**. Where, as in the illustrated embodiment, the channels **20**, **22**, and **24** are PSTN telephone lines in the PSTN network **29**, the channel interface **308** may, for example, be a T**1** port for communication with one or more T**1** lines (illustrated at **27** in Figure **1**) of a PSTN service provider, in a manner well-known in the art. The I/O port in the illustrated embodiment also includes an internet interface **309** for interfacing with the IP network **26** illustrated in Figure **1**. The program memory **304** stores blocks of codes for directing the microprocessor **302** to carry out the functions of the gateway (**18**). It has been found that the AS**5350** Universal Gateway available from Cisco Systems, Inc. of San Jose, California may, for example, be suitable as the gateway (**18**).

Referring back to Figure **1**, and also still to Figure **13**, when a call is received on one of the channels **20**, **22**, or **24**, the microprocessor **302** causes the I/O port **306** to use the internet interface **309** to send a Session Initiation Protocol (SIP) Invite message to a pre-determined node with which the gateway **18** is associated, which in the illustrated embodiment is the first node **11**. Generally, the gateway **18** will be associated with a node that is geographically closest to

the gateway, in order to minimize transmission times over the IP network **26**. In response to the SIP Invite message, the call controller **13** sends an RC request message to the routing controller **30** which makes various enquiries of the database **23** to produce a routing message that is sent back to the call controller **13**. The call controller **13** then communicates with the media relay **28** to cause a communications link including an audio path (and a videopath if a videophone call) to be established through the media relay to the same node, a different node, or to a communications supplier gateway as shown generally at **34** to carry audio, and where applicable, video traffic to the call recipient or callee.

Referring to Figure **14**, an exemplary SIP Invite message is shown generally at **310** and includes a caller identifier field **312**, a callee identifier field **314**, a digest parameter field **315**, a call identifier field **316**, an IP address field **317**, and a gateway UDP port field **318**. Examples of values for the fields in the SIP Invite message **310** are shown for illustration purposes only in Figure **14**. The caller identifier in the caller identifier field **312** is preferably in the form of the telephone number of the caller followed by the "@" symbol, which in turn is followed by the IP address of the gateway (**18**) in the IP network (**26**). The caller identifier may be determined by retrieving calling line identification (CLID) information from the signal provided by the PSTN network (**29**) to the gateway (**18**) for example. Where the caller identification information is not available to the gateway (**18**), the caller identifier in the caller identifier field **312** preferably includes a pre-assigned number (such as **11111**, for example) indicating that the caller identification information was not available, followed by the "@" symbol and then by the IP address of the gateway (**18**).

The callee identifier in the callee identifier field **314** is the access code identifying the channel (**20**, **22**, or **24** in the example of Figure **1**) on which the call was placed, and which was received from the access server (**14**). In the illustrated example, the access code is the PSTN telephone number **1-604-345-1212** corresponding to the channel **20** illustrated in Figure **1**, and to the

access code stored in the access code field **173** of the record **172** in the exemplary access code association table **170** illustrated in Figure **10**.

The digest parameter in the digest parameter field **315** is generated by the gateway (**18**) and may uniquely identify the SIP session that is initiated with the SIP Invite message **310**.

The call identifier in the call identifier field **316** is, in the illustrated embodiment, a four-digit hexadecimal number generated by the gateway (**18**) to identify the call, followed by the "@" symbol, which in turn is followed by the IP address of the gateway.

The IP address in the IP address field **317** is the IP address of the gateway (**18**) in the IP network (**26**), and the gateway UDP port number in the gateway UDP port field **318** includes a UDP port identifier identifying a UDP port at which the audio/video path will be terminated at the gateway (**18**).

It should be noted that throughout the description of the embodiments of this invention, the IP/UDP addresses of all elements such as the gateway (**18**) will be assumed to be valid IP/UDP addresses directly accessible via the Internet or a private IP network, for example, depending on the specific implementation of the system. As such, it will be assumed, for example, that the gateway (**18**) will have an IP/UDP address directly accessible by the call controllers and the media relays on their respective nodes, and those addresses will not be obscured by Network Address Translation (NAT) or similar mechanisms. In other words, the IP/UDP information contained in SIP messages (for example the SIP Invite message or the RC Request message which will be described below) will match the IP/UDP addresses of the IP packets carrying these SIP messages.

It will be appreciated that in many situations, the IP addresses assigned to various elements of the system may be in a private IP address space, and thus not directly accessible from other elements. Furthermore, it will also be

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appreciated that NAT is commonly used to share a "public" IP address between multiple devices, for example between home PCs and IP telephones sharing a single Internet connection. For example, the gateway (**18**) may be assigned an IP address such as **192.168.0.5**. This address is located in so called "non-routable" (IP) address space and cannot be accessed directly from the Internet. In order for this device to communicate with other computers located on the Internet, the IP address has to be converted into a "public" IP address, for example **24.14.102.5** assigned by the Internet Service Provider, by a device performing NAT, typically a router. In addition to translating the IP address, NAT typically also translates UDP port numbers, for example an audio path originating at the gateway (**18**) and using a UDP port **12378** at its private IP address, may have be translated to a UDP port **23465** associated with the public IP address of the NAT device. In other words, when a packet originating from the gateway (**18**) arrives at an Internet-based node, the source IP/UDP address contained in the IP packet header will be **24.14.102.5:23465**, whereas the source IP/UDP address information contained in the SIP message inside this IP packet will be **192.168.0.5:12378**. The mismatch in the IP/UDP addresses may cause a problem for SIP-based VoIP systems because, for example, a node will attempt to send messages to a private address but the messages will never get there.

Call Controller

Referring to Figure **15**, the call controller (**13**) includes a processor circuit shown generally at **320**. The processor circuit **320** includes a microprocessor **322**, program memory **324**, and an I/O port **326**. The program memory **324** and the I/O port **326** are in communication with the microprocessor **322**. The processor circuit **320** may include a plurality of microprocessors, a plurality of program memories, and a plurality of I/O ports to be able to handle a large volume of calls. However, for simplicity, the processor circuit **320** will be described as having only one microprocessor **322**, program memory **324**, and I/O port **326**, it being understood that there may be more.

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Generally, the I/O port **326** includes an input **328** for receiving messages such as the SIP Invite message from the gateway (**18**) or from a VoIP telephone (**36** in Figure **1**, for example). The I/O port **326** also has an RC request message output **330** for transmitting an RC request message to the routing controller **30** of Figure **1**, an RC message input **332** for receiving routing messages from the routing controller **30**, a gateway output **334** for transmitting messages to the gateway **18** and/or **34** shown in Figure **1** to advise the gateway **18** and/or **34** to establish an audio path, for example, and a gateway input **336** for receiving messages from the gateway **18** and/or **34**. The I/O port **326** further includes a SIP output **338** for transmitting messages to the gateway (**18** and/or **34**) or VoIP telephone (**36**, for example) to advise the gateway **18** and/or **34** or IP telephone of the IP addresses of the gateways which will establish the audio/video path. The I/O port **326** further includes a voicemail server input and output **340** and **342** respectively for communicating with the voicemail server **19** shown in Figure **1**.

While certain inputs and outputs have been shown as separate, it will be appreciated that some may be a single IP address and IP port. For example, the messages sent to the routing controller (**30**) and received from the routing controller (**30**) may be transmitted and received on the same single IP port.

The program memory **324** includes blocks of code for directing the microprocessor **322** to carry out various functions of the call controller (**13**). For example, these blocks of code include a first block **344** for causing the processor circuit **320** to execute a SIP Invite to RC Request process to produce an RC Request Message in response to a received SIP Invite message. In addition, there is a Routing Message to Gateway message block **346** which causes the processor circuit **320** of the call controller to produce a gateway query message in response to a received routing message from the routing controller (**30**).

Referring to Figures **15** and **16**, the SIP Invite to RC Request process is shown in more detail at **344**. On receipt of a SIP Invite message of the type

shown in Figure **14**, block **350** directs the processor circuit **320** to produce an RC Request Message. Block **352** then directs the processor circuit **320** to cause the RC Request Message to be sent to the routing controller **30** illustrated in Figure **1**.

Referring to Figure **17**, an exemplary RC request message is shown generally at **360** and includes a caller identifier field **362**, a callee identifier field **364**, a digest parameters field **366**, and a call identifier field **368**. These fields may be populated with the contents of the caller identifier field **312**, callee identifier field **314**, digest parameter field **315**, and call identifier field **316** respectively of the SIP Invite message **310** illustrated in Figure **14**. In other embodiments, the RC request message may further include a type field (not shown) containing a type code to indicate whether the call is from a third party or from a system subscriber. Other variations of an RC request message are explained in PCT Publication No. WO **2008/052340**. A type field (not shown) in the RC request message **360** may be advantageous in embodiments where SIP Invite messages may also be received from an IP telephone that is using VoIP software to make a voice call. However, in the embodiments that are illustrated herein, SIP Invite messages originate from the gateway (**18**), and therefore a type designation is not necessary and may be omitted from the RC request message **360**. In embodiments where a SIP Invite message may be received from an IP telephone, the SIP invite to RC request process shown in Figure **16** may require additional steps, as illustrated in Figure **5** of PCT Publication No. WO **2008/052340**.

RC Request Message Handler

As illustrated in Figure **8**, the program memory **234** includes an RC request message handler **380** which directs the routing controller (**30**) to produce a routing message in response to a received RC request message (**360**). Referring to Figure **18**A, the RC request message handler **380** begins with a first block **382** that directs the RC processor circuit (**230**) to separately store the contents of the callee identifier field **364** and caller identifier field **362** of

the RC request message (**360**) in the callee identifier store **246** and the caller identifier store **248** respectively of Figure **8**.

Block **384** then directs the RC processor circuit (**230**) to use the contents of the caller username store **249** to locate and retrieve from the database (**23**) a dialing profile **200** associated with the caller, as described above and illustrated in Figure **9**, for example. The retrieved dialing profile may then be stored in the temporary memory **240**, for example.

The RC request message handler **380** continues at block **386**, which directs the processor circuit (**230**) of the routing controller to determine whether the contents of the current number of concurrent calls field **222** of the dialing profile **200** shown in Figure **9** are less than the contents of the maximum number of concurrent calls field **220** of the dialing profile for the caller and, if so, block **388** directs the processor circuit to increment the contents of the current number of concurrent calls field **222** and the processor circuit (**230**) is directed to point A in Figure **18**B. If the contents of the current number of concurrent calls field **222** are equal to or greater than the contents of the maximum number of concurrent calls field **220**, then block **390** directs the processor circuit (**230**) to send an error message back to the call controller (**13**) to cause the call controller to notify the caller that the maximum number of concurrent calls has been reached and no further calls can exist concurrently, including the presently requested call.

Assuming that block **386** allows the call to proceed, the RC processor circuit (**230**) is directed to perform certain checks on the callee identifier in the callee identifier field **246** in Figure **8**. These checks are shown in greater detail in Figure **18**B.

Referring to Figure **18**B, the RC processor circuit (**230**) is directed to a first block **392** that causes it to determine whether a digit pattern of the callee identifier includes a pattern that matches the contents of the international dialing digits (IDD) field **208** in the dialing profile **200** (shown in Figure **9**)

associated with the caller. If so, then block **394** directs the RC processor circuit (**230**) to set a call type code identifier variable maintained by the processor to indicate that the call is an international call, and block **396** directs the processor to produce a reformatted callee identifier by reformatting the callee identifier into a predefined digit format. In this embodiment, this is done by removing the pattern of digits matching the IDD field contents (**208**) of the caller dialing profile **200** to effectively shorten the callee identifier. Then, block **398** directs the RC processor circuit (**230**) to determine whether or not the callee identifier has a length which meets criteria establishing it as a number compliant with the E.**164** Standard set by the ITU. If the length does not meet these criteria, then block **400** directs RC processor circuit (**230**) to send back to the call controller (**13**) a message indicating the length is not correct. The process **380** is then ended. At the call controller **13**, routines (not shown) stored in the program memory **324** may direct the processor circuit (**320** of Figure **15**) to respond to the incorrect length message by transmitting a message back to the mobile telephone (**12** shown in Figure **1**) to indicate that an invalid number has been dialled.

If the length of the amended callee identifier meets the criteria set forth at block **398**, then block **402** directs the RC processor circuit (**230**) to make a database request to the database (**23**) to determine whether or not the amended callee identifier is found in the DID field (**373**) of a record such as shown in Figure **11** in the DID bank table. If at block **402** the RC processor circuit (**230**) receives a response from the database (**23**) indicating that the reformatted callee identifier produced at block **396** is found in the DID field (**373**) of a record in the DID bank table, then the callee is a subscriber to the system and the call is classified as a private network call by directing the processor to block **404**, which directs the RC processor circuit (**230**) to copy the contents of the corresponding username field (**371** in Figure **11**) from the callee DID bank table record (**370** in Figure **11**) into the callee identifier store (**246** in Figure **8**). Thus, the RC processor circuit (**230**) locates a subscriber username associated with the reformatted callee identifier. The processor (**232**) is then directed to point B in Figure **18**A.

Calls to Subscribers in Different Nodes

Referring back to Figure **1**, as noted above, the gateway **18** is preferably associated with a pre-determined node, which in the illustrated embodiment is the first node **11**. Referring back to Figure **18**A, block **406** directs the processor (**232** of Figure **8**) to execute a process to determine whether or not the node associated with the reformatted callee identifier in the callee identifier store (**246** in Figure **8**, which, at block **404**, was set to be a username of the callee) is the same node that is associated with the gateway **18** illustrated in Figure **1**.

To do this, the processor (**232**) may, for example, identify a node associated with the gateway (**18**) by using an IP address associated with the gateway to determine a node identifier of the gateway. An IP address associated with the gateway (**18**) may, for example, be obtained from either the caller identifier field **362** or the call identifier field **368** of the RC request message **360** illustrated in Figure **17**, as each of these fields includes a portion following an "@" symbol that indicates an IP address of the gateway. In order to determine a node identifier associated with the gateway (**18**) using the IP address associated with gateway (**18**), the processor **232** (illustrated in Figure **8**) may access a gateway node association table stored in the database **23** (illustrated in Figure **1**).

Referring to Figure **19**, an exemplary gateway node association table is shown generally at **480**. The exemplary gateway node association table **480** includes first and second records **482** and **484**, each having a respective gateway IP address field **486** and a respective node identifier field **488**. It will be appreciated that the exemplary gateway node association table **480** is an example for illustration purposes only. The values in the gateway IP address fields **486** are preferably initialized when a gateway (such as the gateway **18** illustrated in Figure **1**) is installed as part of the system (**10**), and are preferably updated as the IP addresses of the respective gateways may change from time to time. The values in the node identifier fields **488** are also

preferably initialized when a gateway (such as the gateway **18** illustrated in Figure **1**) is installed as part of the system (**10**).

As indicated above, the reformatted callee identifier in the callee identifier store (**246** in Figure **8**) was set at block **404** in Figure **18**B to be a username of the callee from the username field **371** (illustrated in Figure **11**), and in this embodiment, a prefix of the username of the callee preferably indicates a node associated with the callee. In the illustrated embodiment, the left-most digit in the username of the callee is a continent code, which is a sufficient prefix to identify a node associated with the callee. However, it will be appreciated that in other embodiments, other prefixes or other information may identify the associated node. Preferably, the values in the node identifier fields **488** correspond to the prefixes of the usernames in the username fields **371** (illustrated in Figure **11**), so that the node associated with the callee is the same node that is associated with the gateway **18** illustrated in Figure **1** if the prefix of the username of the callee matches the node identifier associated with the gateway (**18**). Therefore, in the illustrated embodiment, if the reformatted callee identifier in the callee identifier store (**246** in Figure **8**) is **2001 1050 8667**, for example, then in the example of Figure **19**, the node associated with the callee is the same node as the node identified by the continent code "**2**" that is associated with the gateway associated with the IP address **20.14.102.5** in the record **482**, but is not the same node as the node identified by the continent code "**5**" that is associated with the gateway associated with the IP address **104.12.131.12** in the record **484**.

Referring back to Figure **18**A, if at block **406** the prefix of the username of the callee does not match the node identifier associated with the gateway (**18**), then the call is a "cross-domain" call, and block **408** in Figure **18**A directs the processor (**232** in Figure **8**) to set a call type flag in the temporary memory (**240** in Figure **8**) to indicate the call is a cross-domain call. Then, block **410** of Figure **18**A directs the processor (**232** of Figure **8**) to produce a routing message identifying an address on the private network with which the callee identified by the contents of the callee ID buffer is associated and to set a time

to live for the call at a maximum value of **99999**, for example. Routing messages and time to live values, and also a method of determining the node in the system with which the callee is associated, are further described in PCT Publication No. WO **2008/052340**. Once a routing message is produced at block **410**, block **412** directs the processor (**232** in Figure **8**) to cause the routing message to be sent to the call controller **13** shown in Figure **1**, and the process ends.

Referring back to Figure **18**B, if at block **392**, the callee identifier stored in the callee identifier store (**246** in Figure **8**) does not begin with an international dialing digit, then block **414** directs the processor (**232**) to determine whether or not the callee identifier begins with the same national dial digit code as assigned to the caller. To do this, the processor (**232**) is directed to refer to the retrieved caller dialing profile as shown in Figure **9**. In Figure **9**, the national dialing digit code **206** is the number **1**. Thus, if the callee identifier begins with the number **1**, then the processor (**232**) is directed to block **416** in Figure **18**B.

Block **416** directs the processor (**232** of Figure **8**) to examine the callee identifier to determine whether or not the digits following the NDD digit identify an area code that is the same as any of the area codes identified in the local area codes field **212** of the caller dialing profile **200** shown in Figure **9**. If not, block **418** of Figure **18**B directs the processor (**232**) to set the call type flag to indicate that the call is a national call. If the digits following the NDD digit identify an area code that is the same as a local area code associated with the caller as indicated by the caller dialing profile, block **420** directs the processor (**232**) to set the call type flag to indicate a local call, national style. After executing block **418** or **420**, block **422** directs the processor (**232**) to format the callee identifier into a pre-defined digit format to produce a re-formatted callee identifier by removing the national dialled digit and prepending a caller country code identified by the country code field **210** of the caller dialing profile shown in Figure **9**. The processor (**232**) is then

directed to block **398** of Figure **18**B to perform other processing as already described above.

If at block **414**, the callee identifier does not begin with a national dialled digit, block **424** directs the processor (**232**) to determine whether the callee identifier begins with digits that identify the same area code as the caller. Again, the reference for this is the retrieved caller dialing profile shown in Figure **9**. The processor (**232**) determines whether or not the first few digits of the callee identifier identify an area code corresponding to the contents of any area code identifier stored in the local area code field **212** of the retrieved caller dialing profile **200** (illustrated in Figure **9**). If so, then block **426** directs the processor (**232**) to set the call type flag to indicate that the call is a local call. It should be noted that the call will not necessarily be a local call in every case where the first few digits of the callee identifier identify an area code corresponding to the contents of an area code identifier stored in the local area code field **212** (illustrated in Figure **9**), and other determinations of when a call is to be considered local may be appropriate. However, it has been found that the determination described above for block **424** is satisfactory for some purposes. Next, block **428** directs the processor (**232**) to format the callee identifier into a pre-defined digit format to produce a reformatted callee identifier by prepending the caller country code to the callee identifier, the caller country code being determined from the country code field **210** of the retrieved caller dialing profile **200** shown in Figure **9**. The processor (**232**) is then directed to block **398** for further processing as described above.

If at block **424**, the callee identifier does not start with the same area code as the caller, block **430** directs the processor (**232** of Figure **8**) to determine whether the number of digits in the callee identifier, *i.e.* the length of the callee identifier, is within the range of digits indicated by the caller minimum local number length field **214** and the caller maximum local number length field **216** of the retrieved caller dialing profile **200** shown in Figure **9**, and whether there is more than one area code identifier stored in the local area code field **212** of the retrieved caller dialing profile. If the number of digits in the callee identifier

is within the aforementioned range and there is only one area code identifier stored in the local area code field (**212**), then block **432** directs the processor (**232**) to set the call type flag to indicate a local call and block **434** directs the processor (**232**) to format the callee identifier into a pre-defined digit format to produce a reformatted callee identifier by prepending to the callee identifier the caller country code (as indicated by the country code field **210** of the retrieved caller dialing profile **200** shown in Figure **9**) followed by the caller area code as indicated by the local area code stored in the local area code field **212** of the caller dialing profile **200** shown in Figure **9**. The processor (**232**) is then directed to block **398** of Figure **18**B for further processing as described above.

If at block **430**, the callee identifier has a length that does not fall within the range specified by the caller minimum local number length field (**214** in Figure **9**) and the caller maximum local number length field (**216** in Figure **9**), or if there is more than one area code identifier stored in the local area code field **212** of the retrieved caller dialing profile **200** illustrated in Figure **9**, then block **436** directs the processor (**232**) to send an error message back to the call controller (**13**), and the process ends.

In alternative embodiments, such as those illustrated in PCT Publication No. WO **2008/052340**, an additional block (**402** in Figure **8**B of PCT Publication No. WO **2008/052340**) may determine whether the callee identifier is a valid username. However, in the embodiment disclosed herein, the callee identifier is assumed to be a telephone number of the callee, and not a username.

From Figure **18**B, it will be appreciated that there are certain groups of blocks of codes that direct the processor **232** in Figure **8** to determine whether the callee identifier has certain features such as an international dialing digit, a national dialing digit, an area code and a length that meet certain criteria, and cause the processor **232** to reformat the callee identifier stored in the callee identifier store **246** in Figure **8**, as necessary into a predetermined target format including only a country code, area code, and a normal telephone number, for example, to cause the callee identifier to be compatible with the

E.**164** number plan standard in this embodiment. This enables block **402** in Figure **18B** to have a consistent format of callee identifiers for use in searching through the DID bank table records **370** of the type shown in Figure **11** to determine how to route calls to subscribers on the same system. Effectively, therefore blocks **392**, **414**, **424**, and **430** establish call classification criteria for classifying the call as a public network call or a private network call. Block **402** classifies the call, depending on whether or not the formatted callee identifier has a DID bank table record, and this depends on how the call classification criteria are met.

Calls to Non-Subscribers

Not all calls will be to subscribers, and this will be detected by the processor **232** of Figure **8** when it executes block **402** in Figure **18B**, and does not find a DID bank table record (**370** illustrated in Figure **11**) that is associated with the callee, in the DID bank table. When this occurs, the call is classified as a public network call, by directing the processor (**232**) to point C in Figure **18C**.

Referring to Figure **18C**, block **438** directs the processor (**232**) to determine whether the formatted callee identifier in the callee identifier store **246** in Figure **8** corresponds to an access code in the access code field **173** of a record in the access code association table **170** illustrated in Figure **10** that is associated with a callee identifier. Because the callee identifier in the callee identifier store **246** in Figure **8** has been formatted as described above with reference to Figure **18B**, block **438** may involve determining whether an access code in the access code field **173** of a record of the access code association table **170** (illustrated in Figure **10**) matches the formatted callee identifier in the callee identifier store **246** in Figure **8**, and also whether a callee identifier (as opposed to the "null" value assigned on initialization) is stored in the callee identifier field **177** in association with the access code. As noted above, for simplicity, this description is directed to embodiments wherein an access code association table **170** associates access codes with respective callee identifiers, caller identifiers, timeout values, and timestamp values, although it will be appreciated that the processes described herein for

records in the access code association table **170** may additionally or alternatively be applied to DID bank table records **370** in an analogous manner.

If at block **438** the formatted callee identifier in the callee identifier store **246** in Figure **8** is the same as an access code in the access code field (**173**) of a record of the access code association table **170** illustrated in Figure **10** that is associated with a callee identifier, then block **440** directs the processor (**232**) to determine whether the caller identifier in the caller identifier store **248** (illustrated in Figure **8**) is the same as the caller identifier in the caller identifier field (**179**) of the record of the access code association table (**170**), and thus whether the caller identifier in the caller identifier field (**179**) of the record of the access code association table (**170**) identifies the mobile telephone identified by the caller identifier in the caller identifier store **248**. If not, then block **442** directs the processor (**232**) to send an error message to the call controller (**13**), and the process ends.

But if at block **440** the caller identifier in the caller identifier store **248** (illustrated in Figure **8**) corresponds to the caller identifier in the caller identifier field (**179**) of the record of the access code association table (**170**), then the routing controller (**30**) will produce a routing message that will cause the call controller to establish communication through the IP network (**26**) to the callee in response to a call received at a channel (**20**, **22**, or **24**). Preferably, block **444** includes codes that direct the processor (**232**) to determine whether the association of the access code with the callee identifier has expired, and thus whether the usability of the access code to initiate a call to the callee has expired, in the manner described above for block **278** in Figure **12**. If at block **444** the association of the access code with the callee identifier has expired, then block **442** directs the processor (**232**) to send an error message to the call controller (**13**), and the process ends. Thus the routing controller produces a routing message that causes the call controller to establish the call only when the association of the access code with the callee identifier has not expired.

It will be appreciated that in alternative embodiments, one or more of the caller identifier, timeout, and timestamp fields **179**, **181**, and **182** may be omitted from the access code association table **170** illustrated in Figure **10**, and in these embodiments, one or more of the blocks **440**, **442**, and **444** may also be omitted.

If at block **444** the association of the access code with the callee identifier has not expired, or if one or both of blocks **440** and **444** is omitted, then block **446** directs the processor (**232**) to store the callee identifier from the callee identifier field **177** of the record of the access code association table (**170**) in the callee identifier store **246** illustrated in Figure **8**. The processor (**232**) is then directed to point A in Figure **18**B to repeat the steps illustrated in Figure **18**B using the callee identifier retrieved from the callee identifier field (**177**) in the record of the access code association table (**170**).

However, if at block **438** the formatted callee identifier in the callee identifier store **246** in Figure **8** does not correspond to an access code in a record of the access code association table **170** illustrated in Figure **10** that is associated with a callee identifier, then block **448** of Figure **18**B causes the processor (**232**) to set the contents of the callee identifier store **246** of Figure **8** to be the newly formatted callee identifier, i.e., a number compatible with the E.**164** standard. Then, block **450** of Figure **18**B directs the processor (**232**) to generate a routing message identifying a gateway to the public network usable by the call controller (**13**) to establish a "public system" call. In one embodiment, block **450** includes codes that, for example, direct the processor (**232**) to search a database of route or master list records and to search a database of supplier records to identify at least one supplier operable to supply a communications link for the call, and to load a routing message buffer with supplier information, time to live values, and timeout values. An example of an implementation of these steps is described with reference to blocks **410**, **412**, **560**, **562**, **563**, **564**, **566**, and **571** in Figures **8**B and **8**D in PCT Publication No. WO **2008/052340**. Next, block **452** directs the processor

232 of Figure **10** to send the routing message to the call controller **13** in Figure **1**, and the process ends.

Calls to Subscribers Within the Same Node

Referring back to Figure **18**A, if at block **406**, the prefix of the username of the callee matches the node identifier associated with the gateway (**18**), then the call is on one domain, and block **454** directs the processor (**232**) to use the callee identifier in the callee identifier store **246** illustrated in Figure **8** (which, at block **404**, was set to be a username of the callee) to locate and retrieve a dialing profile for the callee. The dialing profile may be of the type shown in Figure **9**, for example. Block **456** of Figure **18**A then directs the processor **232** of Figure **8** to get call block, call forward, and voicemail records from the database **23** of Figure **1**, based on the username identified in the callee dialing profile retrieved by the processor at block **454**. Exemplary call block, call forward, and voicemail records are described in PCT Publication No. WO **2008/052340**.

Then block **458** directs the processor **232** of Figure **8** to determine whether or not the caller identifier received in the RC request message matches a block pattern stored in the call block record associated with the callee and retrieved at block **454**. If the caller identifier matches a block pattern, then block **460** directs the processor to send a drop call or non-completion message to the call controller (**13**) and the process is ended. If the caller identifier does not match a block pattern associated with the callee, then block **462** directs the processor (**232**) to determine whether or not call forwarding is required, as described in PCT Publication No. WO **2008/052340**.

If at block **462**, the call forwarding record for the callee indicates that no call forwarding is required, then the processor (**232**) is directed to block **464**, which directs the processor (**232**) to generate a routing message identifying an address on the private network, associated with the callee for a "private system" call. In one embodiment, block **464** includes codes that, for example, direct the processor (**232**) to store, in a routing message buffer, a username

and domain of the callee, time to live values, and an IP address of the current node, to determine whether or not the user identified by the callee identifier has paid for voicemail service and if so, to store voicemail information in the routing message buffer. An example of an implementation of these steps is described with reference to blocks **609**, **620**, **640**, **642**, and **644** in Figures **8**A and **8**C in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference. Next, block **466** directs the processor **232** of Figure **8** to cause the routing message to be sent to the call controller **13** in Figure **1**, and the process ends.

But if at block **462**, the call forwarding record for the callee indicates that call forwarding is required, then block **468** directs the processor (**232**) to search a dialing profile table to find a dialing profile record as shown in Figure **9**, for the user identified by the destination number field of the call forward record, as illustrated in PCT Publication No. WO **2008/052340**. The processor (**232**) is further directed to store the username and domain for that user and a time to live value in a routing message buffer, an example of which is described in PCT Publication No. WO **2008/052340**. This process is repeated for each call forwarding record associated with the callee identified by the callee identifier store **246** in Figure **8** to add to the routing message buffer all call forwarding usernames and domains associated with the callee.

Referring to Figures **1**, **18**A, and **18**C, the routing message sent at one of blocks **412**, **452**, and **466** is received at the call controller **13** and the call controller interprets the receipt of the routing message as a request to establish a call. Referring to Figure **15**, the program memory **324** of the call controller **13** includes a routing to gateway routine depicted generally at **346**.

Where a routing message received at the call controller **13** is of the type produced at block **464** shown in Figure **18**A, indicating that the callee is a system subscriber on the same node as the gateway (**18**) (such as a user of the VoIP telephone **36** illustrated in Figure **1**), the routing to gateway routine **346** may direct the microprocessor **322** to cause a message to be sent back

through the IP network **26** shown in Figure **1** to the VoIP telephone (**36**), using the IP address of the VoIP telephone (**36**) that is available from the callee username.

Alternatively, if the routing message received at the call controller **13** is of the type produced at block **410** shown in Figure **18**A, identifying a domain associated with another node in the system, the call controller **13** may send a SIP invite message along the high speed/high data throughput link **17** in communication with the other node. The other node may function as explained above and in PCT Publication No. WO **2008/052340**, in response to receipt of a SIP invite message.

If the routing message received at the call controller **13** is of the type produced at block **450** shown in Figure **18**C, indicating that the callee is not a subscriber to the system (such as a user of the PSTN telephone **32** that is in communication with the IP network **26** through the gateway **34** as illustrated in Figure **1**), the call controller sends one or more SIP invite messages to the suppliers identified in the routing message to identify the IP address of a supplier that is able to carry the call, such as the IP address of the gateway **34** illustrated in the example of Figure **1**. A process for identifying the IP address of a supplier that is able to carry the call is given in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference. In some cases, the gateway of the supplier that is able to carry the call will be the gateway **18** illustrated in Figure **1**, that is, the same gateway through which the caller telephone (**12**) initiated the call. For simplicity, the following description assumes that the gateways **18** and **34** are distinct gateways. It will be understood that in some cases, they may be the same gateway, but in these cases, the following steps may still be applied.

Referring to Figure **1**, the IP address of the gateway **34** is sent in a message from the call controller **13** to the media relay **28**, which responds with a message indicating an IP address to which the gateway **18** should send its audio/video traffic, and an IP address to which the gateway **34** should send its

audio/video for the call. The call controller conveys the IP address at which the media relay **28** expects to receive audio/video from the gateways **18** and **34**, to the gateways **18** and **34** in one or more messages. The gateway **18** replies to the call controller **13** with an IP address at which it would like to receive audio/video, and the call controller conveys that IP address to the media relay **28**. The call may then be conducted between the caller and callee through the media relay **28** and the gateways **18** and **34**.

If the call controller **13** receives a routing message of the type produced at block **464** shown in Figure **18**A, indicating that the callee is a system subscriber on the same node as the gateway (**18**) (such as a user of the VoIP telephone **36** illustrated in Figure **1**), and which has at least one call forwarding number and/or a voicemail number, the call controller attempts to establish a call to the callee VoIP telephone **36** by seeking from the callee telephone a message indicating an IP address to which the media relay **28** should send audio/video. If no such message is received from the callee telephone, no call is established. If no call is established within a pre-determined time, the call controller **13** attempts to establish a call with the next user identified in the call routing message in the same manner. This process is repeated until all call forwarding possibilities have been exhausted, in which case the call controller communicates with the voicemail server **19** identified in the routing message to obtain an IP address to which the media relay **28** should send audio/video and the remainder of the process mentioned above for establishing IP addresses at the media relay and the caller telephone is carried out to establish audio/video paths to allowing the caller to leave a voicemail message with the voicemail server.

When an audio/video path through the media relay **28** is established, a call timer maintained by the call controller **13** preferably logs the start date and time of the call and logs the call ID and an identification of the route (i.e., audio/video path IP address) for later use in billing.

Terminating the Call

Referring back to Figure **1**, in the event that the caller terminates a call, the gateway **18** sends a SIP bye message to the call controller **13**. Similarly, in the event that the callee terminates the call, the gateway **34** or the VoIP telephone **36** of the callee sends a SIP bye message to the call controller **13**. Exemplary SIP bye messages are described in PCT Publication No. WO **2008/052340**. The SIP bye message is received at the call controller **13**, and the call controller executes a process that involves decrementing the contents of the current number of concurrent calls field **222** dialing profile **200** of the caller as illustrated in Figure **9**, generating an RC call stop message (not shown), sending the RC call stop message to the routing controller **30**, and sending a "bye" message to the party that did not terminate the call. An exemplary RC call stop message, and an example of how these steps may be implemented, are described in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference.

When the routing controller **30** receives the RC call stop message from the call controller **13**, the routing controller executes an RC call stop message process that involves making various updates to subscriber, reseller, and supplier account records (not shown) following the call. Examples of subscriber, reseller, and supplier account records, and of updates to subscriber, reseller, and supplier account records, are described in PCT Publication No. WO **2008/052340**, which is incorporated herein by reference.

While specific embodiments of the invention have been described and illustrated, such embodiments should be considered illustrative of the invention only and not as limiting the invention.

What is claimed is:

1. A method of initiating a call to a callee using a mobile telephone, the method comprising:

receiving, from a user of the mobile telephone, a callee identifier associated with the callee;

transmitting an access code request message to an access server, said access code request message including said callee identifier;

receiving an access code reply message from the access server in response to said access code request message, said access code reply message including an access code different from said callee identifier and associated with said callee identifier; and

initiating a call with the mobile telephone using said access code to identify the callee.

2. The method of claim **1** wherein transmitting comprises transmitting said access code request message to said access server on a non-voice network.

3. The method of claim **1** wherein transmitting comprises transmitting a location identifier of a location associated with the mobile telephone to said access server.

4. The method of claim **3** wherein transmitting said location identifier comprises transmitting an IP address of the mobile telephone in a wireless IP network.

5. The method of claim **3** wherein transmitting said location identifier comprises transmitting an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

6. The method of claim **3** wherein transmitting said location identifier comprises transmitting a user-configured identifier of a location associated with the mobile telephone.

7. The method of claim **1** wherein receiving said access code reply message comprises receiving said access code reply message from said access server on a non-voice network.

8. The method of claim **1** wherein receiving said access code reply message comprises receiving, in said access code reply message, said access code temporarily associated with said callee identifier.

9. The method of claim **1** wherein receiving said access code reply message comprises receiving, in said access code reply message, a telephone number identifying a channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

10. The method of claim **9** wherein initiating said call comprises engaging a routing controller to route said call on said IP network to the callee.

11. The method of any one of claims **1** to **10** further comprising:

receiving from the mobile telephone said access code request message;

communicating with a routing controller to obtain from the routing controller said access code wherein said access code identifies a channel and is useable by the mobile telephone to

cause the routing controller to establish a call to the callee using the channel; and

transmitting said access code reply message to the mobile telephone.

12. A mobile telephone apparatus comprising:

means for receiving, from a user of the mobile telephone, a callee identifier associated with the callee;

transmitting means for transmitting an access code request message to an access server, said access code request message including said callee identifier;

means for receiving an access code reply message from the access server in response to said access code request message, said access code reply message including an access code different from said callee identifier and associated with said callee identifier; and

means for initiating a call using said access code to identify the callee.

13. The mobile telephone apparatus of claim **12** wherein said transmitting means comprises a non-voice network interface for transmitting said access code request message to said access server on a non-voice network.

14. The mobile telephone apparatus of claim **12** wherein said access code request message further includes a location identifier of a location associated with the mobile telephone.

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15. The mobile telephone apparatus of claim **14** wherein said location identifier comprises an IP address of the mobile telephone in a wireless IP network.

16. The mobile telephone apparatus of claim **14** wherein said location identifier comprises an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

17. The mobile telephone apparatus of claim **14** wherein said location identifier comprises a user-configured identifier of a location associated with the mobile telephone.

18. The mobile telephone apparatus of claim **12** wherein said means for receiving an access code reply message comprises a non-voice network interface for receiving said access code reply message on a non-voice network.

19. The mobile telephone apparatus of claim **12** wherein said access code includes a telephone number.

20. The mobile telephone apparatus of claim **12** wherein said means for initiating comprises a mobile telephone network interface.

21. A system for initiating a call to a callee, the system comprising:

the mobile telephone apparatus of any one of claims **12** – **20**, and further comprising;

a routing controller; and

an access server comprising:

means for receiving from the mobile telephone said access code request message;

means for communicating with said routing controller to obtain from said routing controller said access code wherein said access code identifies a channel and is useable by the mobile telephone to cause the routing controller to establish a call to the callee using the channel; and

means for transmitting said access code reply message including said access code to the mobile telephone.

22. A mobile telephone apparatus comprising:

a processor circuit;

a network interface in communication with said processor circuit; and

a computer readable medium in communication with said processor circuit and encoded with codes for directing said processor circuit to:

receive, from a user of the mobile telephone, a callee identifier associated with the callee;

cause an access code request message to be transmitted to an access server, said access code request message including said callee identifier;

receive an access code reply message from the access server in response to said access code request message,

said access code reply message including an access code different from said callee identifier and associated with said callee identifier; and

initiate a call using said access code to identify the callee.

23. The mobile telephone apparatus of claim **22** wherein said network interface comprises a non-voice network interface, and wherein said codes for directing said processor circuit to cause said access code request message to be transmitted include codes for directing said processor circuit to cause said access code request message to be transmitted to said access server using said non-voice network interface on a non-voice network.

24. The mobile telephone apparatus of claim **22** wherein said access code request message further includes a location identifier of a location associated with the mobile telephone.

25. The mobile telephone apparatus of claim **24** wherein said location identifier comprises an IP address of the mobile telephone in a wireless IP network.

26. The mobile telephone apparatus of claim **24** wherein said location identifier comprises an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

27. The mobile telephone apparatus of claim **24** wherein said location identifier comprises a user-configured identifier of a location associated with the mobile telephone.

28. The mobile telephone apparatus of claim **22** wherein said network interface comprises a non-voice network interface, and wherein said codes for directing said processor circuit to receive an access code

reply message include codes for directing said processor circuit to cause said access code reply message to be received from said access server using said non-voice network interface on a non-voice network.

29. The mobile telephone apparatus of claim **22** wherein said access code includes a telephone number identifying a channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

30. The mobile telephone apparatus of claim **22** wherein said network interface comprises a mobile telephone network interface, and wherein said codes for directing said processor circuit to initiate include codes for directing said processor circuit to cause a call to be initiated using said mobile telephone network interface on a mobile telephone network.

31. A system for initiating a call to a callee, the system comprising:

the mobile telephone of any one of claims **22 – 30,** and further comprising;

a routing controller; and

an access server comprising a processor circuit and a computer readable medium in communication with the processor circuit, the computer readable medium encoded with codes for directing said processor circuit to:

receive from the mobile telephone said access code request message;

communicate with said routing controller to obtain from said routing controller said access code wherein said access code identifies a channel and is useable by the mobile telephone to cause the routing controller to establish a call to the callee using the channel; and

transmit said access code reply message to the mobile telephone.

32. A computer readable medium encoded with codes for directing a processor circuit to initiate a call to a callee using a mobile telephone, said codes being operable to direct the processor circuit to:

receive, from a user of a mobile telephone, a callee identifier associated with a callee;

transmit an access code request message to an access server, said access code request message including said callee identifier;

receive an access code reply message from the access server in response to said access code request message, said access code reply message including an access code different from said callee identifier and associated with said callee identifier; and

initiate a call using said access code to identify the callee.

33. A method for enabling a mobile telephone to initiate a call to a callee through a channel, the method comprising:

receiving from the mobile telephone an access code request message including a callee identifier associated with the callee;

communicating with a routing controller to obtain from the routing controller an access code identifying the channel, said access code being different from the callee identifier and useable by the mobile telephone to initiate a call to the callee using the channel; and

transmitting an access code reply message including said access code to the mobile telephone.

34. The method of claim **33** wherein receiving comprises receiving said access code request message on a non-voice network.

35. The method of claim **33** further comprising causing the routing controller to produce said access code.

36. The method of claim **35** wherein producing comprises selecting said access code from a pool of access codes, where each access code in said pool of access codes identifies a respective telephone number.

37. The method of claim **36** further comprising determining a local calling area associated with the mobile telephone.

38. The method of claim **37** wherein determining comprises accessing a dialing profile associated with the caller, said dialing profile including a location field having contents identifying at least a default location of the caller.

39. The method of claim **37** wherein determining comprises receiving an IP address of the mobile telephone in a wireless IP network.

40. The method of claim **37** wherein determining comprises receiving an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

41. The method of claim **37** wherein determining comprises receiving a user-configured identifier of a location associated with the mobile telephone.

42. The method of claim **37** wherein selecting comprises selecting an access code in said local calling area associated with the mobile telephone.

43. The method of claim **36** wherein each access code in said pool of access codes further identifies a respective channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

44. The method of claim **43** further comprising causing said routing controller to establish communication through said IP network in response to a call received on said channel.

45. The method of claim **44** wherein producing further comprises storing a caller identifier associated with the mobile telephone in association with said access code.

46. The method of claim **45** wherein causing said routing controller to establish communication comprises causing said routing controller to establish communication only if said caller identifier associated with said access code identifies the mobile telephone.

47. The method of claim **36** wherein producing further comprises storing said callee identifier in association with said access code.

48. The method of claim **47** wherein producing further comprises searching said pool of access codes for an access code associated with said callee identifier to identify the channel usable by the mobile telephone to initiate a call to the callee.

49. The method of claim **35** wherein producing further comprises storing, in association with said access code, a timestamp for use in determining when the usability of said access code to initiate a call to the callee will expire.

50. The method of claim **49** wherein causing said routing controller to establish communication comprises causing said routing controller to establish communication only if the usability of said access code to initiate a call to the callee has not expired.

51. The method of claim **33** wherein transmitting comprises transmitting said access code reply message on a non-voice network.

52. A system for enabling a mobile telephone to initiate a call to a callee through a channel, the system comprising:

means for receiving from the mobile telephone an access code request message including a callee identifier associated with the callee;

means for communicating with said routing controller to obtain from said routing controller an access code identifying the channel, said access code being different from the callee identifier and useable by the mobile telephone to initiate a call to the callee using the channel; and

means for transmitting an access code reply message including said access code to the mobile telephone.

53. The system of claim **52** wherein said means for receiving comprises a non-voice network interface for receiving said access code request message on a non-voice network.

54. The system of claim **52** further comprising means for producing said access code.

55. The system of claim **54** wherein said means for producing comprises a processor circuit operably configured to select said access code from a pool of access codes, where each access code in said pool of access codes identifies a respective telephone number.

56. The system of claim **55** wherein said processor circuit is operably configured to determine a local calling area associated with the mobile telephone.

57. The system of claim **56** wherein said processor circuit is operably configured to determine a local calling area associated with the mobile telephone using a dialing profile associated with the caller, said dialing profile including a location field having contents identifying at least a default location of the caller.

58. The system of claim **56** wherein said processor circuit is operably configured to determine a local calling area associated with the mobile telephone using an IP address of the mobile telephone in a wireless IP network.

59. The system of claim **56** wherein said processor circuit is operably configured to determine a local calling area associated with the mobile telephone using an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

60. The system of claim **56** wherein said processor circuit is operably configured to determine a local calling area associated with the mobile telephone using a user-configured identifier of a location associated with the mobile telephone.

61. The system of claim **56** wherein said processor circuit is operably configured to select an access code in said local calling area associated with the mobile telephone.

62. The system of claim **55** wherein each access code in said pool of access codes further identifies a respective channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

63. The system of claim **62** wherein said processor circuit is operably configured to establish communication through said IP network in response to a call received on said channel.

64. The system of claim **63** wherein said processor circuit is operably configured to store a caller identifier associated with the mobile telephone in association with said access code.

65. The system of claim **64** wherein said processor circuit is operably configured to cause said routing controller to establish communication only if said caller identifier associated with said access code identifies the mobile telephone.

66. The system of claim **55** wherein said processor circuit is operably configured to store said callee identifier in association with said access code.

67. The system of claim **66** wherein said processor circuit is operably configured to search said pool of access codes for an access code

associated with said callee identifier to identify the channel usable by the mobile telephone to initiate a call to the callee.

68. The system of claim **54** wherein said processor circuit is operably configured to store, in association with said access code, a timestamp for use in determining when the usability of said access code to initiate a call to the callee will expire.

69. The system of claim **68** wherein said processor circuit is operably configured to establish communication only if the usability of said access code to initiate a call to the callee has not expired.

70. The system of claim **52** wherein said means for transmitting comprises a non-voice network interface for transmitting said access code reply message on a non-voice network.

71. A system for enabling a mobile telephone to initiate a call to a callee through a channel, the system comprising:

a processor circuit;

a network interface in communication with said processor circuit; and

a computer readable medium in communication with said processor circuit and encoded with codes for directing said processor circuit to:

receive from the mobile telephone an access code request message including a callee identifier associated with the callee;

communicate with said routing controller to obtain from said routing controller an access code identifying the channel, said access code being different from the callee identifier and useable by the mobile telephone to initiate a call to the callee using the channel; and

cause an access code reply message including said access code to be transmitted to the mobile telephone.

72. The system of claim **71** wherein said network interface comprises a non-voice network interface, and wherein said codes for directing said processor circuit to receive include codes for directing said processor circuit to cause said access code request message to be received using said non-voice network interface on a non-voice network.

73. The system of claim **71** wherein said computer readable medium is further encoded with codes for directing said processor circuit to cause said access code to be produced.

74. The system of claim **73** wherein said codes for directing said processor circuit to cause said access code to be produced cause said access code to be selected from a pool of access codes, where each access code in said pool of access codes identifies a respective telephone number.

75. The system of claim **74** wherein said computer readable medium is further encoded with codes for directing said processor circuit to cause to be determined a local calling area associated with the mobile telephone.

76. The system of claim **75** wherein said codes for directing said processor circuit to cause to be determined cause a dialing profile associated with

the caller to be accessed, said dialing profile including a location field having contents identifying at least a default location of the caller.

77. The system of claim **75** wherein said codes for directing said processor circuit to cause to be determined cause to be received an IP address of the mobile telephone in a wireless IP network.

78. The system of claim **75** wherein said codes for directing said processor circuit to cause to be determined cause to be received an identifier of a wireless voice signal station in wireless communication with the mobile telephone.

79. The system of claim **75** wherein said codes for directing said processor circuit to cause to be determined cause to be received a user-configured identifier of a location associated with the mobile telephone.

80. The system of claim **75** wherein said codes for directing said processor circuit to cause said access code to be produced further cause to be selected an access code in said local calling area associated with the mobile telephone.

81. The system of claim **74** wherein each access code in said pool of access codes further identifies a respective channel operably configured to cooperate with an IP network to cause a call involving the mobile telephone and the callee to be routed through the IP network.

82. The system of claim **81** wherein said computer readable medium is further encoded with codes for directing said processor circuit to cause communication through said IP network to be established in response to a call received on said channel.

83. The system of claim **82** wherein said codes for directing said processor circuit to cause said access code to be produced cause a caller

identifier associated with the mobile telephone to be stored in association with said access code.

84. The system of claim **83** wherein said codes for directing said processor circuit to cause communication to be established cause communication to be established only if said caller identifier associated with said access code identifies the mobile telephone.

85. The system of claim **74** wherein said codes for directing said processor circuit to cause said access code to be produced cause said callee identifier to be stored in association with said access code.

86. The system of claim **85** wherein said codes for directing said processor circuit to cause said access code to be produced cause said pool of access codes to be searched for an access code associated with said callee identifier to identify the channel usable by the mobile telephone to initiate a call to the callee.

87. The system of claim **73** wherein said codes for directing said processor circuit to cause said access code to be produced cause a timestamp for use in determining when the usability of said access code to initiate a call to the callee will expire, to be stored in association with said access code.

88. The system of claim **87** wherein said codes for directing said processor circuit to cause communication to be established cause communication to be established only if the usability of said access code to initiate a call to the callee has not expired.

89. The system of claim **71** wherein said network interface comprises a non-voice network interface, and wherein codes for directing said processor circuit to transmit include codes for directing said processor

circuit to cause said access code reply message to be transmitted using said non-voice network interface on a non-voice network.

90. A computer readable medium encoded with codes for directing a processor circuit to enable a mobile telephone to initiate a call to a callee through a channel, the codes being operable to direct the processor circuit:

receive from the mobile telephone an access code request message including a callee identifier associated with the callee;

communicate with said routing controller to obtain from said routing controller an access code identifying the channel, said access code being different from the callee identifier and useable by the mobile telephone to initiate a call to the callee using the channel; and

cause an access code reply message including said access code to be transmitted to the mobile telephone.

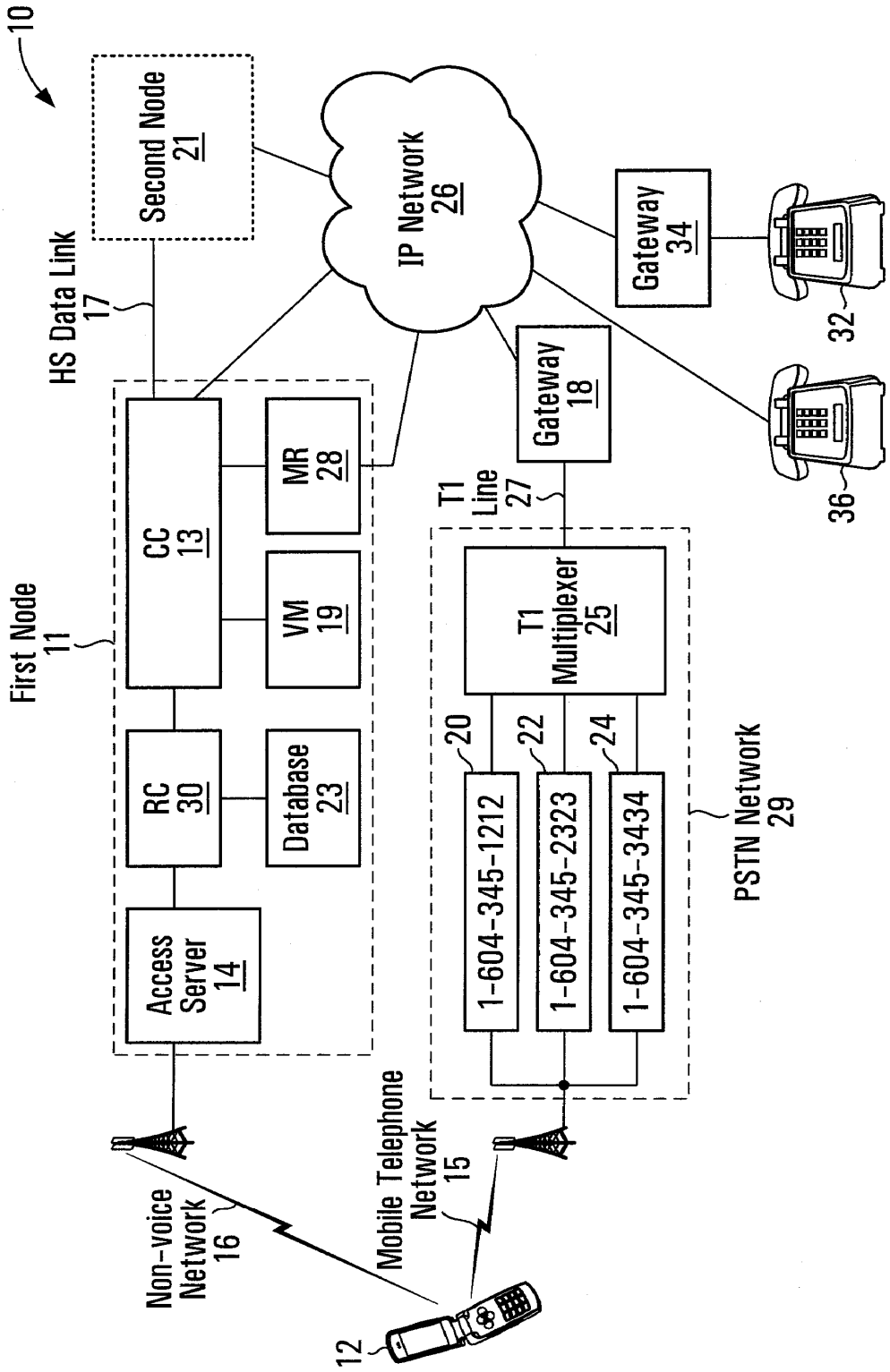

10
Second Node 21
IP Network 26
Gateway 34
32
HS Data Link 17
Gateway 18
36
T1 Line 27
First Node 11
CC 13
MR 28
VM 19
T1 Multiplexer 25
RC 30
Database 23
20
22
24
1-604-345-1212
1-604-345-2323
1-604-345-3434
PSTN Network 29
Access Server 14
Mobile Telephone Network 15
Non-voice Network 16
12

FIG. 1

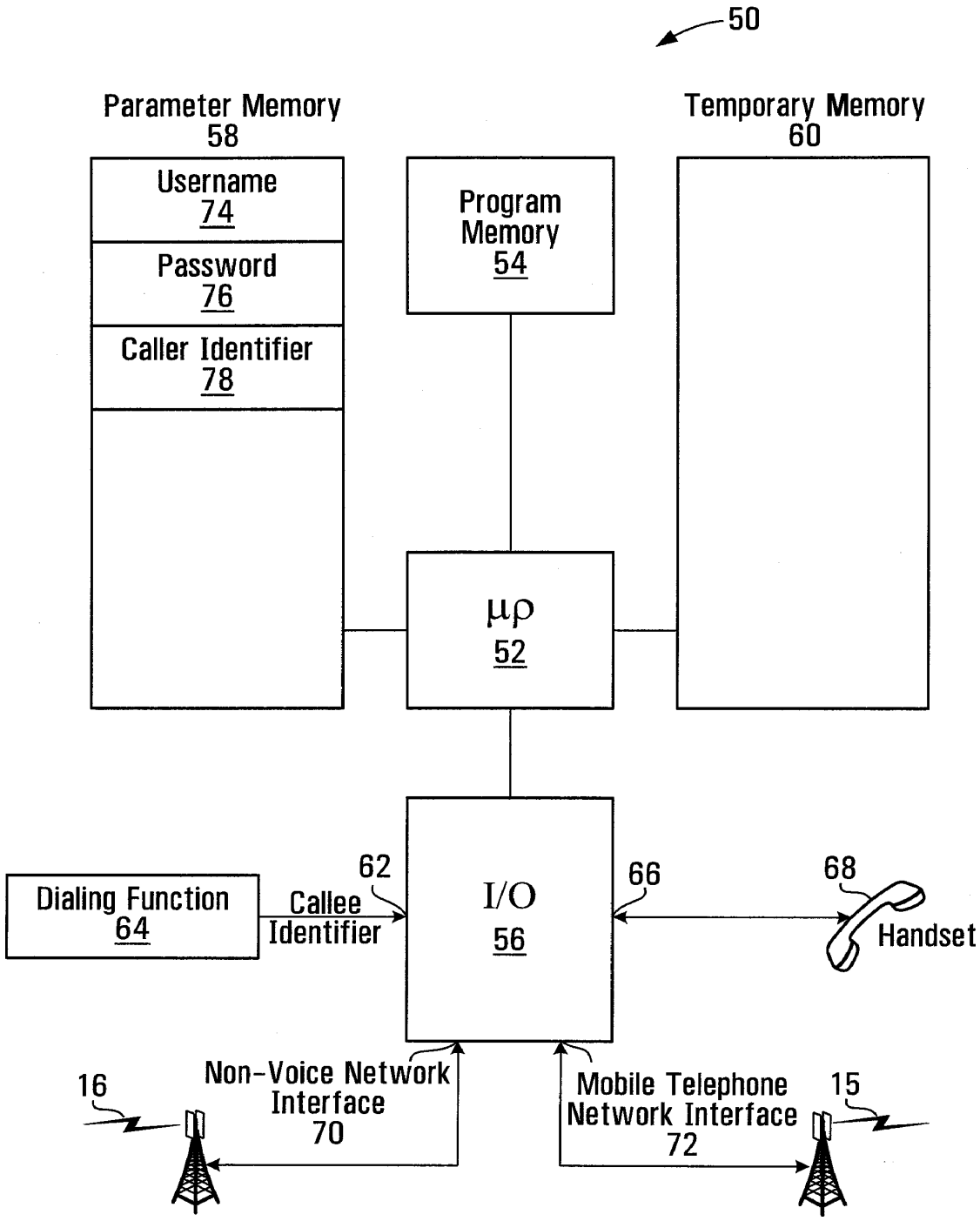

Mobile Telephone (12)
50
Parameter Memory
58
Username
74
Password
76
Caller Identifier
78
Program
Memory
54
Temporary Memory
60
μρ
52
I/O
56
Dialing Function
64
62
Callee
Identifier
66
68
Handset
16
Non-Voice Network
Interface
70
Mobile Telephone
Network Interface
72
15

FIG. 2

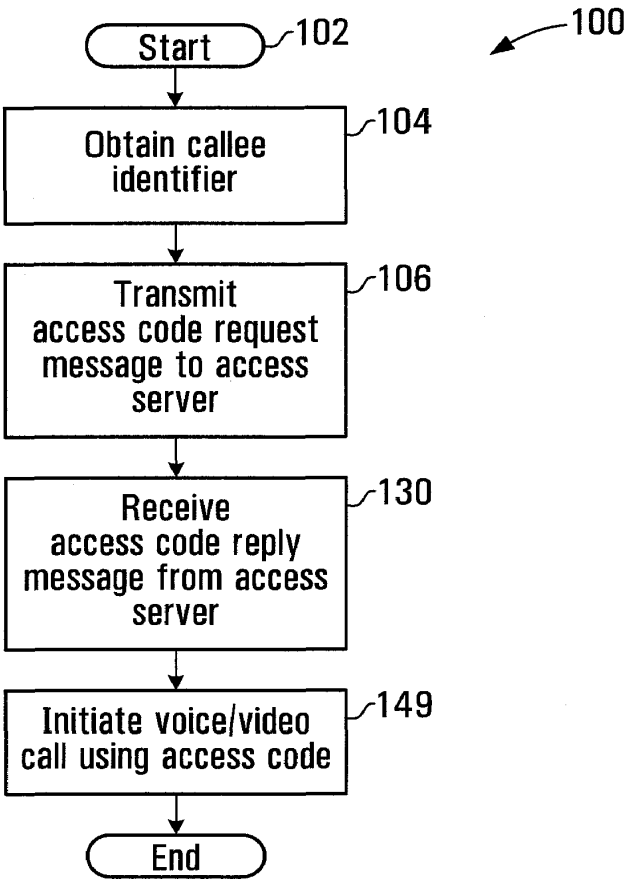


FIG. 3

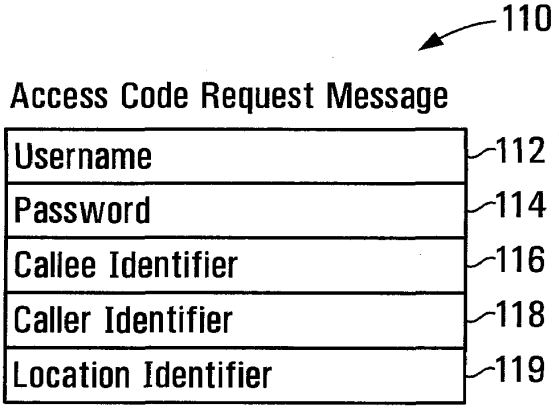


FIG. 4

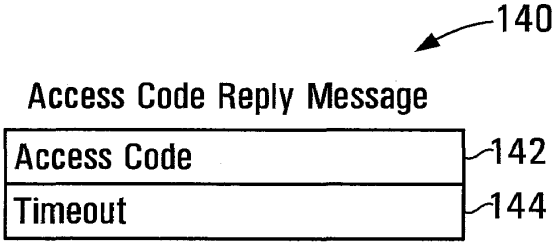


FIG. 5

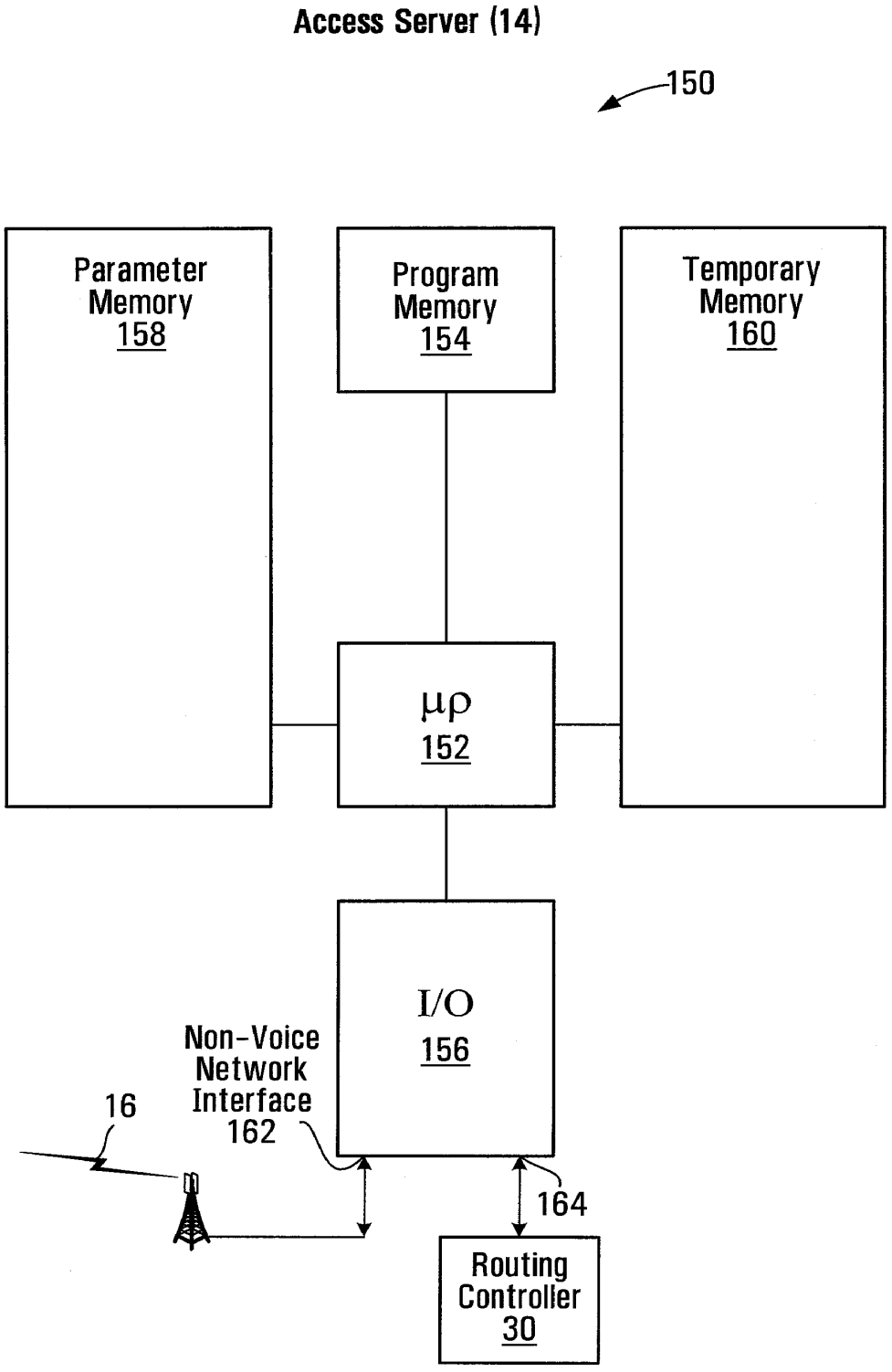


FIG. 6

META 1002
META V. VOIP-PAL

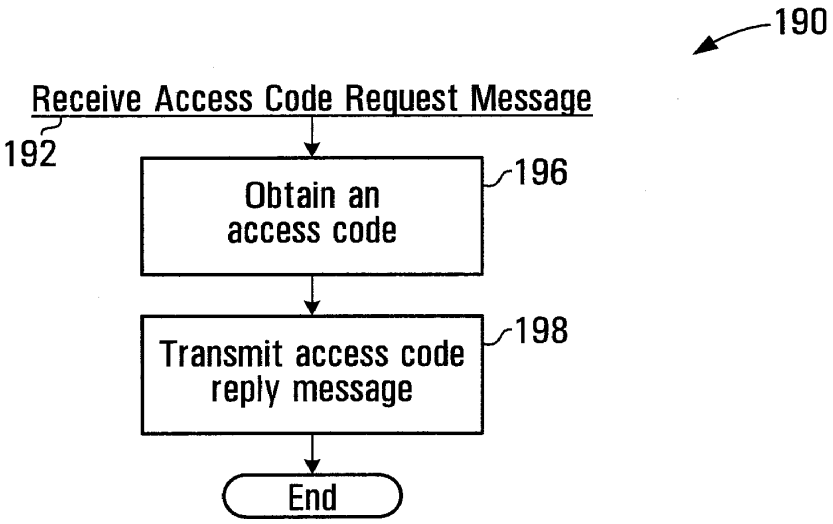


FIG. 7

META 1002
META V. VOIP-PAL

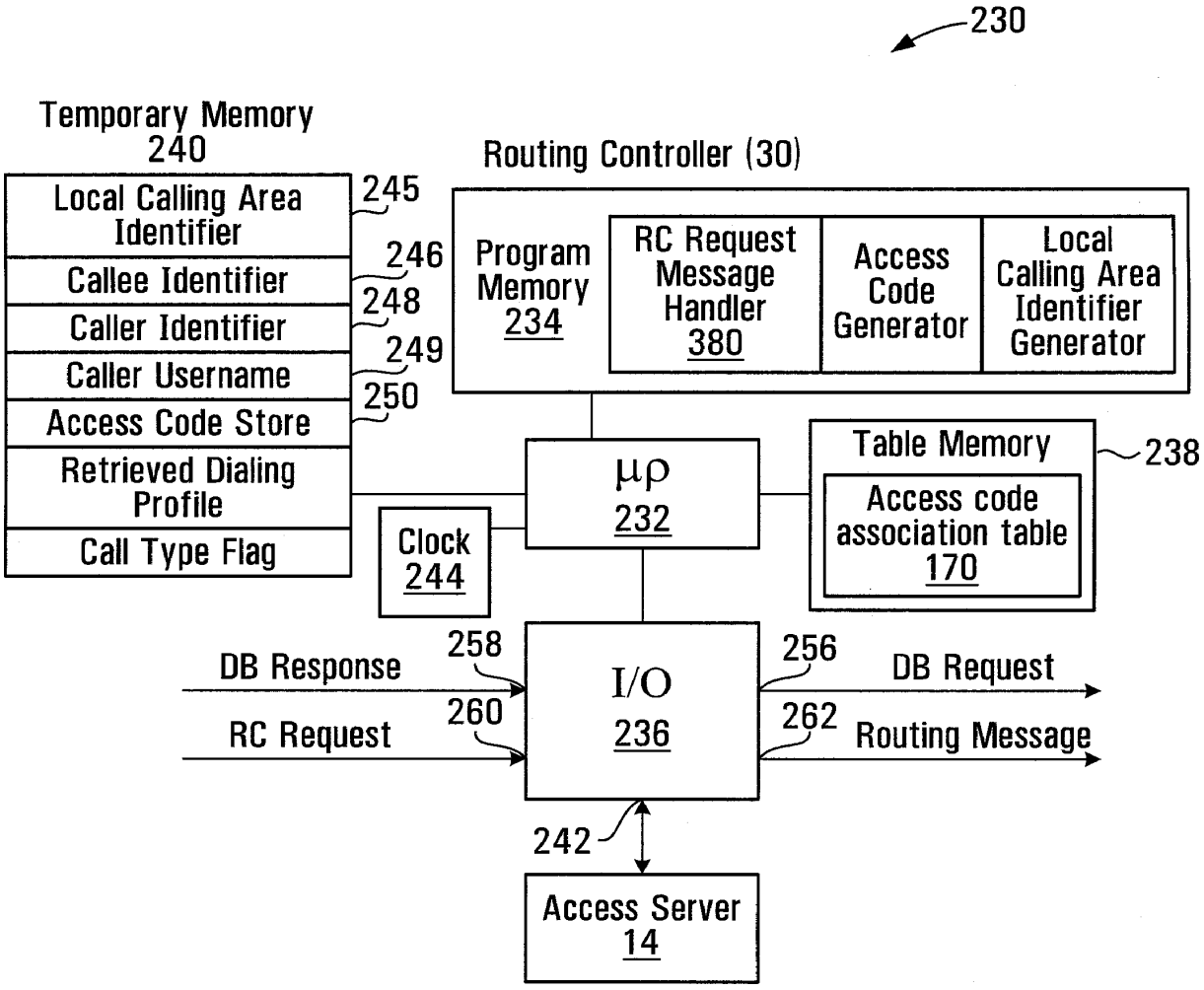

230
Temporary Memory
240
Local Calling Area Identifier
Callee Identifier
Caller Identifier
Caller Username
Access Code Store
Retrieved Dialing Profile
Call Type Flag
245
246
248
249
250
Routing Controller (30)
Program Memory 234
RC Request Message Handler 380
Access Code Generator
Local Calling Area Identifier Generator
μρ 232
Clock 244
Table Memory
238
Access code association table 170
DB Response
RC Request
258
260
I/O 236
256
262
DB Request
Routing Message
242
Access Server 14

FIG. 8

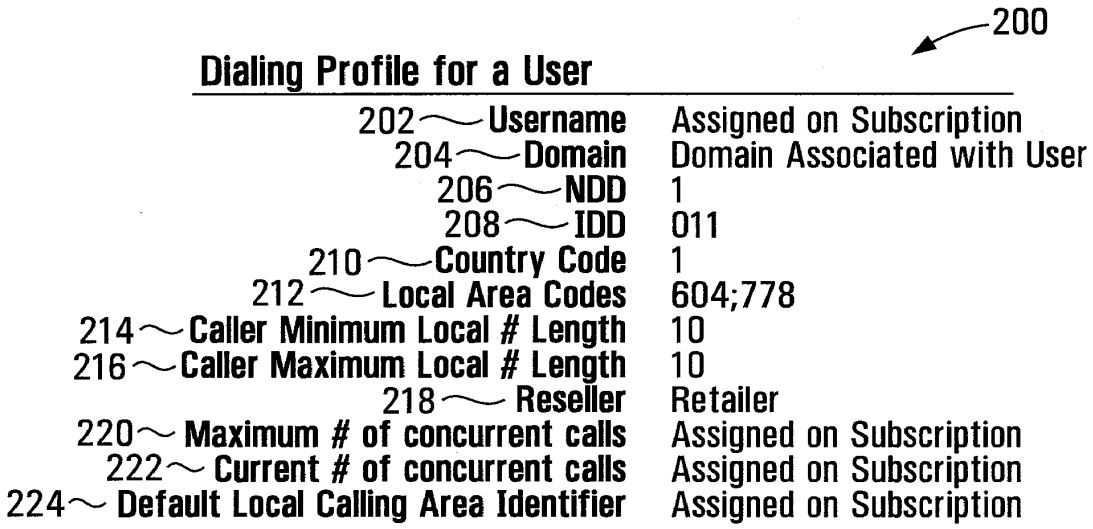

200
Dialing Profile for a User
202 Username
Assigned on Subscription
204 Domain
Domain Associated with User
206 NDD
1
208 IDD
011
210 Country Code
1
212 Local Area Codes
604;778
214 Caller Minimum Local # Length
10
216 Caller Maximum Local # Length
10
218 Reseller
Retailer
220 Maximum # of concurrent calls
Assigned on Subscription
222 Current # of concurrent calls
Assigned on Subscription
224 Default Local Calling Area Identifier
Assigned on Subscription

FIG. 9

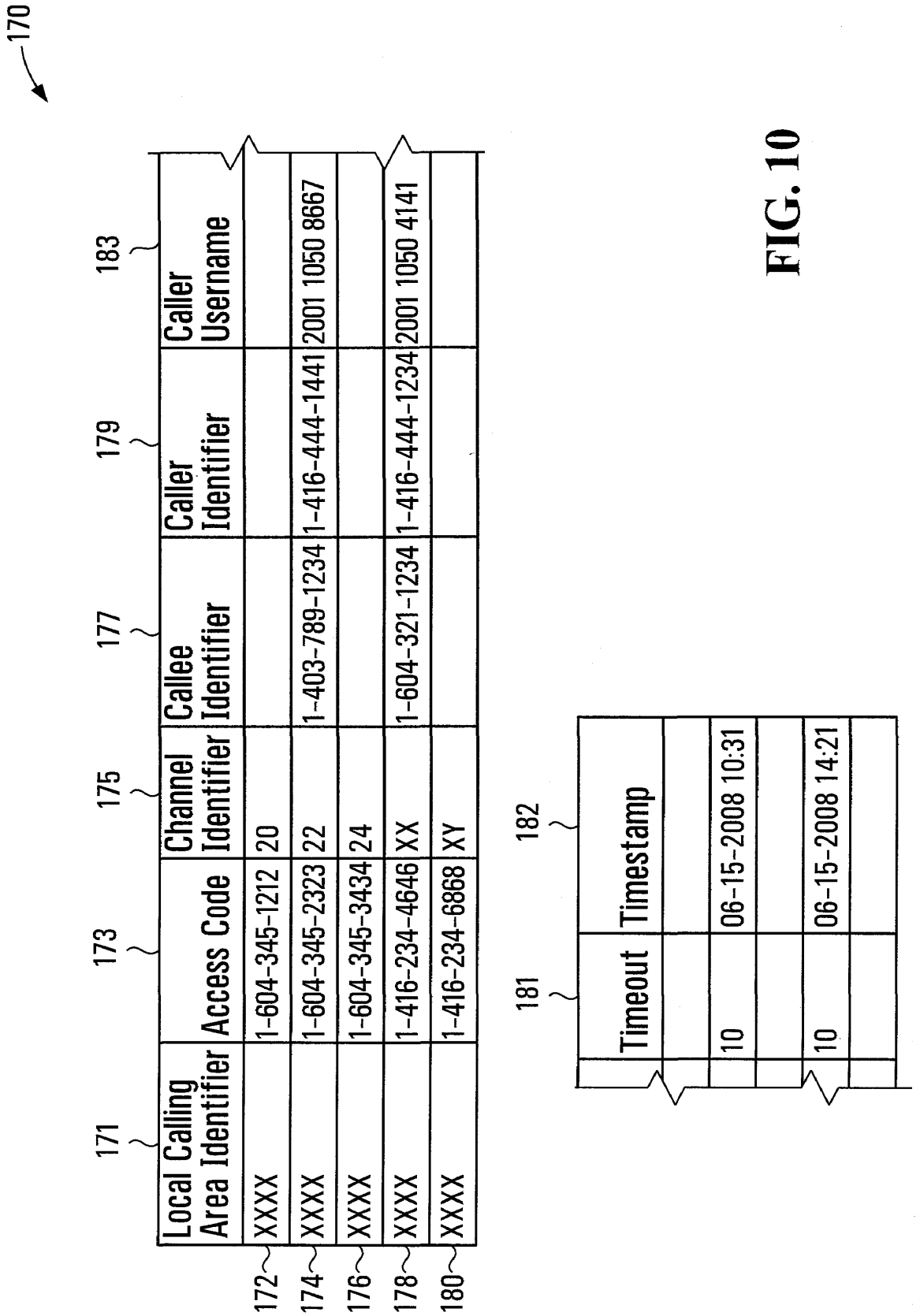

170
171
173
175
177
179
183
Local Calling Area Identifier
Access Code
Channel Identifier
Callee Identifier
Caller Identifier
Caller Username
172
XXXX
1-604-345-1212
20
174
XXXX
1-604-345-2323
22
1-403-789-1234
1-416-444-1441
2001 1050 8667
176
XXXX
1-604-345-3434
24
178
XXXX
1-416-234-4646
XX
1-604-321-1234
1-416-444-1234
2001 1050 4141
180
XXXX
1-416-234-6868
XY
181
182
Timeout
Timestamp
10
06-15-2008 10:31
10
06-15-2008 14:21

FIG. 10

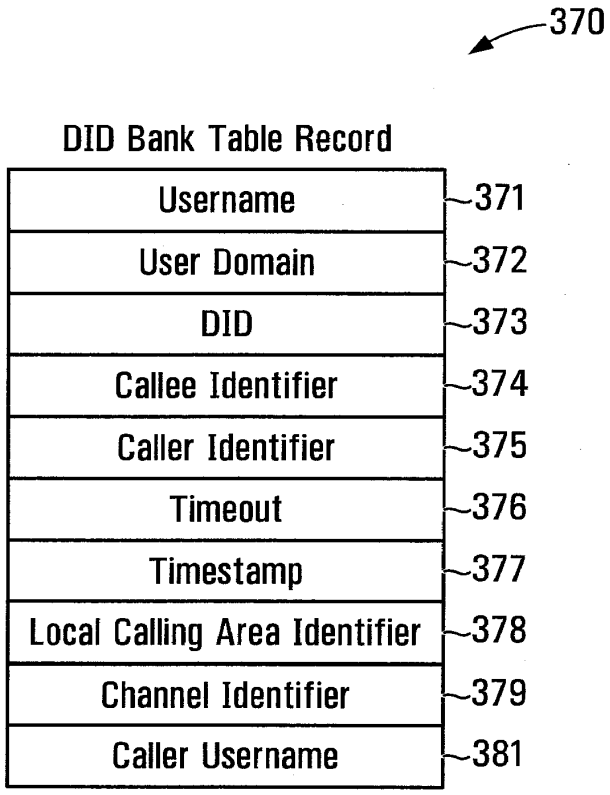

370
DID Bank Table Record
Username
371
User Domain
372
DID
373
Callee Identifier
374
Caller Identifier
375
Timeout
376
Timestamp
377
Local Calling Area Identifier
378
Channel Identifier
379
Caller Username
381

FIG. 11

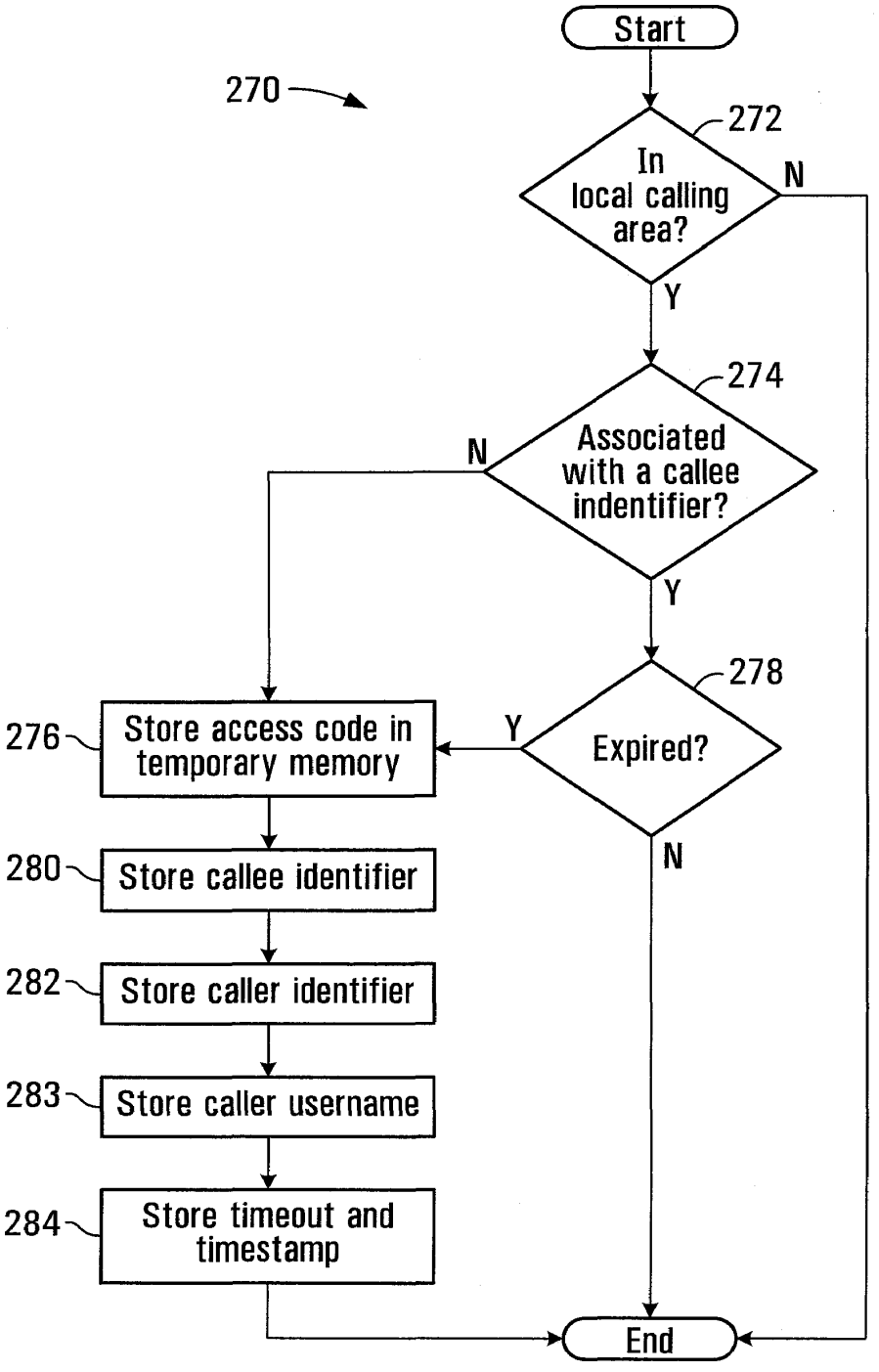


FIG. 12

META 1002
META V. VOIP-PAL

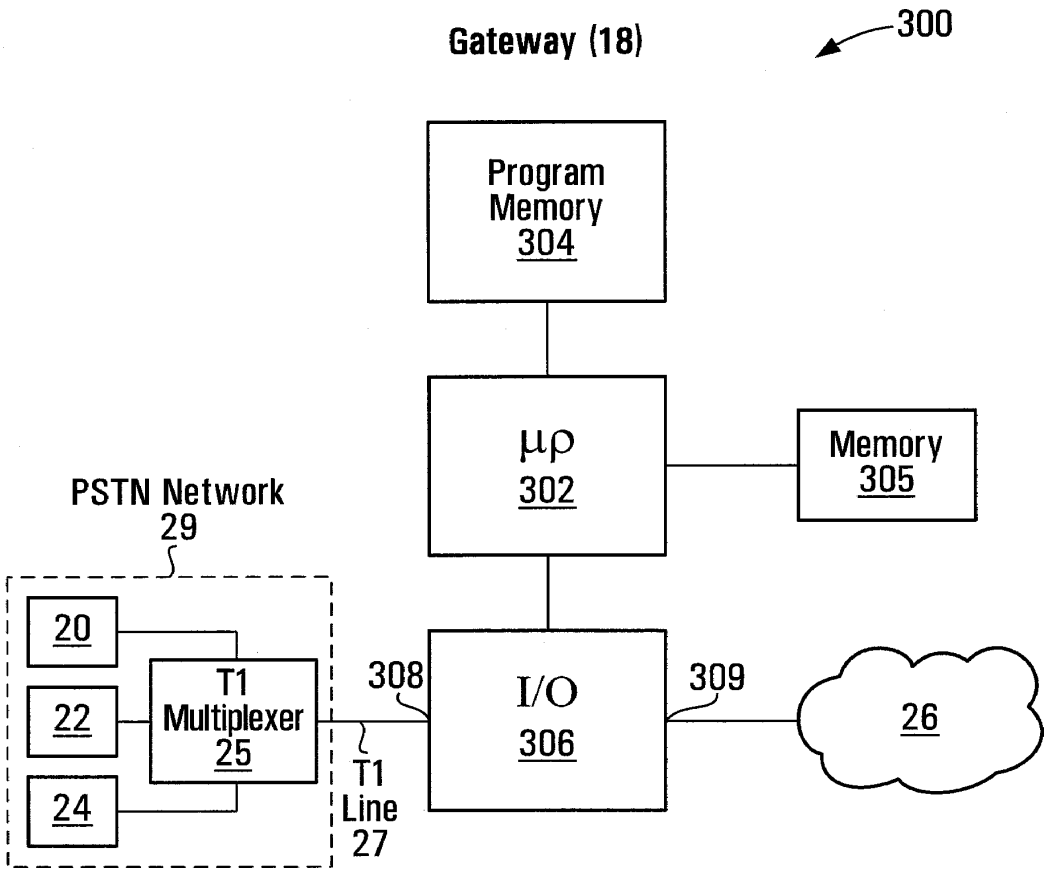


FIG. 13

310

SIP Invite Message

312	Caller Identifier	1-604-678-1234@20.14.102.5
314	Callee Identifier	1-604-345-1212
315	Digest Parameter	XXXXXXX
316	Call Identifer	FF10@20.14.102.5
317	IP Address	20.14.102.5
318	Gateway UDP Port	12378

FIG. 14

META 1002
META V. VOIP-PAL

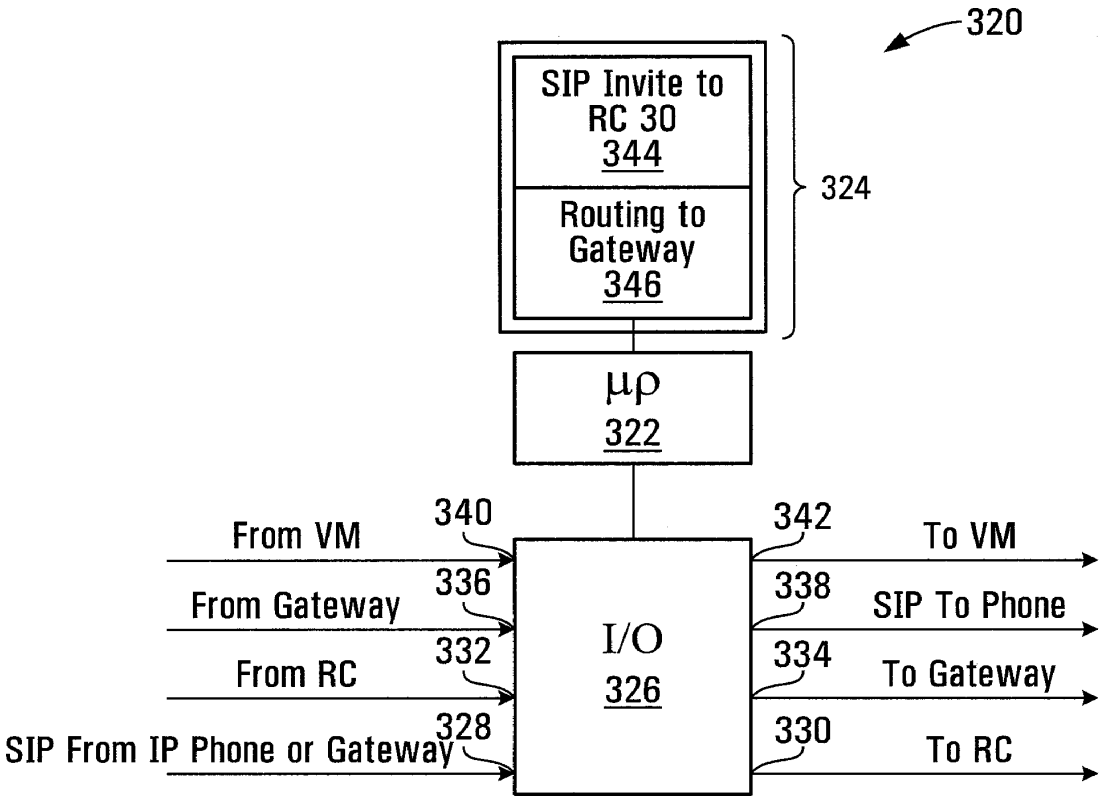

Call Controller (13)
320
SIP Invite to RC 30
344
Routing to Gateway
346
324
μρ
322
From VM
340
From Gateway
336
From RC
332
SIP From IP Phone or Gateway
328
I/O
326
342
To VM
338
SIP To Phone
334
To Gateway
330
To RC

FIG. 15

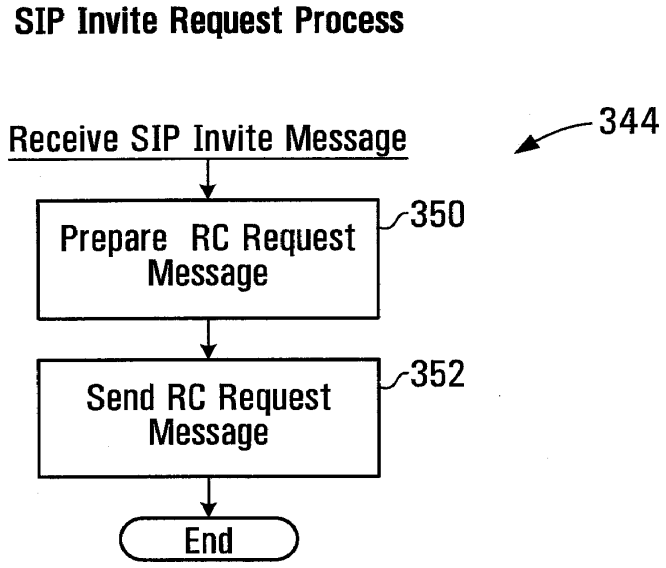

SIP Invite Request Process
Receive SIP Invite Message
344
Prepare RC Request Message
350
Send RC Request Message
352
End

FIG. 16

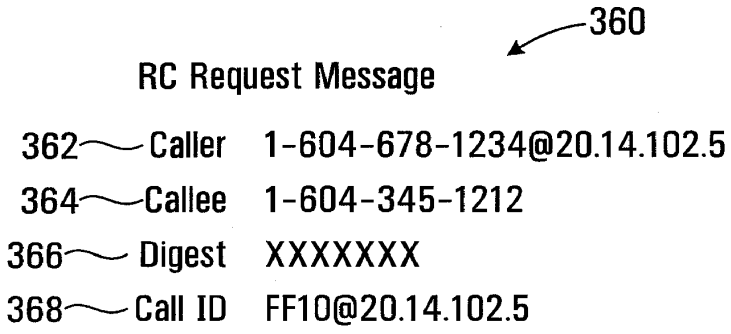

360
RC Request Message
362 Caller 1-604-678-1234@20.14.102.5
364 Callee 1-604-345-1212
366 Digest XXXXXXX
368 Call ID FF10@20.14.102.5

FIG. 17

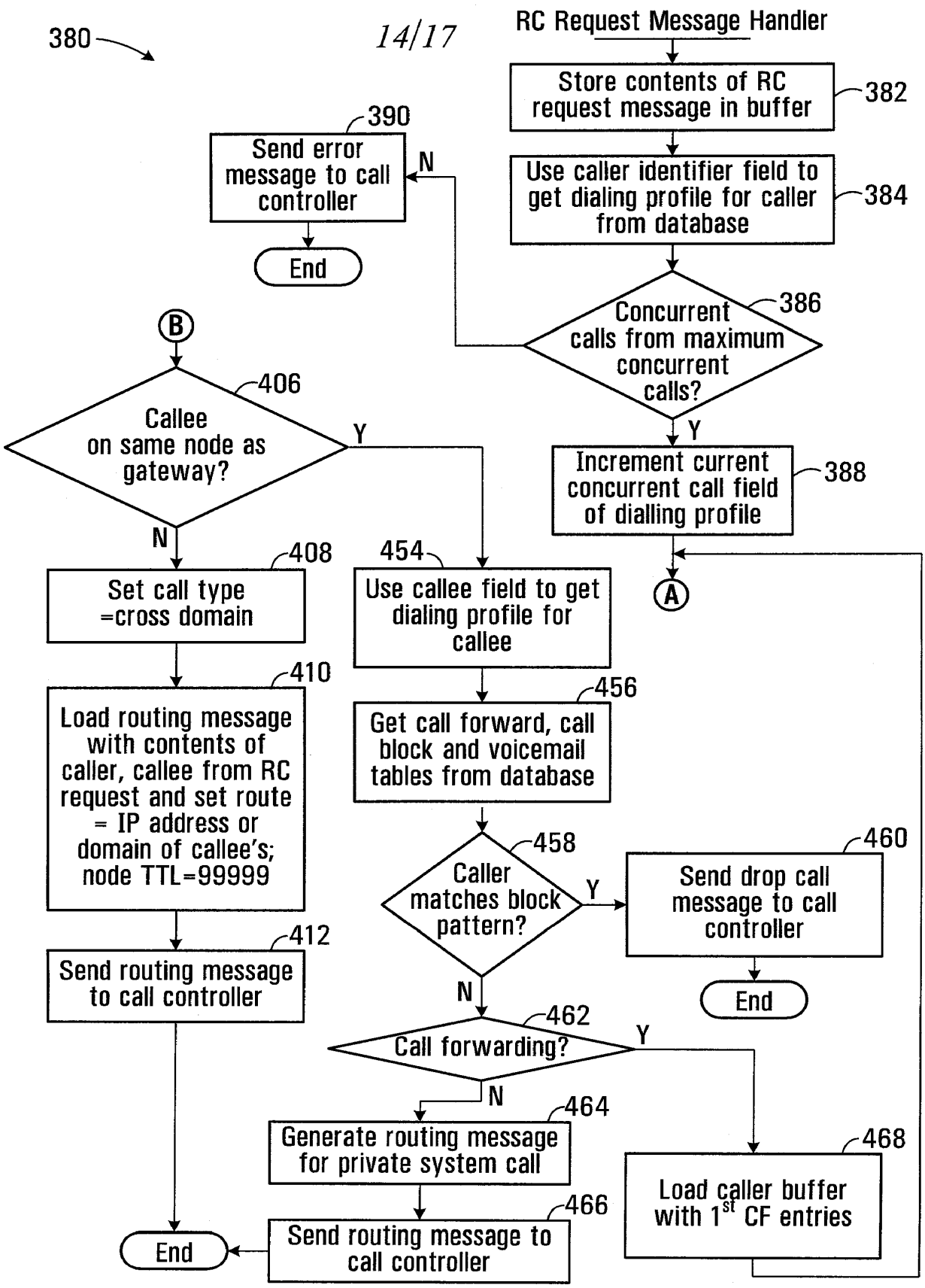


FIG. 18A

META 1002
META V. VOIP-PAL

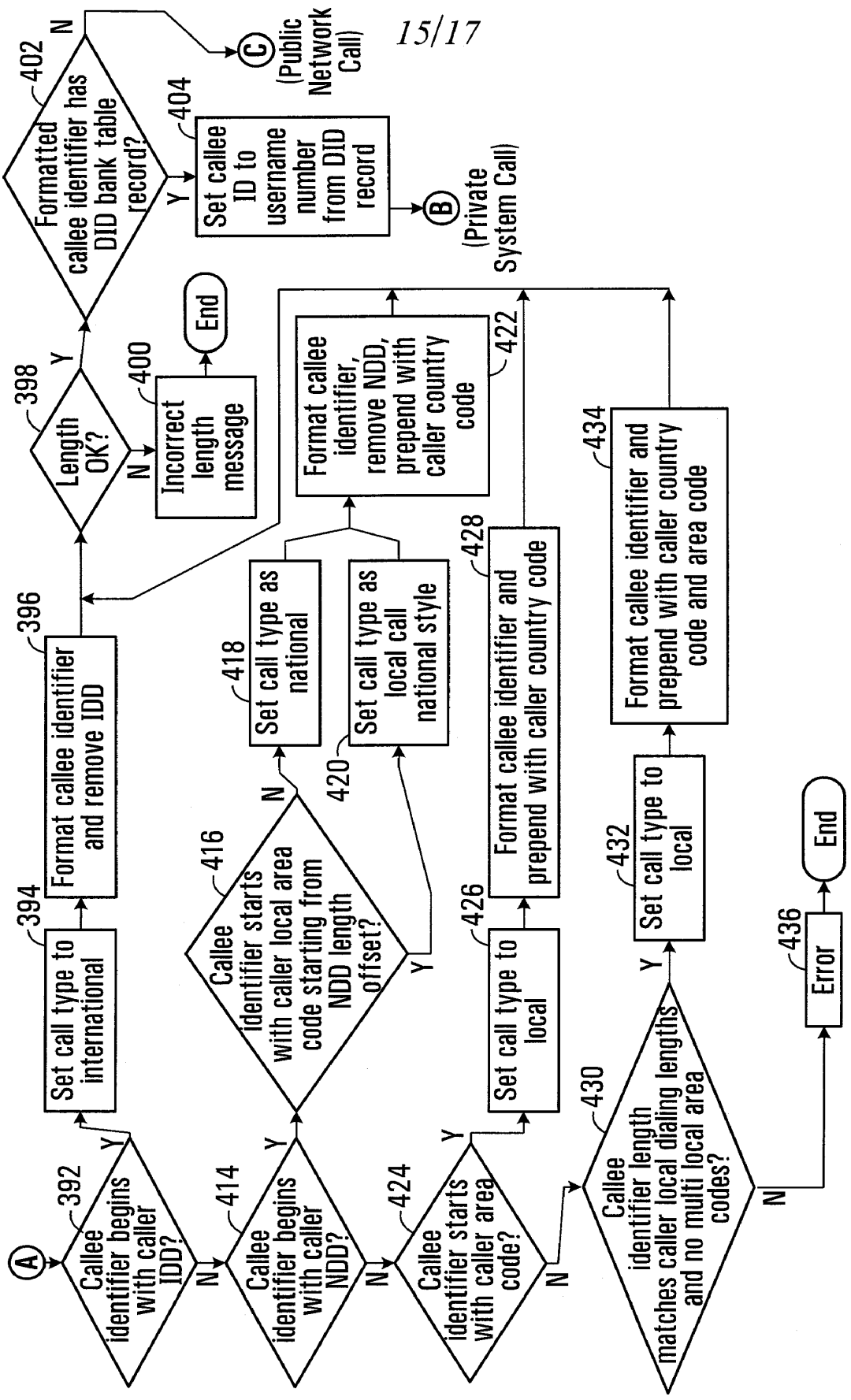

A
Callee identifier begins with caller IDD?
392
Y
N
Callee identifier begins with caller NDD?
414
Y
N
Callee identifier starts with caller area code?
424
Y
N
Callee identifier length matches caller local dialing lengths and no multi local area codes?
430
Y
N
Error
436
End
Set call type to international
394
Format callee identifier and remove IDD
396
Callee identifier starts with caller local area code starting from NDD length offset?
416
Y
N
Set call type as local call national style
420
Set call type as national
418
Set call type to local
426
Format callee identifier and prepend with caller country code
428
Set call type to local
432
Format callee identifier and prepend with caller country code and area code
434
Length OK?
398
Y
N
Incorrect length message
400
End
Format callee identifier, remove NDD, prepend with caller country code
422
Formatted callee identifier has DID bank table record?
402
Y
N
Set callee ID to username number from DID record
404
B
(Private System Call)
C
(Public Network Call)

FIG. 18B

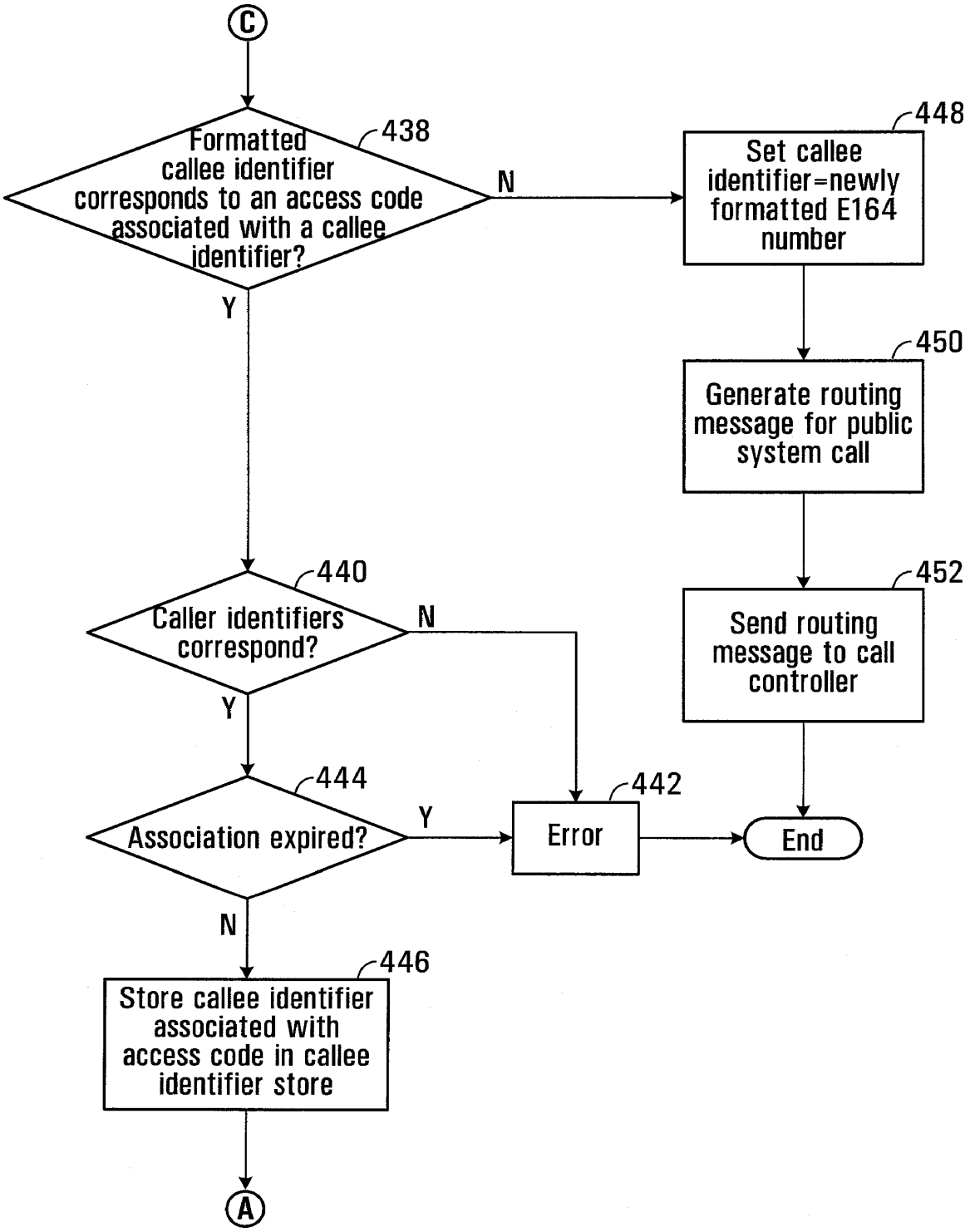

C
Formatted callee identifier corresponds to an access code associated with a callee identifier?
438
N
Y
448
Set callee identifier=newly formatted E164 number
450
Generate routing message for public system call
452
Send routing message to call controller
End
440
Caller identifiers correspond?
N
Y
444
Association expired?
Y
442
Error
N
446
Store callee identifier associated with access code in callee identifier store
A

FIG. 18C

17/17

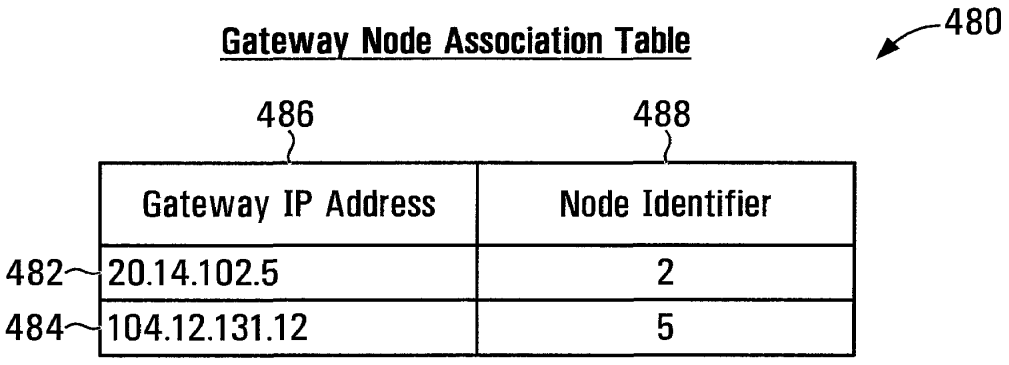


FIG. 19

META 1002
META V. VOIP-PAL

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

U.S. APPLICATION NUMBER NO.	FIRST NAMED APPLICANT	ATTY. DOCKET NO.
13/056,277	Johan Emil Viktor BJORSELL	SMARB19.004APC

20995
KNOBBE MARTENS OLSON & BEAR LLP
2040 MAIN STREET
FOURTEENTH FLOOR
IRVINE, CA 92614

INTERNATIONAL APPLICATION NO.	
PCT/CA2009/001062	
I.A. FILING DATE	PRIORITY DATE
07/28/2009	07/28/2008

CONFIRMATION NO. 2403
371 FORMALITIES LETTER

OC000000046061096

Date Mailed: 02/18/2011

Notification of Insufficient Fees (DO/EO/US)

The following items have been submitted by the applicant or the IB to the United States Patent and Trademark Office as a Designated Office (37 CFR 1.494):

- Indication of Small Entity Status
- Priority Document
- Copy of the International Application filed on 01/27/2011
- Copy of the International Search Report filed on 01/27/2011
- Preliminary Amendments filed on 01/27/2011
- Information Disclosure Statements filed on 01/27/2011
- Oath or Declaration filed on 01/27/2011
- U.S. Basic National Fees filed on 01/27/2011
- Priority Documents filed on 01/27/2011
- Power of Attorney filed on 01/27/2011
- Specification filed on 01/27/2011
- Claims filed on 01/27/2011
- Abstracts filed on 01/27/2011
- Drawings filed on 01/27/2011

The following items **MUST** be furnished within the period set forth below in order to complete the requirements for acceptance under 35 U.S.C. 371:

- Additional claim fees of $**2370** as a small entity, including any required multiple dependent claim fee, are required. Applicant must submit the additional claim fees or cancel the additional claims for which fees are due.

SUMMARY OF FEES DUE:

Total additional fees required for this application is $**2370** for a Small Entity:

Total additional claim fee(s) for this application is $**2370**
- $**550** for **5** independent claims over 3.
- $**1820** for **70** total claims over 20.

ALL OF THE ITEMS SET FORTH ABOVE MUST BE SUBMITTED WITHIN TWO (2) MONTHS FROM THE DATE OF THIS NOTICE OR BY 32 MONTHS FROM THE PRIORITY DATE FOR THE APPLICATION, WHICHEVER IS LATER. FAILURE TO PROPERLY RESPOND WILL RESULT IN ABANDONMENT.

FORM PCT/DO/EO/923 (371 Formalities Notice)

The time period set above may be extended by filing a petition and fee for extension of time under the provisions of 37 CFR 1.136(a).

Applicant is reminded that any communications to the United States Patent and Trademark Office must be mailed to the address given in the heading and include the U.S. application no. shown above (37 CFR 1.5)

Registered users of EFS-Web may alternatively submit their reply to this notice via EFS-Web.
https://sportal.uspto.gov/authenticate/AuthenticateUserLocalEPF.html

For more information about EFS-Web please call the USPTO Electronic Business Center at **1-866-217-9197** or visit our website at http://www.uspto.gov/ebc.

If you are not using EFS-Web to submit your reply, you must include a copy of this notice.

VONDA M WALLACE

Telephone: (571) 272-3734

FORM PCT/DO/EO/923 (371 Formalities Notice)

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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
13/056,277	01/27/2011		490	SMARB19.004APC	90	8

CONFIRMATION NO. 2403

20995
KNOBBE MARTENS OLSON & BEAR LLP
2040 MAIN STREET
FOURTEENTH FLOOR
IRVINE, CA 92614

FILING RECEIPT

OC000000046061095

Date Mailed: 02/18/2011

Receipt is acknowledged of this non-provisional patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. **If an error is noted on this Filing Receipt, please submit a written request for a Filing Receipt Correction. Please provide a copy of this Filing Receipt with the changes noted thereon. If you received a "Notice to File Missing Parts" for this application, please submit any corrections to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections**

Applicant(s)

Johan Emil Viktor BJORSELL, Vancouver, CANADA;
Maksym SOBOLYEV, New Westminster, CANADA;
Pentti Kalevi HUTTUNEN, Vancouver, CANADA;

Assignment For Published Patent Application

DIGIFONICA (INTERNATIONAL) LIMITED, Vancouver ,BC, CANADA

Power of Attorney: The patent practitioners associated with Customer Number 20995

Domestic Priority data as claimed by applicant

This application is a 371 of PCT/CA2009/001062 07/28/2009
which claims benefit of 61/129,898 07/28/2008

Foreign Applications (You may be eligible to benefit from the **Patent Prosecution Highway** program at the USPTO. Please see http://www.uspto.gov for more information.)

If Required, Foreign Filing License Granted: 02/16/2011

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 13/056,277**

Projected Publication Date: 05/26/2011

Non-Publication Request: No

Early Publication Request: No
**** SMALL ENTITY ****

Title

MOBILE GATEWAY

Preliminary Class

PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

Since the rights granted by a U.S. patent extend only throughout the territory of the United States and have no effect in a foreign country, an inventor who wishes patent protection in another country must apply for a patent in a specific country or in regional patent offices. Applicants may wish to consider the filing of an international application under the Patent Cooperation Treaty (PCT). An international (PCT) application generally has the same effect as a regular national patent application in each PCT-member country. The PCT process **simplifies** the filing of patent applications on the same invention in member countries, but **does not result** in a grant of "an international patent" and does not eliminate the need of applicants to file additional documents and fees in countries where patent protection is desired.

Almost every country has its own patent law, and a person desiring a patent in a particular country must make an application for patent in that country in accordance with its particular laws. Since the laws of many countries differ in various respects from the patent law of the United States, applicants are advised to seek guidance from specific foreign countries to ensure that patent rights are not lost prematurely.

Applicants also are advised that in the case of inventions made in the United States, the Director of the USPTO must issue a license before applicants can apply for a patent in a foreign country. The filing of a U.S. patent application serves as a request for a foreign filing license. The application's filing receipt contains further information and guidance as to the status of applicant's license for foreign filing.

Applicants may wish to consult the USPTO booklet, "General Information Concerning Patents" (specifically, the section entitled "Treaties and Foreign Patents") for more information on timeframes and deadlines for filing foreign patent applications. The guide is available either by contacting the USPTO Contact Center at 800-786-9199, or it can be viewed on the USPTO website at http://www.uspto.gov/web/offices/pac/doc/general/index.html.

For information on preventing theft of your intellectual property (patents, trademarks and copyrights), you may wish to consult the U.S. Government website, http://www.stopfakes.gov. Part of a Department of Commerce initiative, this website includes self-help "toolkits" giving innovators guidance on how to protect intellectual property in specific countries such as China, Korea and Mexico. For questions regarding patent enforcement issues, applicants may call the U.S. Government hotline at 1-866-999-HALT (1-866-999-4158).

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Title 35, United States Code, Section 184

Title 37, Code of Federal Regulations, 5.11 & 5.15

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The applicant has been granted a license under 35 U.S.C. 184, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" followed by a date appears on this form. Such licenses are issued in all applications where the conditions for issuance of a license have been met, regardless of whether or not a license may be required as set forth in 37 CFR 5.15. The scope and limitations of this license are set forth in 37 CFR 5.15(a) unless an earlier

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NOT GRANTED

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PATENT APPLICATION FEE DETERMINATION RECORD

Substitute for Form PTO-875

Application or Docket Number: 13/056,277

APPLICATION AS FILED - PART I

FOR	(Column 1) NUMBER FILED	(Column 2) NUMBER EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY FEE($)
BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	165		N/A	
SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	215		N/A	
EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	110		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	90 minus 20 =	* 70	x 26 =	1820	OR		
INDEPENDENT CLAIMS (37 CFR 1.16(h))	8 minus 3 =	* 5	x 110 =	550			
APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $270 ($135 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			0.00			
MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				0.00			
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	2860		TOTAL	

APPLICATION AS AMENDED - PART II

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY ADDITIONAL FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY ADDITIONAL FEE($)
AMENDMENT A	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY ADDITIONAL FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY ADDITIONAL FEE($)
AMENDMENT B	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest found in the appropriate box in column 1.

MULTIPLE DEPENDENT CLAIM FEE CALCULATION SHEET Substitute for Form PTO-1360 (For use with Form PTO/SB/06)	Application Number 13056277	Filing Date
	Applicant(s) Johan BJORSELL	
	* May be used for additional claims or amendments	

CLAIMS	AS FILED		AFTER FIRST AMENDMENT		AFTER SECOND AMENDMENT	
	Indep	Depend	Indep	Depend	Indep	Depend
1	1		1			
2		1		1		
3		1		1		
4		1		1		
5		1		1		
6		1		1		
7		1		1		
8		1		1		
9		1		1		
10		1		1		
11		10		1		
12	1		1			
13		1		1		
14		1		1		
15		1		1		
16		1		1		
17		1		1		
18		1		1		
19		1		1		
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47		1		1		
48		1		1		
49		1		1		
50		1		1		
Total Indep	8		8		0	
Total Depend	107	↵	82	↵	0	↵
Total Claims	115		90		0	

	*		*		*	
	Indep	Depend	Indep	Depend	Indep	Depend
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87		1		1		
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90	1		1			
91						
92						
93						
94						
95						
96						
97						
98						
99						
100						

Response to Missing Requirements Under 35 USC § 371	Attorney Docket No.: SMARB19.004APC First Named Inventor: Johan Emil Viktor Bjorsell et al. Int'l Application No.: PCT/CA2009/001062 US Application No.: 13/056277 Entered National Phase: January 27, 2011 Title: MOBILE GATEWAY
Direct all correspondence to Customer # 20995	Page 1 of 1

Mail Stop PCT
United States Patent and Trademark Office
PO Box 1450
Alexandria, VA 22313-1450

Applicant herewith submits to the US Designated/Elected Office (DO/EO/US) the following items and other information:

(X) This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 USC 371.

(X) A supplemental Declaration signed by the inventors (35 USC 371(c)(4)) in 3 pages.

FILING FEES NOT YET PAID:

FEE CALCULATION				
FEE TYPE		**LARGE FEE**	**CALCULATION**	**TOTAL**
Excess Claims	90 - 20 = 70	1615 ($52)	70 x 52 =	$3,640
Excess Ind. Claims	8 - 3 = 5	1614 ($220)	5 x 220 =	$1,100
			SUB TOTAL	$4,740
The present application qualifies for Small Entity status under 37 CFR § 1.27. **Fee reduced by 1/2.**				($2,370)
			TOTAL FEE DUE	**$2,370**

(X) Fees will be paid via EFS Web. Extension of time is requested by payment of any extension fee.

The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. 11-1410.

John M. Carson
Registration No. 34,303
Attorney of Record
Customer No. 20995
(858) 836-9000

11005522\cey\040511

META 1002
META V. VOIP-PAL

DECLARATION FOR UTILITY OR DESIGN APPLICATION UNDER 37 CFR 1.63

Docket No.: SMARB19.004APC

Title: MOBILE GATEWAY

Inventors: Johan Emil Viktor Bjorsell, Maksym Sobolyev, and Pentti Kalevi Huttunen

Please Direct All Correspondence to Customer Number 20995

This Declaration is directed to the invention described and claimed the application that was filed on July 28, 2009, was assigned International App. No. PCT/CA2009/001062, entered the national phase on January 27, 2011, and was further assigned U.S. Patent App. No. 13/056277.

As a below named inventor:

I believe the inventors named below to be the original and first inventors of the subject matter which is described and claimed and for which a patent is sought;

I have reviewed and understand the contents of the above-identified application, including the claims; and

I acknowledge the duty to disclose information which is material to patentability as defined in 37 CFR 1.56.

The application was originally filed with an Application Data Sheet (ADS). The ADS sets forth any applicable Foreign Priority Claims under 35 USC § 119, and sets forth the full mailing address and residence of each inventor whose signature appears below as allowed under 37 CFR 1.63(c). The ADS also sets forth any Domestic Priority Claims under 35 USC §§ 119(e), 120, 121, and 365.

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful, false statements may jeopardize the validity of the application or any patent issued thereon.

Full name of **first** inventor: Johan Emil Viktor Bjorsell

Signature: _____ Date: 23 March 2011

Citizenship: SE

META 1002
META V. VOIP-PAL

DECLARATION FOR UTILITY OR DESIGN APPLICATION UNDER 37 CFR 1.63

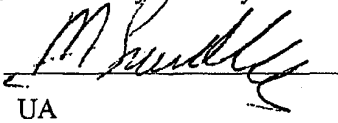
Docket No.: SMARB19.004APC

Title: MOBILE GATEWAY

Inventors: Johan Emil Viktor Bjorsell, Maksym Sobolyev, and Pentti Kalevi Huttunen

Please Direct All Correspondence to Customer Number 20995

Full name of **second** inventor: Maksym Sobolyev

Signature:  Date: 22 March 2011

Citizenship: UA

META 1002
META V. VOIP-PAL

DECLARATION FOR UTILITY OR DESIGN APPLICATION UNDER 37 CFR 1.63

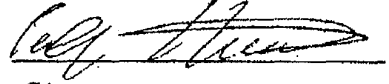
Docket No.: SMARB19.004APC

Title: MOBILE GATEWAY

Inventors: Johan Emil Viktor Bjorsell, Maksym Sobolyev, and Pentti Kalevi Huttunen

Please Direct All Correspondence to Customer Number 20995

Full name of **third** inventor: Pentti Kalevi Huttunen

Signature:  Date: March 23, 2011

Citizenship: CA

Send Correspondence To:
KNOBBE, MARTENS, OLSON & BEAR, LLP
Customer No. 20995

10791959
022811

Electronic Patent Application Fee Transmittal

Application Number:	13056277
Filing Date:	27-Jan-2011
Title of Invention:	MOBILE GATEWAY
First Named Inventor/Applicant Name:	Johan Emil Viktor BJORSELL
Filer:	John M Carson/Caity Yaussi
Attorney Docket Number:	SMARB19.004APC

Filed as Small Entity

U.S. National Stage under 35 USC 371 Filing Fees

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Basic Filing:				
Pages:				
Claims:				
Claims in excess of 20	2615	70	26	1820
Independent claims in excess of 3	2614	5	110	550
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD($)
Extension-of-Time:				
Miscellaneous:				
		Total in USD ($)		**2370**

Electronic Acknowledgement Receipt	
EFS ID:	9885020
Application Number:	13056277
International Application Number:	
Confirmation Number:	2403
Title of Invention:	MOBILE GATEWAY
First Named Inventor/Applicant Name:	Johan Emil Viktor BJORSELL
Customer Number:	20995
Filer:	John M Carson/ThuyQuyen Nguyen
Filer Authorized By:	John M Carson
Attorney Docket Number:	SMARB19.004APC
Receipt Date:	14-APR-2011
Filing Date:	27-JAN-2011
Time Stamp:	19:01:37
Application Type:	U.S. National Stage under 35 USC 371

Payment information:

Submitted with Payment	yes
Payment Type	Credit Card
Payment was successfully received in RAM	$2370
RAM confirmation Number	8757
Deposit Account	111410
Authorized User	KNOBBE MARTENS OLSON AND BEAR
The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows: Charge any Additional Fees required under 37 C.F.R. 1.492 (National application filing, search, and examination fees) Charge any Additional Fees required under 37 C.F.R. Section 1.17 (Patent application and reexamination processing fees)	

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1		SMARB19_004APC.pdf	125357 e8b60a02ce1e8dd04a831dcb70e07f448f2298ff	yes	4
	Multipart Description/PDF files in .zip description				
	Document Description		**Start**	**End**	
	Applicant Response to Pre-Exam Formalities Notice		1	1	
	Oath or Declaration filed		2	4	
Warnings:					
Information:					
2	Fee Worksheet (PTO-875)	fee-info.pdf	31863 1868ccc4b308496fdb55e95956d0aa0fa52dbc84	no	2
Warnings:					
Information:					
		Total Files Size (in bytes):	157220		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111
If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371
If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office
If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY.DOCKET.NO	TOT CLAIMS	IND CLAIMS
13/056,277	01/27/2011	2473	2860	SMARB19.004APC	90	8

CONFIRMATION NO. 2403

UPDATED FILING RECEIPT

20995
KNOBBE MARTENS OLSON & BEAR LLP
2040 MAIN STREET
FOURTEENTH FLOOR
IRVINE, CA 92614

OC000000047287594

Date Mailed: 04/25/2011

Receipt is acknowledged of this non-provisional patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. **If an error is noted on this Filing Receipt, please submit a written request for a Filing Receipt Correction. Please provide a copy of this Filing Receipt with the changes noted thereon. If you received a "Notice to File Missing Parts" for this application, please submit any corrections to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections**

Applicant(s)

Johan Emil Viktor Bjorsell, Vancouver, BC, CANADA;
Maksym Sobolyev, New Westminster, BC, CANADA;
Pentti Kalevi Huttunen, Vancouver, BC, CANADA;

Assignment For Published Patent Application

DIGIFONICA (INTERNATIONAL) LIMITED, Vancouver ,BC, CANADA

Power of Attorney: The patent practitioners associated with Customer Number 20995

Domestic Priority data as claimed by applicant

This application is a 371 of PCT/CA2009/001062 07/28/2009
which claims benefit of 61/129,898 07/28/2008

Foreign Applications (You may be eligible to benefit from the **Patent Prosecution Highway** program at the USPTO. Please see http://www.uspto.gov for more information.)

If Required, Foreign Filing License Granted: 02/16/2011

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 13/056,277**

Projected Publication Date: 05/26/2011

Non-Publication Request: No

Early Publication Request: No
**** SMALL ENTITY ****

Title

MOBILE GATEWAY

Preliminary Class

370

PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

Since the rights granted by a U.S. patent extend only throughout the territory of the United States and have no effect in a foreign country, an inventor who wishes patent protection in another country must apply for a patent in a specific country or in regional patent offices. Applicants may wish to consider the filing of an international application under the Patent Cooperation Treaty (PCT). An international (PCT) application generally has the same effect as a regular national patent application in each PCT-member country. The PCT process **simplifies** the filing of patent applications on the same invention in member countries, but **does not result** in a grant of "an international patent" and does not eliminate the need of applicants to file additional documents and fees in countries where patent protection is desired.

Almost every country has its own patent law, and a person desiring a patent in a particular country must make an application for patent in that country in accordance with its particular laws. Since the laws of many countries differ in various respects from the patent law of the United States, applicants are advised to seek guidance from specific foreign countries to ensure that patent rights are not lost prematurely.

Applicants also are advised that in the case of inventions made in the United States, the Director of the USPTO must issue a license before applicants can apply for a patent in a foreign country. The filing of a U.S. patent application serves as a request for a foreign filing license. The application's filing receipt contains further information and guidance as to the status of applicant's license for foreign filing.

Applicants may wish to consult the USPTO booklet, "General Information Concerning Patents" (specifically, the section entitled "Treaties and Foreign Patents") for more information on timeframes and deadlines for filing foreign patent applications. The guide is available either by contacting the USPTO Contact Center at 800-786-9199, or it can be viewed on the USPTO website at http://www.uspto.gov/web/offices/pac/doc/general/index.html.

For information on preventing theft of your intellectual property (patents, trademarks and copyrights), you may wish to consult the U.S. Government website, http://www.stopfakes.gov. Part of a Department of Commerce initiative, this website includes self-help "toolkits" giving innovators guidance on how to protect intellectual property in specific countries such as China, Korea and Mexico. For questions regarding patent enforcement issues, applicants may call the U.S. Government hotline at 1-866-999-HALT (1-866-999-4158).

LICENSE FOR FOREIGN FILING UNDER

Title 35, United States Code, Section 184

Title 37, Code of Federal Regulations, 5.11 & 5.15

GRANTED

The applicant has been granted a license under 35 U.S.C. 184, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" followed by a date appears on this form. Such licenses are issued in all applications where the conditions for issuance of a license have been met, regardless of whether or not a license may be required as

set forth in 37 CFR 5.15. The scope and limitations of this license are set forth in 37 CFR 5.15(a) unless an earlier license has been issued under 37 CFR 5.15(b). The license is subject to revocation upon written notification. The date indicated is the effective date of the license, unless an earlier license of similar scope has been granted under 37 CFR 5.13 or 5.14.

This license is to be retained by the licensee and may be used at any time on or after the effective date thereof unless it is revoked. This license is automatically transferred to any related applications(s) filed under 37 CFR 1.53(d). This license is not retroactive.

The grant of a license does not in any way lessen the responsibility of a licensee for the security of the subject matter as imposed by any Government contract or the provisions of existing laws relating to espionage and the national security or the export of technical data. Licensees should apprise themselves of current regulations especially with respect to certain countries, of other agencies, particularly the Office of Defense Trade Controls, Department of State (with respect to Arms, Munitions and Implements of War (22 CFR 121-128)); the Bureau of Industry and Security, Department of Commerce (15 CFR parts 730-774); the Office of Foreign AssetsControl, Department of Treasury (31 CFR Parts 500+) and the Department of Energy.

NOT GRANTED

No license under 35 U.S.C. 184 has been granted at this time, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" DOES NOT appear on this form. Applicant may still petition for a license under 37 CFR 5.12, if a license is desired before the expiration of 6 months from the filing date of the application. If 6 months has lapsed from the filing date of this application and the licensee has not received any indication of a secrecy order under 35 U.S.C. 181, the licensee may foreign file the application pursuant to 37 CFR 5.15(b).

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

U.S. APPLICATION NUMBER NO.	FIRST NAMED APPLICANT	ATTY. DOCKET NO.
13/056,277	Johan Emil Viktor Bjorsell	SMARB19.004APC

20995
KNOBBE MARTENS OLSON & BEAR LLP
2040 MAIN STREET
FOURTEENTH FLOOR
IRVINE, CA 92614

INTERNATIONAL APPLICATION NO.	
PCT/CA2009/001062	
I.A. FILING DATE	PRIORITY DATE
07/28/2009	07/28/2008

CONFIRMATION NO. 2403
371 ACCEPTANCE LETTER

OC000000047287595

Date Mailed: 04/25/2011

NOTICE OF ACCEPTANCE OF APPLICATION UNDER 35 U.S.C 371 AND 37 CFR 1.495

The applicant is hereby advised that the United States Patent and Trademark Office in its capacity as a Designated / Elected Office (37 CFR 1.495), has determined that the above identified international application has met the requirements of 35 U.S.C. 371, and is ACCEPTED for national patentability examination in the United States Patent and Trademark Office.

The United States Application Number assigned to the application is shown above and the relevant dates are:

01/27/2011	01/28/2011
DATE OF RECEIPT OF 35 U.S.C. 371(c)(1), (c)(2) and (c)(4) REQUIREMENTS	DATE OF COMPLETION OF ALL 35 U.S.C. 371 REQUIREMENTS

A Filing Receipt (PTO-103X) will be issued for the present application in due course. **THE DATE APPEARING ON THE FILING RECEIPT AS THE " FILING DATE" IS THE DATE ON WHICH THE LAST OF THE 35 U.S.C. 371 (c)(1), (c)(2) and (c)(4) REQUIREMENTS HAS BEEN RECEIVED IN THE OFFICE. THIS DATE IS SHOWN ABOVE.** *The filing date of the above identified application is the international filing date of the international application (Article 11(3) and 35 U.S.C. 363).* Once the Filing Receipt has been received, send all correspondence to the Group Art Unit designated thereon.

The following items have been received:

- Indication of Small Entity Status
- Copy of the International Application filed on 01/27/2011
- Copy of the International Search Report filed on 01/27/2011
- Preliminary Amendments filed on 01/27/2011
- Information Disclosure Statements filed on 01/27/2011
- Oath or Declaration filed on 01/27/2011
- U.S. Basic National Fees filed on 01/27/2011
- Priority Documents filed on 01/27/2011
- Power of Attorney filed on 01/27/2011
- Specification filed on 01/27/2011
- Claims filed on 01/27/2011
- Abstracts filed on 01/27/2011
- Drawings filed on 01/27/2011

FORM PCT/DO/EO/903 (371 Acceptance Notice)

Applicant is reminded that any communications to the United States Patent and Trademark Office must be mailed to the address given in the heading and include the U.S. application no. shown above (37 CFR 1.5)

VONDA M WALLACE

Telephone: (571) 272-3734

FORM PCT/DO/EO/903 (371 Acceptance Notice)

PATENT APPLICATION FEE DETERMINATION RECORD

Substitute for Form PTO-875

Application or Docket Number: 13/056,277

APPLICATION AS FILED - PART I

FOR	(Column 1) NUMBER FILED	(Column 2) NUMBER EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY FEE($)
BASIC FEE (37 CFR 1.16(a), (b), or (c))	N/A	N/A	N/A	165		N/A	
SEARCH FEE (37 CFR 1.16(k), (i), or (m))	N/A	N/A	N/A	215		N/A	
EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))	N/A	N/A	N/A	110		N/A	
TOTAL CLAIMS (37 CFR 1.16(i))	90 minus 20 =	* 70	x 26 =	1820	OR		
INDEPENDENT CLAIMS (37 CFR 1.16(h))	8 minus 3 =	* 5	x 110 =	550			
APPLICATION SIZE FEE (37 CFR 1.16(s))	If the specification and drawings exceed 100 sheets of paper, the application size fee due is $270 ($135 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).			0.00			
MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))				0.00			
* If the difference in column 1 is less than zero, enter "0" in column 2.			TOTAL	2860		TOTAL	

APPLICATION AS AMENDED - PART II

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY ADDITIONAL FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY ADDITIONAL FEE($)
AMENDMENT A	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

		(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE($)	SMALL ENTITY ADDITIONAL FEE($)	OR	OTHER THAN SMALL ENTITY RATE($)	OTHER THAN SMALL ENTITY ADDITIONAL FEE($)
AMENDMENT B	Total (37 CFR 1.16(i))	*	Minus	**	=	x =		OR	x =	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	x =		OR	x =	
	Application Size Fee (37 CFR 1.16(s))									
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))							OR		
						TOTAL ADD'L FEE		OR	TOTAL ADD'L FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest found in the appropriate box in column 1.

INFORMATION DISCLOSURE STATEMENT BY APPLICANT	Application No.	13/056277
	Filing Date	01-27-2011
	First Named Inventor	Johan Emil Viktor Björsell et al
	Art Unit	2473
(Multiple sheets used when necessary)	Examiner	Unassigned
SHEET 1 OF 5	Attorney Docket No.	SMARB19.004APC

U.S. PATENT DOCUMENTS					
Examiner Initials	Cite No.	Document Number *Number - Kind Code (if known)* Example: 1,234,567 B1	Publication Date MM-DD-YYYY	Name of Patentee or Applicant	Pages, Columns, Lines Where Relevant Passages or Relevant Figures Appear
	1	4,916,491	04-10-1990	Katoh	
	2	5,146,491	09-08-1992	Silver et al.	
	3	5,247,571	09-21-1993	Kay et al .	
	4	5,303,297	04-12-1994	Hillis	
	5	5,359,642	10-25-1994	Castro	
	6	5,425,085	06-13-1995	Weinberger et al.	
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	8	5,469,497	11-21-1995	Pierce et al.	
	9	5,506,893	04-09-1996	Buscher et al.	
	10	5,519,769	05-21-1996	Weinberger et al.	
	11	5,559,871	09-24-1996	Smith	
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	13	5,608,786	05-04-1997	Gordon	
	14	5,621,787	04-15-1997	McKoy et al.	
	15	5,661,790	08-26-1997	Hsu	
	16	5,712,907	01-27-1998	Wegner et al.	
	17	5,724,355	05-03-1998	Bruno et al.	
	18	5,726,984	05-10-1998	Kubler et al.	
	19	5,737,414	04-07-1998	Walker et al.	
	20	5,751,961	05-12-1998	Smyk	
	21	5,793,762	08-11-1998	Penners et al.	
	22	5,799,072	08-25-1998	Vulcan et al.	
	23	5,802,502	09-01-1998	Gell et al.	
	24	5,825,863	10-20-1998	Walker	
	25	5,828,740	10-27-1998	Khuc et al.	
	26	5,838,682	11-17-1998	Dekelbaum et al.	
	27	5,845,267	12-01-1998	Ronen	
	28	5,850,433	12-15-1998	Rondeau	
	29	5,864,610	01-26-1999	Ronen	

Examiner Signature	Date Considered
***Examiner:** Initial if reference considered, whether or not citation is in conformance with MPEP 609. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.	

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT	Application No.	13/056277
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	First Named Inventor	Johan Emil Viktor Björsell et al
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SHEET 2 OF 5	Attorney Docket No.	SMARB19.004APC

U.S. PATENT DOCUMENTS					
Examiner Initials	Cite No.	Document Number *Number - Kind Code (if known)* Example: 1,234,567 B1	Publication Date MM-DD-YYYY	Name of Patentee or Applicant	Pages, Columns, Lines Where Relevant Passages or Relevant Figures Appear
	30	5,867,495	02-02-1999	Elliott et al.	
	31	5,883,891	05-16-1999	Williams et al.	
	32	5,889,774	05-30-1999	Mirashrafi et al.	
	33	5,905,736	05-18-1999	Ronen et al.	
	34	5,907,547	05-25-1999	Foladare et al.	
	35	5,910,946	06-08-1999	Csapo	
	36	5,915,005	06-22-1999	He	
	37	5,923,659	01-30-2001	Curry et al.	
	38	5,930,343	07-27-1999	Vasquez	
	39	5,937,045	08-10-1999	Yaoya et al.	
	40	5,940,598	08-17-1999	Strauss et al.	
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	42	5,956,391	09-21-1999	Melen et al.	
	43	5,970,477	10-19-1999	Roden	
	44	5,974,043	10-26-1999	Solomon	
	45	5,991,291	11-23-1999	Asai et al .	
	46	6,005,926	12-21-1999	Mashinsky	
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	49	6,052,445	10-28-2003	Bashoura et al.	
	50	6,058,300	05-02-2000	Hanson	
	51	6,069,890	05-30-2000	White et al.	
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	53	6,104,704	08-15-2000	Buhler et al.	
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Examiner Signature	Date Considered
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	First Named Inventor	Johan Emil Viktor Björsell et al
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SHEET 3 OF 5	Attorney Docket No.	SMARB19.004APC

U.S. PATENT DOCUMENTS					
Examiner Initials	Cite No.	Document Number *Number - Kind Code (if known)* Example: 1,234,567 B1	Publication Date MM-DD-YYYY	Name of Patentee or Applicant	Pages, Columns, Lines Where Relevant Passages or Relevant Figures Appear
	59	6,188,752	02-13-2001	Lesley	
	60	6,282,574	08-28-2001	Voit	
	61	6,298,062	10-02-2001	Gardell et al.	
	62	6,351,464	02-26-2002	Galvin et al.	
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	64	6,430,275	08-06-2002	Voit et al	
	65	6,507,644	01-14-2003	Henderson et al.	
	66	6,766,159	07-20-2004	Lindholm	
	67	6,819,929	11-16-2004	Antonucci et al.	
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	69	7,068,772	06-27-2006	Widger et al.	
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	71	7,212,522	05-01-2007	Shankar et al.	
	72	7,454,510	11-18-2008	Kleyman et al.	
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	77	2004/0165709 A1	08-26-2004	Pence et al.	
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	86	2006/0072547 A1	04-06-2006	Florkey et al.	
	87	2006/0111116 A1	05-25-2006	Palmer et al.	

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INFORMATION DISCLOSURE STATEMENT BY APPLICANT	Application No.	13/056277
	Filing Date	01-27-2011
	First Named Inventor	Johan Emil Viktor Björsell et al
	Art Unit	2473
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SHEET 4 OF 5	Attorney Docket No.	SMARB19.004APC

U.S. PATENT DOCUMENTS					
Examiner Initials	Cite No.	Document Number *Number - Kind Code (if known)* Example: 1,234,567 B1	Publication Date MM-DD-YYYY	Name of Patentee or Applicant	Pages, Columns, Lines Where Relevant Passages or Relevant Figures Appear
	88	2006/0160565 A1	07-20-2006	Singh et al.	
	89	2006/0177035 A1	08-10-2006	Cope et al.	
	90	2006/0209768 A1	09-21-2006	Yan et al.	
	91	2007/0036143 A1	02-15-2007	Alt et al.	
	92	2007/0253418	11-01-2007	Shiri et al.	
	93	2008/0037715 A1	02-14-2008	Prozeniuk et al.	
	94	2008/0063153 A1	03-13-2008	Krivorot et al.	
	95	2009/0028146 A1	01-29-2009	Kleyman et al.	

FOREIGN PATENT DOCUMENTS						
Examiner Initials	Cite No.	Foreign Patent Document *Country Code-Number-Kind Code* Example: JP 1234567 A1	Publication Date MM-DD-YYYY	Name of Patentee or Applicant	Pages, Columns, Lines Where Relevant Passages or Relevant Figures Appear	T[1]
	96	CA 2,249,668	04-07-1999	Bruno et al.		
	97	EP 1 389 862 A1	02-18-2004	Shen et al.		
	98	WO 2007/044454 A2	04-19-2007	Croy et al.		
	99	WO 2008/052340 A1	05-08-2008	Perreault et al.		
	100	WO 2008/064481 A1	06-05-2008	Bjorsell et al.		
	101	WO 2008/116296 A1	10-02-2008	Bjorsell et al.		

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	102	A copy of the International Preliminary Report on Patentability and Written Opinion of the International Searching Authority mailed February 10, 2011 for related PCT Application No. PCT/CA2009/001062., February 10, 2011.	
	103	A copy of the International Preliminary Report on Patentability mailed February 13, 2009 for corresponding PCT/CA2007/002150.	
	104	A copy of the International Preliminary Report on Patentability mailed May 14, 2009 for related PCT/CA2007/001956.	
	105	A copy of the International Search Report and Written Opinion of the International Searching Authority completed February 6, 2008 for corresponding PCT/CA2007/001956.	
	106	A copy of the International Search Report and Written Opinion of the International Searching Authority completed June 6, 2008 for corresponding PCT/CA2008/000545.	

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	107	A copy of the International Search Report completed on March 3, 2008 for corresponding PCT/CA2007/002150.	
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	111	F. Baker et al. "RFC 3924 - Cisco Architecture for Lawful Intercept in IP Networks." October 2004.	
	112	M. Handley et al. "RFC 2543 - SIP: Session Initiation Protocol." March 1999.	
	113	Sippy SIP B2BUA. "About Sippy RTPproxy." http://www.rtpproxy.org. July 15, 2009.	

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Electronic Acknowledgement Receipt

EFS ID:	10024105
Application Number:	13056277
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Confirmation Number:	2403
Title of Invention:	MOBILE GATEWAY
First Named Inventor/Applicant Name:	Johan Emil Viktor Bjorsell
Customer Number:	20995
Filer:	John M Carson/Alexandra Benitez
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Attorney Docket Number:	SMARB19.004APC
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Application Type:	U.S. National Stage under 35 USC 371

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Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Foreign Reference	CA_2249668.pdf	1272336 093d0424de15364b3d993028ea9c35262cd1596f	no	40
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2	Foreign Reference	EP_1389862.pdf	551343 938583fb2be7f59e4a66cff10aadc2f1b708fdcb	no	11
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3	Foreign Reference	WO_07_044454.pdf	995614 2386d2e6624a65bb511e22fcfce256b8d1335f55	no	27
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4	Foreign Reference	WO_08_052340.pdf	4872564 084a9402effde4a1b20cb79a3ba4d05e3793fc0a	no	138
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6	Foreign Reference	WO_08_116296.pdf	3390565 299c609e3cdf1084bb99f5f27c4f4798a19b283a	no	89
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9	NPL Documents	Baker.pdf	666996 7c3acfab99c9bb403b2cbc0fd8176e2987b6eca6	no	17
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10	NPL Documents	Handley.pdf	5635414 65a910242ad994baf7a1100b3ebb8e858ec138ee	no	143
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12	NPL Documents	IPRP_001062.pdf	293201 dcd77431b47d1b19e6453c616531c29a8801cfbb	no	7
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13	NPL Documents	IPRP_002150.pdf	1138415 ac7f2a930f363ff846c404dfb423e748360f67b3	no	27
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14	NPL Documents	IPRP_001956.pdf	535190 50b674114b078f2c665d9e060a0eb18335273e69	no	10
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15	NPL Documents	ISR_WO_000545.pdf	465629 73e1c60478b6c0911685c4bc4120f9798f2a74b7	no	9
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16	NPL Documents	ISR_WO_001317.pdf	323665 4f6d39f37ec65a31c3062c9be2f84cee455e374c	no	7
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17	NPL Documents	ISR_WO_001956.pdf	703721 f9277a31dd1e5e1f8593ab3faa963fa6ef100cb5	no	12
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If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

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(11) CA 2 249 668 (13) C

(40) 23.12.2003
(43) 07.04.1999
(45) 23.12.2003

(12)

(21) 2 249 668

(51) Int. Cl.[6]: H04L 12/56, H04L 12/46

(22) 05.10.1998

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BRUNO, RICHARD FRANK (US).
MARKOWITZ, ROBERT EDWARD (US).
ROBINSON, BETHANY SCOTT (US).
KATSEFF, HOWARD PAUL (US).

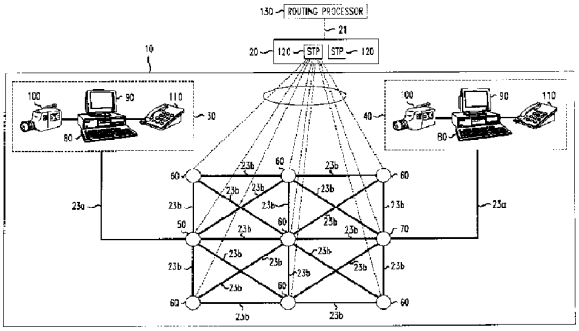
(74) KIRBY EADES GALE BAKER

(54) METHODES ET SYSTEMES DE GESTION DYNAMIQUE DU ROUTAGE D'INFORMATION SUR UN RESEAU UNIVERSEL INTEGRE DE COMMUNICATION

(54) METHODS AND SYSTEMS FOR DYNAMICALLY MANAGING THE ROUTING OF INFORMATION OVER AN INTEGRATED GLOBAL COMMUNICATION NETWORK

(57)

Methods and systems for routing information in an integrated global communications network comprising a signalling system interconnecting a source router, a plurality of intermediate routers, a destination router and a routing processor for routing the multimedia information. The source router receives the information including the routing requirements associated with the information and transmits a routing query signal to the routing processor. The routing processor evaluates the routing requirements, determines which routers and communication paths are capable and available to route the information, evaluates the statistical availability of such routers and selects an optimal routing path comprising select ones of the available routers and communication paths.



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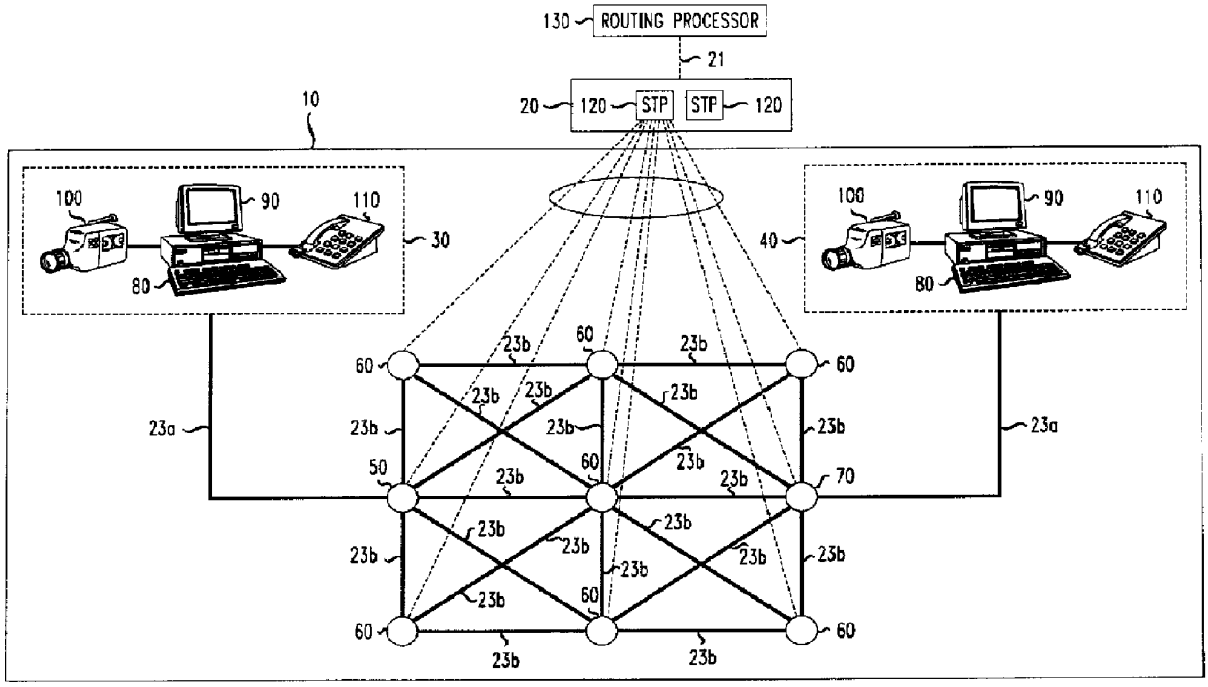
(72) Inventeurs/Inventors:
BRUNO, RICHARD FRANK, US;
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ROSEN, KENNETH H., US

(73) Propriétaire/Owner:
AT&T CORP., US

(74) Agent: KIRBY EADES GALE BAKER

(54) Titre : METHODES ET SYSTEMES DE GESTION DYNAMIQUE DU ROUTAGE D'INFORMATION SUR UN RESEAU UNIVERSEL INTEGRE DE COMMUNICATION

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(57) Abrégé/Abstract:
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(11)(21) 2 249 668

(13) C

(57) Abrégé(suite)/Abstract(continued):
information and transmits a routing query signal to the routing processor. The routing processor evaluates the routing requirements, determines which routers and communication paths are capable and available to route the information, evaluates the statistical availability of such routers and selects an optimal routing path comprising select ones of the available routers and communication paths.

Methods And Systems For Dynamically Managing The Routing Of Information Over An Integrated Global Communication Network

ABSTRACT

Methods and systems for routing information in an integrated global communications network comprising a signalling system interconnecting a source router, a plurality of intermediate routers, a destination router and a routing processor for routing the multimedia information. The source router receives the information including the routing requirements associated with the information and transmits a routing query signal to the routing processor. The routing processor evaluates the routing requirements, determines which routers and communication paths are capable and available to route the information, evaluates the statistical availability of such routers and selects an optimal routing path comprising select ones of the available routers and communication paths.

METHODS AND SYSTEMS FOR DYNAMICALLY MANAGING THE ROUTING OF INFORMATION OVER AN INTEGRATED GLOBAL COMMUNICATION NETWORK

FIELD OF THE INVENTION

The present invention relates generally to the field of managing the routing of information over an integrated global communication network, such as the Internet, and, more particularly, to the methods and systems for reserving routing paths for routing information over the network.

BACKGROUND OF THE INVENTION

As is well known, the Internet is a vast collection of computers that communicate over a packet network via high-speed communication paths ranging from ISDN to T1, T3, FDDI, SONET, SMDS, OT1, etc. A personal computer typically accesses the Internet through a modem on a user's "Plain Old Telephone Service" (POTS) line or through a switched ISDN. Alternatively, a personal computer can access the Internet through a X.25, Frame Relay (FR) or Asynchronous Transfer Mode (ATM) connection on a high-speed local area network (LAN) or wide area network (WAN). Connecting a personnel computer to the LAN requires a card known as a LAN adapter that plugs into the computer's expansion bus. Once Internet access is

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established, the Internet communicates information from the source computer to a destination over a routing path using any one of a number of protocols, such as the Internet Protocol (IP).

With the increased routing of multimedia (voice, video and data) calls over the Internet, there continues to be a growing demand for modifications to the routing component of the Internet infrastructure to support real-time quality transmission of information. For example, video applications have characteristics including file size, flow rate and sensitivity to delay that distinguish them from other applications that share the same network. A single video file can be hundreds of megabytes or gigabytes long, whereas traditional data is significantly shorter. Further, a traditional data application is sent over a network as a burst of packets, whereas a video application is sent as a continuous stream of data which must be delivered quickly and regularly. When packetized, transmission delays must not vary by more than a few milliseconds. Data that arrives too late is discarded resulting in diminished quality of display. Moreover, the frames occupy space in queues, delaying the rest of the traffic and possibly causing such traffic to be discarded as well. Thus, the key requirement is to reduce the end-to-end packet delays (e.g. jitter) in order to satisfy real-time delivery constraints and achieve the necessary high nodal throughput (e.g. message latency) for the transfer of real-time voice and video.

Currently, the management of routing between the Internet and

the various routers are commonly performed over a signalling network, such as Resource Reservation Protocol (RSVP), which visits routers in the network and attempts to make a resource reservation to obtain routes that are capable of routing particular multimedia information.

The current Internet routing structure, however, lacks the ability to provide advanced reservation of optimal routing paths for such real-time multimedia information. Such advanced reservation is desirable in reducing jitter and achieving high message latency in the transmission of such information. It is therefore desirable to improve the routing of multimedia information through the integrated global communication network such that the desired transmission is achieved.

SUMMARY OF THE INVENTION

In one aspect, the invention features a system for managing the routing of information to a destination through a packet network that includes a plurality of routers, wherein each router in the packet network is linked to at least one other router by a first communication medium. The system comprises a routing processor for receiving a routing query specifying a destination to which the information will be routed and a memory for storing at least one characteristic of each of the routers in the packet network, wherein the characteristic includes statistical information. The processor has direct access to the characteristic information of the routers and determines a route for the transmission of the information based on the routing query and on the characteristics stored in the memory, wherein the route comprises at least two of the routers. The system also comprises a second communication medium for transmitting signals from the processor to the router, wherein the second

communication medium differs from the first communication medium.

In another aspect of the invention, the invention features a system for routing information to a destination, which comprises a packet network and a routing processor. The packet network includes a plurality of routers, wherein each router is linked to at least one other router by a communication medium. The routing processor receives a routing query signal from a first one of the routers and information concerning at least one characteristic of each of the remaining routers. The routing processor then determines a transmission path for routing the information through the packet network based on the routing query and on the characteristics stored in the memory, wherein the route comprises at least two of the routers.

In accordance with one aspect of the present invention there is provided a system for managing the routing of information to a destination through a packet network that includes a plurality of routers, each router being linked to at least one other router by a first communication medium, said system comprising: a routing processor for receiving a routing query specifying a destination to which said information will be routed; a memory for storing at least one characteristic of each of said routers in said packet network, said characteristic including statistical information; wherein said processor has direct access to said characteristic of said routers and determines a route for the transmission of said information based on said routing query and on said characteristics stored in said memory, said route comprising at least two of said routers; and a second communication medium for transmitting signals from said processor to said routers, wherein the second communication medium differs from the first communication medium.

In accordance with another aspect of the present invention there is provided a method for managing the routing of information to a destination through a packet network, wherein said network includes a plurality of routers, each router being linked to at least one other router by a first communication medium, said method comprising the steps of: receiving a routing query specifying a destination to which said information will be routed at a routing processor; storing at least one characteristic of each of said routers, said characteristic including statistical information, wherein said characteristic of said routers may be directly accessed by said processor; determining a route for the transmission of said information based on said routing query and on said stored characteristics, said route comprising at least two of the routers; and transmitting said route from said processor to said routers via a second communication medium, wherein the second communication medium differs from the first communication medium.

In yet another aspect of the invention, the invention features a method for managing the routing of information to a destination in a packet network that includes a plurality of routers, wherein each router is linked to at least one other router by a communication medium. The method comprises the steps of receiving a routing query signal from a first one of the routers and information concerning at least one characteristic of each of the remaining routers and determining a transmission path for routing the information through the packet network. The transmission path comprises at least one router in addition to the first router. Such routing can be implemented for both Internet and Intranet traffic.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram illustrating a system for managing the routing of information over an integrated global communication network in accordance with one embodiment of the present invention;

FIG. 2 is a functional block diagram of a method for managing the routing of information over an integrated global communication network using the system of FIG. 1;

FIG. 3 is a block diagram illustrating an embodiment of the routing processor of FIG. 1;

FIG. 4 is a block diagram illustrating an embodiment of the

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routers of FIG 1.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

FIG. 1 shows a simplified integrated global data communication network suitable for use in accordance with an embodiment of the present invention. It will of course be recognized that the network of FIG. 1 includes other known elements, but those elements have been omitted for simplicity.

Referring to FIG. 1, there is illustrated an integrated global communication network **10**, such as the Internet, that uses a signalling system **20** for communicating messages. The network **10** includes plural multimedia terminals **30** and **40**, communication paths **23a** and **23b**, a source router **50**, a plurality of intermediate routers **60** and a destination router **70**. Of course, an actual network could include many more such terminals, paths and routers.

As shown, multimedia terminals **30** and **40** each include a processing unit **80**, a CRT **90**, and a camera **100**. The terminals **30** and **40** are used merely as an example and may include a variety of other communication devices, including but not limited to an associated telephone **110** located external to each of the processing units **80** for purposes of dialing another terminal's telephone number when the processing units **80** are incapable of doing so directly. Because of the numerous types of multimedia capable devices which may be utilized in connection with the invention, the integrated global communication network **10** may also include additional processing equipment (not shown) at the destination, for example, a cable converter,

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satellite transmission converter or personal computer may be necessary to convert a message to a format compatible with that destination. A more detailed general background on multimedia communication, and particularly, desktop videoconferencing and collaborative video equipment may be found in Don Labriola, Meeting on the Edge, Windows Sources, Sept. 1994, p. 96 et seq.

Each of the processing units **80** may be a general purpose computer with multimedia capable equipment and/or packetized voice hardware and/or software incorporated therein. A more detailed description of the construction and operation of the Vistium video system may be found in Andrew W. Davis, VISTIUM: AT&T's Board-Level Videoconferencing at the Desktop, Advanced Imaging, Sept. 1994, p. 46 et seq. Alternatively, the processing unit may be any multimedia specific device.

The signalling system **20** interconnects the various components of the network **10**, including source router **50**, intermediate routers **60**, and destination router **70** to routing processor **130**. Preferably, the signalling system **20** used by the network is an SS7 network of interconnected STPs **120**. As is well known, the SS7 network utilizes a SS7 protocol which consists of four basic sub-protocols. These sub-protocols are: message transfer part (MTP) which provides the function of basic routing of signalling messages between signalling points; a signalling connection control part (SCCP) which

provides additional routing and management functions for transfer of messages other than call set-up between the signaling points; an integrated services digital network (ISDN) user part (ISUP) which transfers call set-up signaling information between signaling points; and a transaction capabilities part (TCAP) which transfers non-circuit related information between the signaling points.

The signalling system **20** illustrated in FIG. 1 includes the preferred SS7 network of interconnected STPs **120**. Those skilled in the art will recognize that STP **120** is a multi-port, high speed packet switch that directs traffic among entities on a network. It should be noted that signal transfer point devices such as STP **120** are conventionally installed in redundant pairs within a network, as illustrated, so that if one device fails, its mate takes over until the first STP **120** is able to return to service.

As is illustrated in FIG. 1, STP **120** is connected to routing processor **130** by communication path **21** and the source router **50**, the intermediate routers **60**, and the destination router **70** are connected to STP **120** by communication path **22**. It should be noted that separate paths are provided for communication paths **21** and **22**, that communicate routing message signals, and communication paths **23a** and **23b**, that actually communicate the information. Communication paths **21** and **22** are preferably high digital serial A-links, which transfer routing message signals between elements **50**, **60**, **70**, **120**, and **130** using out-of-band signaling with other

communications protocols. It is understood that communication paths **21** and **22** are logical paths, although the communication paths could be physical paths.

After the signalling system **20** completes sending the routing message signals via STP **120** through communication paths **21** and **22**, communication paths **23a** and **23b** transfer the multimedia information from terminal **30** to terminal **40** through a routing path including source router **50**, select intermediate routers **60**, and destination router **70** over communication paths **23a** and select communication paths **23b**. In most instances, the multimedia information will be routed from multimedia terminal **30** to source router **50** and destination router **70** to terminal **40**, through communication paths **23a** via an LEC (not shown) in a public switched network or via a private switched network (not shown) and through intermediate routers **60** via communication paths **23b**. Communication paths **23a** and **23b** comprise digital links, which are preferably high speed (1.544 Mbps) T-1 span over which information is transmitted using in-band-signaling in a serial fashion. Other known communication paths, besides the preferred T-1 links, which are also suitable for use in conjunction with this invention include, for example, Feature Group D Data Trunks (sometimes referred to as "FG-D"), as well as ATM, FR, ISDN BRI, ISDN PRI, T1, Switched 56 (SW56), 45Kbps DS-3/Datapath, 56 Kbps DDS/Datapath, or nxDS-0 paths.

It will be recognized that the types and combinations of

communication paths **23a** and **23b** which may be employed are too numerous to discuss in detail. It should therefore be recognized that the preferred embodiment will work with communication paths **23a** and **23b** that are capable of supporting packetized voice and/or a combination of single or multi-channel video, audio or data requiring different sizes of bandwidth. By way of example, where the path is an ISDN path it should support about 64 Kbps access for combinations of single channel video, audio and data or about 128 Kbps access for combinations of two channel video, audio and data. Where the path is T1 it should support about 1.5 Mbps access for combinations of single channel video, audio and data or about 3 Mbps access for combinations of two channel video, audio and data. Where the path is switched 56 it should support about 56 Kbps access for combinations of single channel video, audio and data or about 112 Kbps access for combinations of two channel video, audio and data. Where the path is DS-3 it should support about 45 Kbps access for combinations of single channel video, audio and data or about 90 Kbps access for combinations of two channel video, audio and data. Where the path is nxDS-0, the path should support n x 64 Kbps access, where n is the number of channels.

In the embodiment illustrated in FIG. 1, the source router **50**, intermediate routers **60**, and destination router **70** could be any number of conventional routers capable of routing information over the network **10** and processing signalling messages over signalling system **20**. Similarly, the

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routing processor **130** is illustratively shown as and may be implemented using a 2NCP processor also available from Lucent Technologies Inc., but could otherwise be any processing means capable of processing signalling messages communicated over the signalling system **20**. The routing processor **130** is utilized to control routing of multimedia (i.e, real-time video or audio) information associated with a network subscriber.

While only a specific number of intermediate routers **60**, STPs **120** and routing processor **130** are illustrated in FIG. 1, it is understood that the communication network **10** may include additional routers, STPs and/or routing processors to process and complete such multimedia information over the network **10**. It is also understood that various trunks and other channels may be provided in the network **10** to connect, for instance, two or more routers to one another. It is further understood that various modifications may be made to the network **10** without departing from the scope of the invention.

Referring to FIG. 2, there is illustrated a block flowchart of the preferred method of using the system illustrated in FIG. 1, in which routing processor **130** controls the routing of multimedia information from multimedia terminal **30** to multimedia terminal **40** in network **10**.

In step **200**, source router **50** begins receiving a plurality of information packets that form a multimedia session from multimedia terminal **30**. Preferably, the packets are sent to source router **50** using Internet IP addressing and UDP protocol. Upon receiving the first packets, which include

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the source and destination addresses and routing requirement information (e.g. bandwidth and time limitations on information transfer), the source router **50** sends a flow control ON message to the multimedia terminal **30** to hold further transmission of the remaining information packets that form the session. As is known in the art, upon receiving the packet, the source router **50** translates the destination network address to a physical address in the Internet network.

In step **201**, the source router **50** sends a routing query to the service processor **130** via STP **120**. The routing query includes the source and destination addresses as well as routing requirements (e.g. bandwidth requirements). The routing query is preferably a routing TCAP query message.

In step **202**, the routing processor **130** evaluates the list of intermediate routers **60** and corresponding communication paths **23b**, in Router Table **140** and Inventory Control Table **150**, and identifies intermediate routers **60** (including any backup or secondary intermediate routers) and communication paths **23b** connected therebetween that have the appropriate bandwidth capability to route particular multimedia information.

In step **203**, the routing processor **130** identifies each intermediate router **60** in the list of intermediate routers **60** and corresponding communication paths **23b**, previously identified by the routing processor **130** in step **202**, that is capable of processing the information to determine which routers are currently available.

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Specifically, in step **203**, the routing processor **130** uses a pointer in a list of intermediate routers identified by the routing processor **130** to consider each router. The routing processor **130** determines whether the router, on the list to which the pointer is directed, is available. The routing processor **130** will then point to the "next" router in the list and repeat the process until the routing processor **130** has checked all of the routers in the list to ascertain their respective availability conditions. Once all of the routers in the list have been considered and the pointer is incremented back to the first router then the routing processor **130** proceeds to step **204**.

If the routing processor **130** determines in step **203** that all or some of the intermediate routers **60** in the list of routers **60** capable of routing particular multimedia information are available, then the routing processor **130** provides such router **60** and communication path **23b** information to its optimal route program **160** for determining the optimal routing path of intermediate routers **60** and communication paths **23b** between the source router **50** and the destination router **70** and its statistical program **170** for determining the frequency of use of such intermediate routers **60** (step **204**).

The optimal routing program **160**, discussed in more detail below in the description of FIG. 3, provides intelligence or decision making capabilities based upon stored programs and data which is available either in the **140** or Inventory Control Table **150** (see FIG. 3) or obtained from other data sources such as local dynamic routing databases **225** in individual routers

60 to calculate the optimal route. The optimal routing program **160** may for example be the Real-Time Network Routing (RTNR) program described in Ash et al. United States Patent Number 5,101,451.

The statistical program **170**, discussed in more detail below in the description of FIG. 3, utilizes the status information stored in the Inventory Control Table **150** to maintain a statistical record of performance of the system. The statistical program **160** and the optimal routing program **170** are both used by the routing processor **130** in determining several routing paths including an optimal routing path (step **204**).

After the routing paths are determined, the routing processor **130** selects a routing path (preferably, the optimal routing path, discussed in more detail below in the description of the optimal routing program **160** of FIG. 3) and the routing processor **130** retrieves from the Router Table **140** the SS7 network address of the desired routers **60** (step **206**). The routing processor **130** also creates and transmits "Reserve" messages, preferably SS7 messages, instructing each of the intermediate routers **60** on this path to reserve a routing path including communication paths **23b** with the specific bandwidth and length capabilities to provide the requested quality of service. The intermediate routers **60** then send "Acknowledgement" messages to routing processor **130** (step **206**).

Routing processor **130** transmits a "Proceed" message to the

source router **50** identifying the reserved router path and instructing the source router **50** to proceed to route the information within the network **10** (step **207**).

Source router **50** sends a flow control OFF message to terminal **30** instructing terminal **30** to continue to transmit the remaining information packets (step **208**). Terminal **30** then sends the remaining packets to source router **50** which forwards such packets to destination router **70** through intermediate router(s) **60** and communication paths **23b** on the reserved router path. (step **209**). In the preferred embodiment, the reserved routing path forms a permanent virtual connection, which will remain up even if the session is disconnected in error.

When terminal **40** receives all the information, either terminal **30** or terminal **40** transmit a "Disconnect" message to source router **50** or destination router **70**, respectively. (step **210**). Router **50** or **70** then forward this "Disconnect" message to all intermediate routers **60** on the reserved routing path, thereby disconnecting the session and ensuring that the session status is correct. (step **210**).

The routing processor **130** then updates its **140** and Inventory Control Table **150** changing the status of the intermediate routers **60** and communication paths **23b** in its lists from unavailable to spare. (step **211**).

Preferably, the routing processor then sends this information to the reserved intermediate routers **60** which update their local dynamic routing databases **225** from unavailable to spare. (step **212**).

Referring to FIG. 3, there is illustrated an embodiment of the routing processor **130** as shown in FIG. 1. The routing processor **130** includes an optimal routing program **160** and statistical program **170** each comprising a microprocessing unit (MPU) and each operating under program control as supported by a ROM, a RAM, an input keyboard, an output display monitor, and an input/output interface device which provide communication capability, i.e. transmission and reception, of information and commands to other elements of the routing processor **130** such as the **140** and Inventory Control Table **150** and routers **60** such as local dynamic database **225**. The MPU is also supported by a non-volatile data storage device, such as a hard disk, which provides general storage of program controlled instructions and data which is utilized in decision making carried out by the optimal routing program **160** and statistical program **170**. Each of the storage devices can be considered as storing data associated with different databases, Router Table **140** and Inventory Control Table **150**. The routing processor **130** also includes an input/output module **190** which accommodates communication with the illustrated source router **50**, intermediate routers **60** and destination router **70** via STPs **120**.

The Router Table **140**, in the illustrative embodiment is preferably stored within a database of the routing processor **130** and is utilized by the routing processor **130** to determine the SS7 network routing information associated with one or more intermediate routers **60** capable of

routing the information in accordance with the present invention. The Router Table **140** illustrated in FIG. 3 preferably comprises at least three columns. For example, the first column of the Router Table **140** identifies the routers **50**, **60**, **70** and corresponding communication paths **23a** and **23b**. The second column of the Router Table **140** contains each router Internet address and corresponding SS7 network address. Various router SS7 network addresses may be utilized to reach a router **60** capable of routing the information. The third column of the Router Table **140** contains the status of each router (e.g., available). The status of the router SS7 address is determined dynamically based on its SS7 network routing status.

The Inventory Control Table **150**, in the illustrative embodiment, stores logical, physical, and statistical information relating to intermediate router(s) **60** and communication paths **23b** (e.g. bandwidth and distance between routers **60**). For example, assuming the total bandwidth for the T-1 transport sections of intermediate router communication paths **23b** is 15.44 Mbps and that the bandwidth is divided into chunks of 8 Kbps. This bandwidth segment information is placed in the Inventory Control Table **150**.

The Inventory Control Table **150** also stores information relating to the availability and bandwidth consumption status of intermediate routers **60** and communication paths **23b** such routers **60**. One feature of the preferred embodiment is the capability of routing information based on current router availability. In order to minimize multimedia information-processing

time, it is desirable to have real-time status information concerning the availability of a router to accept incoming information. As used herein, "real-time" information means information available in a short time, e.g., a few seconds, as opposed to longer delays. Such real-time information concerning each of routers **60** and communication paths **23b** availability is preferably stored in the Inventory Control Table **150**. Preferably, the higher speed communication paths **21** and **22** are utilized in order to provide the most current real-time status of each router **60** and communication path **23b** for utilization by routing processor **130**. Specifically, the stored information includes records based on time of day routing, day of week routing, location routing, bandwidth consumption routing, and alternate routing in cases of busy conditions.

The Inventory Control Table **150** preferably, also contains data which reflects the availability of each router and contains corresponding time data. For example, this database information can permit routers to be selected based upon the router that has been available for the longest time.

The optimal routing program **160**, in the illustrative embodiment, preferably determines the optimum routing path required to minimize the number of routing hops involved in completing the multimedia call across the network and thus, results in reduced jitter. For example, assuming that each router introduces 50 milliseconds of delay, the optimal routing path used to transport voice over the integrated global communication

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network should include no more than three router hops. Further, in computing the optimal path, the optimal routing program **160** takes into consideration the bandwidth required for the call and selects routers and associated communication paths that are capable of providing the best high nodal throughput for the call. For example, assuming the call requires bandwidth of about 1.5 Mbps the optimization will consider all routers **60** and associated communication paths **23b**, identified in the routing processor's **130** Routing Table **140** and Inventory Control Table **150**, with this bandwidth capability and then determine the optimal path utilizing select ones of these routers **60** and associated communication paths **23b**.

The statistical program **170**, in the illustrative embodiment, reviews the components of the network (i.e. has a global view of the network and the routers and the corresponding communication paths included therein) stored in the routing processor's **130** Routing Table **140** and Inventory Control Table **150** and determines the percentage of time each router is available or busy. The statistical program **170** also determines the percentage of time the corresponding communication paths are available to provide the requested bandwidth. Such statistical calculations are computed at different times of days and on different days of the week and the results of such calculations are then stored in the Inventory Control Table **150**.

Referring to FIG. 4, there is illustrated a block diagram of an embodiment of routers **50**, **60**, and **70** illustrated in FIG. 1. For discussion

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purposes, the router is marked as an intermediate router **60**. Router **60** includes a packet scheduler **215** to hold information packets in a queue, a routing protocol **220** to store handling instructions and decision making processes for routing information based on the specific routing requirements, a local dynamic routing data base **225** to store individual router information including bandwidth consumption and communication path **23b** information and a local route selection **230** which is used by the router in determining which communication paths **23b** are capable of routing multimedia information. Router **60** routes information over communication paths, such as communication path **23b**, as discussed above. These existing routers can be modified, however, to also include input/output module (I/O) **190** to accommodate communication of routing message signals between such routers and the routing processor **130** through communication paths **22** via STPs **120**.

Although an illustrative preferred embodiment has been described herein in detail, it should be noted and will be appreciated by those skilled in the art that numerous variations may be made within the scope of this invention without departing from the principle of this invention and without sacrificing its chief advantages. For instance, it is understood that, while the present invention has been described above with respect to "SS7" messaging over a signalling system, other types of signalling systems may be utilized to interconnect and communicate signalling messages between the

various components of the integrated global communication network. The terms and expressions have been used herein as terms of description and not terms of limitation. There is no intention to use the terms or expressions to exclude any equivalents of features shown and described or portions thereof and this invention should be defined in accordance with the claims which follow.

CLAIMS:

1. A system for managing the routing of information to a destination through a packet network that includes a plurality of routers, each router being linked to at least one other router by a first communication medium, said system comprising:

a routing processor for receiving a routing query specifying a destination to which said information will be routed;

a memory for storing at least one characteristic of each of said routers in said packet network, said characteristic including statistical information:

wherein said processor has direct access to said characteristic of said routers and determines a route for the transmission of said information based on said routing query and on said characteristics stored in said memory, said route comprising at least two of said routers; and

a second communication medium for transmitting signals from said processor to said routers, wherein the second communication medium differs from the first communication medium.

2. The system according to claim 1, wherein said characteristic includes information relating to the bandwidth of said communication medium.

3. The system according to claim 1, wherein said characteristic includes statistical information relating to the availability of each of said routers.

4. The system according to claim 3, wherein said statistical information includes information relating to the availability of each of said routers at specific times of day.

5. The system according to claim 3, wherein said statistical information includes information relating to the availability of each of said routers on specific days of the week.

6. A system for routing information to a destination, said system comprising:

a packet network that includes a plurality of routers, wherein each router is linked to at least one other router by a first communication medium;

a routing processor for receiving a routing query signal from a first one of said routers and information concerning at least one characteristic of each of said remaining routers, said characteristic including statistical information, said routing processor determining a transmission path for routing said information through said packet network, said transmission path comprising at least one router in addition to said first router;

wherein said routing processor has direct access to said characteristic of said routers and determines said transmission path based on said routing query signal and on said received characteristics; and

a second communication medium for transmitting signals from said processor to said routers, wherein the second communication medium differs from the first communication medium.

7. The system according to claim 6, wherein said routing processor transmits reservation signals to each of said routers on said transmission path.

8. The system according to claim 6, wherein said characteristic includes at least one address for each of said routers.

9. The system according to claim 6, wherein said characteristic includes information relating to the bandwidth of said communication medium.

10. The system according to claim 6, wherein said characteristic includes statistical information relating to the availability of each of said router.

11. The system according to claim 10, wherein said statistical information includes information relating to the availability of each of said routers at specific times of day.

12. The system according to claim 10, wherein said statistical information includes information relating to the availability of each of said routers on specific days of the week.

13. The system according to claim 6, wherein said routing processor includes a program for calculating statistical information about each of said routers based on characteristics stored in memory.

14. A system for managing the routing of information to a destination, said system comprising:

a packet network that includes a plurality of routers, wherein each router is linked to at least one other router by a first communication medium; and

a router processor in communication with said routers through a signalling system, wherein said signalling system transmits signals using a second communication medium that differs from the first communication medium;

wherein said routing processor receives a routing query signal from a first one of said routers and information concerning at least one characteristic of each of said remaining routers through said signalling system, said characteristic including statistical information, said routing processor determining a transmission path for routing said information through said packet network, said transmission path comprising at least one router in addition to said first router;

wherein said routing processor has direct access to said characteristic of said routers and determines said transmission path based on said routing query signal and on said received characteristics.

15. The system according to claim 14, wherein said routing processor transmits reservation signals to each of said routers on said transmission path.

16. The system according to claim 14, wherein said characteristic includes at least one address for each of said routers.

17. The system according to claim 14, wherein said characteristic includes information relating to the bandwidth of said communication medium.

18. The system according to claim 14, wherein said characteristic includes statistical information relating to the availability of each of said routers.

19. The system according to claim 18, wherein said statistical information includes information relating to the availability of each of said routers at specific times of day.

20. The system according to claim 18, wherein said statistical information includes information relating to the availability of each of said routers on specific days of the week.

21. The system according to claim 14, wherein said routing processor includes a program for calculating statistical information about each of said routers based on characteristics stored in memory.

22. The system according to claim 14, wherein said signalling system utilizes SS7 messaging capabilities.

23. The system according to claim 14, wherein said signalling system includes at least one signal transfer point.

24. A system for managing the routing of multimedia information to a destination, said system comprising:

a packet network that includes a plurality of routers, wherein each router is linked to at least one other router by a first communication medium; and

a routing processor in communication with said routers through a signalling system, wherein said signalling system transmits signals using a second communication medium that differs from the first communication medium;

wherein said routing processor receives a routing query signal including routing requirements from a first one of said routers and information concerning at least one characteristic of each of said remaining routers through said signalling system, said characteristic including statistical information, said router processor identifies a plurality of routers that are capable of routing said information, said routing processor has direct access to said characteristic of

said routers and determines a transmission path for routing said information based on said routing query and on said characteristic, through said packet network, said transmission path comprising at least one router in addition to said first router.

25. The system according to claim 24, wherein said routing processor transmits reservation signals to each of said routers on said transmission path.

26. The system according to claim 24, wherein said signalling system utilizes SS7 messaging capabilities.

27. The system according to claim 24, wherein said signalling system includes at least one signal transfer point.

28. A method for managing the routing of information to a destination through a packet network, wherein said network includes a plurality of routers, each router being linked to at least one other router by a first communication medium, said method comprising the steps of:

receiving a routing query specifying a destination to which said information will be routed at a routing processor;

storing at least one characteristic of each of said routers, said characteristic including statistical information, wherein said characteristic of said routers may be directly accessed by said processor;

determining a route for the transmission of said information based on said routing query and on said stored characteristics, said route comprising at least two of the routers; and

transmitting said route from said processor to said routers via a second communication medium, wherein the second communication medium differs from the first communication medium.

29. The method according to claim 28, wherein said step of storing characteristics includes the step of storing at least one address for each of said routers.

30. The method according to claim 28, wherein said step of storing characteristics includes the step of storing information relating to the bandwidth of said communication medium.

31. The method according to claim 28, wherein said step of storing characteristics includes the step of storing statistical information relating to the availability of each of said routers.

32. The method according to claim 31, wherein said step of storing statistical information further includes the step of storing information relating to the availability of each of said routers at specific times of day.

33. The method according to claim 31, wherein said step of storing statistical information further includes the step of storing information relating to the availability of each of said routers on specific days of the week.

34. A method for managing the routing of information to a destination in a packet network that includes a plurality of routers, wherein each router is linked to at least one other router by a first communication medium, said method comprising the steps of:

receiving a routing query signal from a first one of said routers and information concerning at least one characteristic of each of said remaining routers, said characteristic including statistical information, wherein said characteristic of said routers may be directly accessed by said processor;

determining a transmission path for routing said information through said packet network, said transmission path comprising at least one

router in addition to said first router, wherein said step of determining is based on said received routing query and on said received characteristics; and

transmitting said transmission path from said processor to said routers via a second communication medium, wherein the second communication medium differs from the first communication medium.

35. The method according to claim 34, further including the step of transmitting reservation signals to each of said routers on said transmission path.

36. The method according to claim 34, wherein said step of determining a transmission path includes the step of evaluating the bandwidth of said communication medium.

37. The method according to claim 34, wherein said step of determining a transmission path includes the step of evaluating the availability of each of said routers.

38. The method according to claim 34, wherein the step of determining a transmission path includes the step of calculating statistical information about each of said routers based on characteristics stored in memory.

39. The method according to claim 38, wherein said step of calculating statistical information includes the step of evaluating the availability of each of said routers at specific times of day.

40. The method according to claim 38, wherein said step of calculating statistical information includes the step of evaluating the availability of each of said routers on specific days of the week.

41. A method for managing the routing of information to a destination in a packet network that includes a plurality of routers, wherein each router is linked to at least one other router by a first communication medium, said method comprising the steps of:

receiving a routing query signal including routing requirements from a first one of said routers and information concerning at least one characteristic of each of said remaining router at a routing processor over a signalling system, wherein said signalling system transmits signals using a second communication medium that differs from the first communication medium, said characteristic including statistical information, wherein said characteristic of said routers may be directly accessed by said processor;

identifying a plurality of routers that are capable of routing said information;

determining the availability of said identified routers; and

determining a transmission path for routing said information through said packet network based on said routing query and on said characteristic, wherein said transmission path comprises at least one of said identified routers in addition to said first router.

42. The method according to claim 41, wherein said step of determining a transmission path includes the step of evaluating the bandwidth of said communication medium.

43. The method according to claim 41, wherein said step of determining a transmission path includes the step of evaluating the availability of each of said routers.

44. The method according to claim 41, wherein the step of determining a transmission path includes the step of calculating statistical information about each of said routers based on characteristics stored in memory.

45. The method according to claim 44, wherein said step of calculating statistical information includes the step of evaluating the availability of each of said routers at specific times of day.

46. The method according to claim 44, wherein said step of calculating statistical information includes the step of evaluating the availability of each of said routers on specific days of the week.

47. A method for managing the routing of multimedia information to a destination in a packet network that includes a plurality of routers, wherein each router is linked to at least one other router by a first communication medium, said method comprising the steps of:

receiving a routing query signal including routing requirements from a first one of said routers and information concerning at least one characteristic of each of said remaining router at a routing processor over a signalling system, wherein said signalling system transmits signals using a second communication medium that differs from the first communication medium, said characteristic including statistical information, wherein said characteristic of said routers may be directly accessed by said processor;

identifying a plurality of routers that are capable of routing said information;

determining the availability of each of said identified routers;

determining a plurality of transmission paths for routing said information through said packet network based on said routing query and on said characteristic, each of said transmission paths comprising at least one of said identified routers in addition to said first router, wherein said step of determining each of said transmission paths is based on said routing query;

selecting one of said transmission paths for routing said information through said packet network;

sending reservation signals from said routing processor to said routers on said select routing path over said signalling system;

sending a routing response signal from said routing processor to said first router over said signalling system; and

routing said multimedia information over said select path.

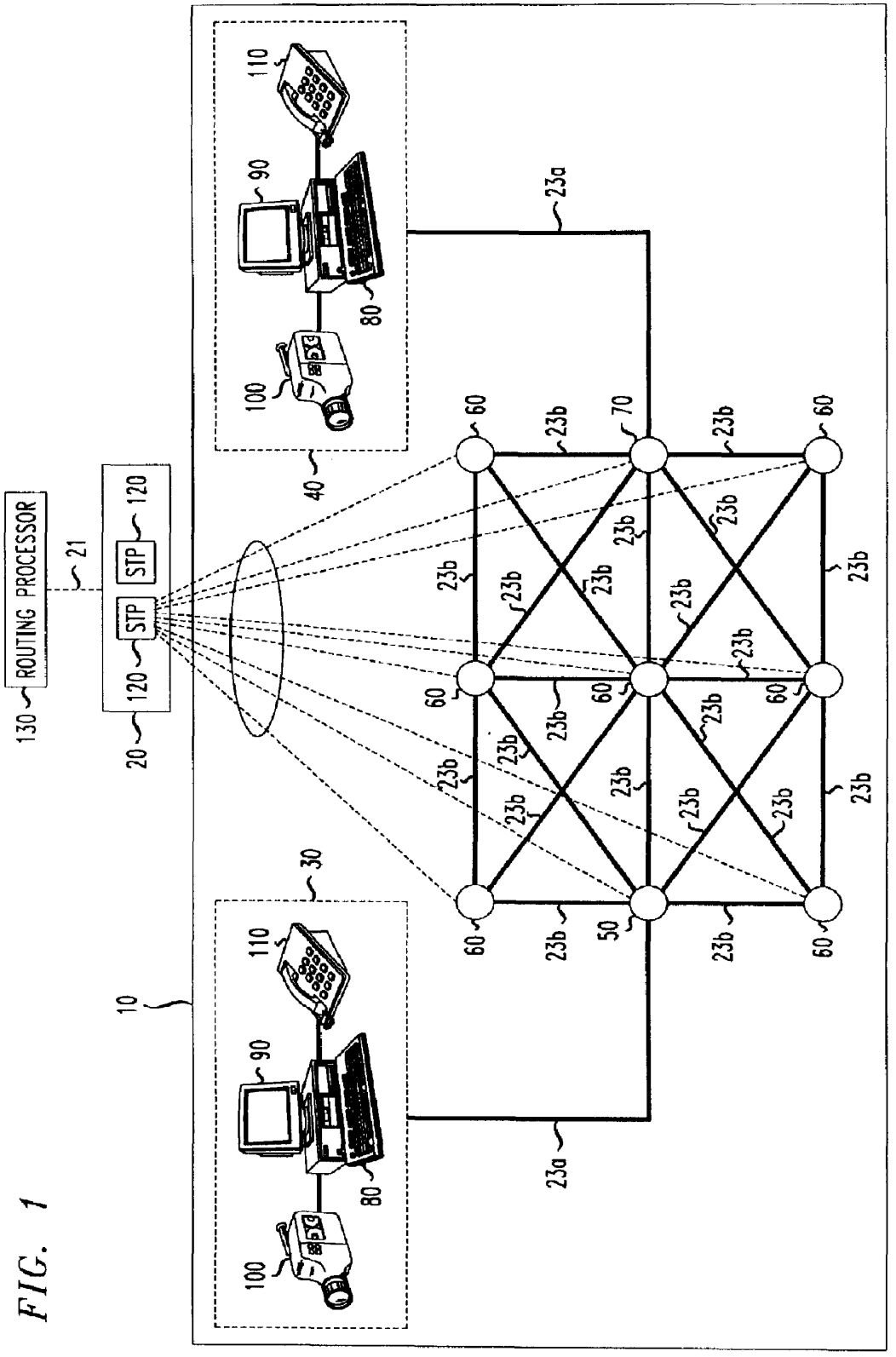

ROUTING PROCESSOR
130
21
120
STP
120
STP
20
10
40
30
110
90
80
100
60
70
50
23a
23b

FIG. 1

FIG. 2

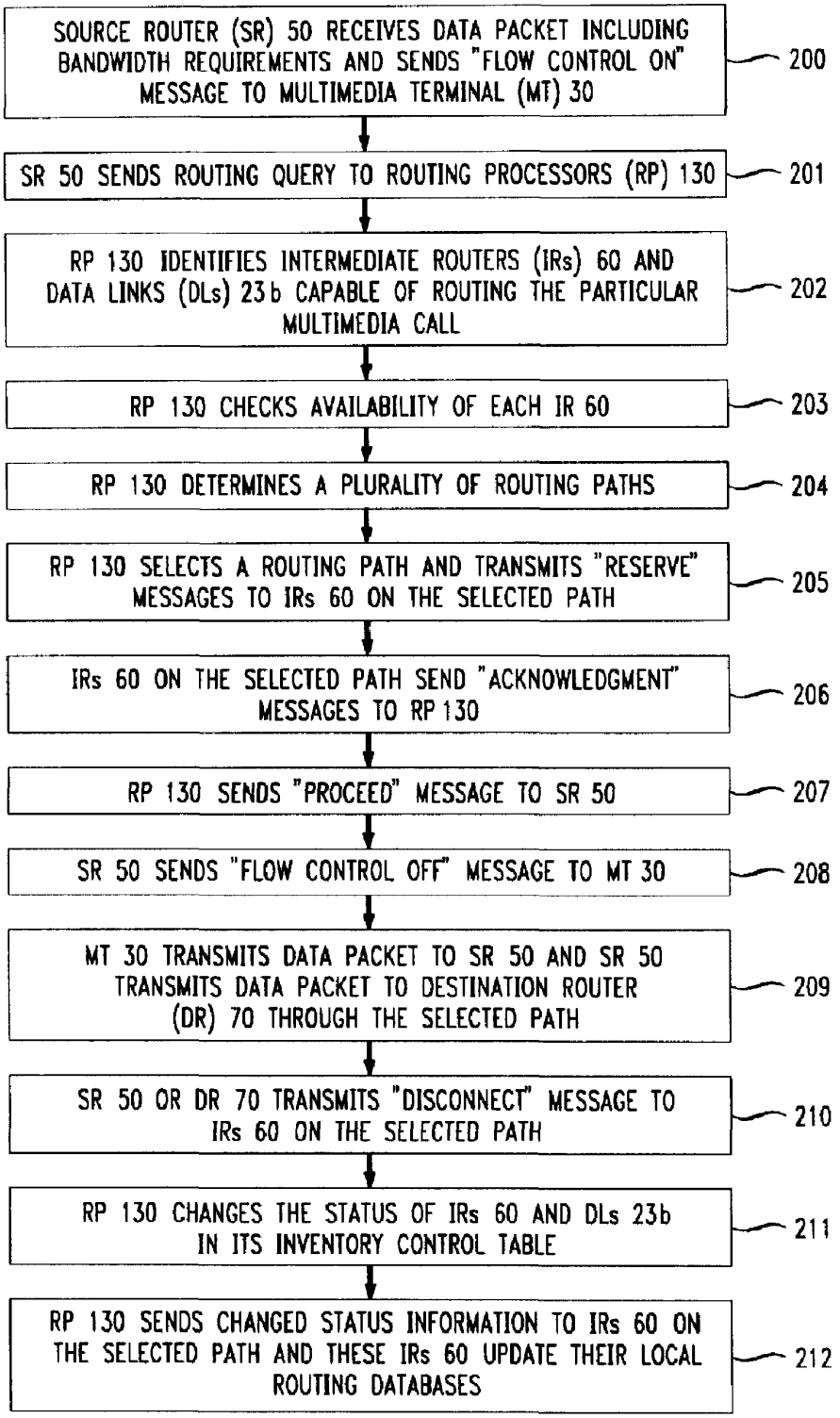

SOURCE ROUTER (SR) 50 RECEIVES DATA PACKET INCLUDING BANDWIDTH REQUIREMENTS AND SENDS "FLOW CONTROL ON" MESSAGE TO MULTIMEDIA TERMINAL (MT) 30
200
SR 50 SENDS ROUTING QUERY TO ROUTING PROCESSORS (RP) 130
201
RP 130 IDENTIFIES INTERMEDIATE ROUTERS (IRs) 60 AND DATA LINKS (DLs) 23b CAPABLE OF ROUTING THE PARTICULAR MULTIMEDIA CALL
202
RP 130 CHECKS AVAILABILITY OF EACH IR 60
203
RP 130 DETERMINES A PLURALITY OF ROUTING PATHS
204
RP 130 SELECTS A ROUTING PATH AND TRANSMITS "RESERVE" MESSAGES TO IRs 60 ON THE SELECTED PATH
205
IRs 60 ON THE SELECTED PATH SEND "ACKNOWLEDGMENT" MESSAGES TO RP 130
206
RP 130 SENDS "PROCEED" MESSAGE TO SR 50
207
SR 50 SENDS "FLOW CONTROL OFF" MESSAGE TO MT 30
208
MT 30 TRANSMITS DATA PACKET TO SR 50 AND SR 50 TRANSMITS DATA PACKET TO DESTINATION ROUTER (DR) 70 THROUGH THE SELECTED PATH
209
SR 50 OR DR 70 TRANSMITS "DISCONNECT" MESSAGE TO IRs 60 ON THE SELECTED PATH
210
RP 130 CHANGES THE STATUS OF IRs 60 AND DLs 23b IN ITS INVENTORY CONTROL TABLE
211
RP 130 SENDS CHANGED STATUS INFORMATION TO IRs 60 ON THE SELECTED PATH AND THESE IRs 60 UPDATE THEIR LOCAL ROUTING DATABASES
212

FIG. 3

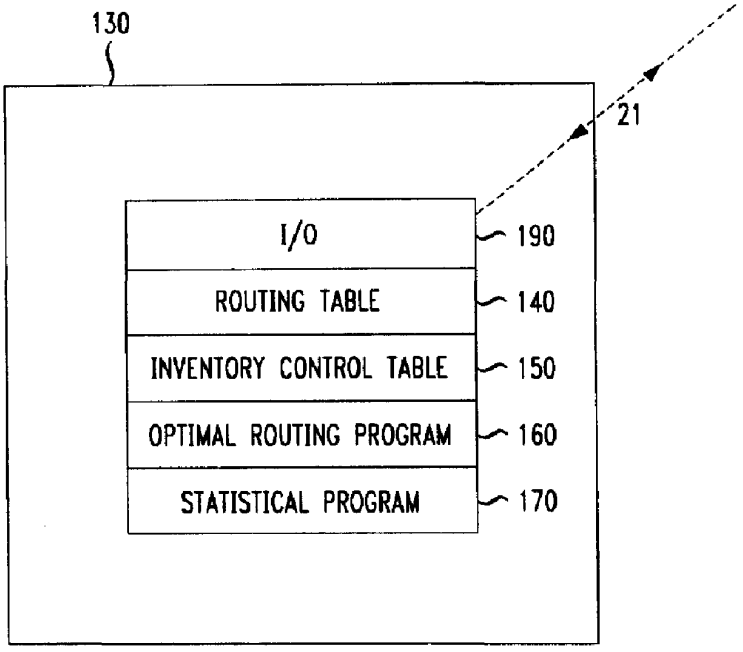
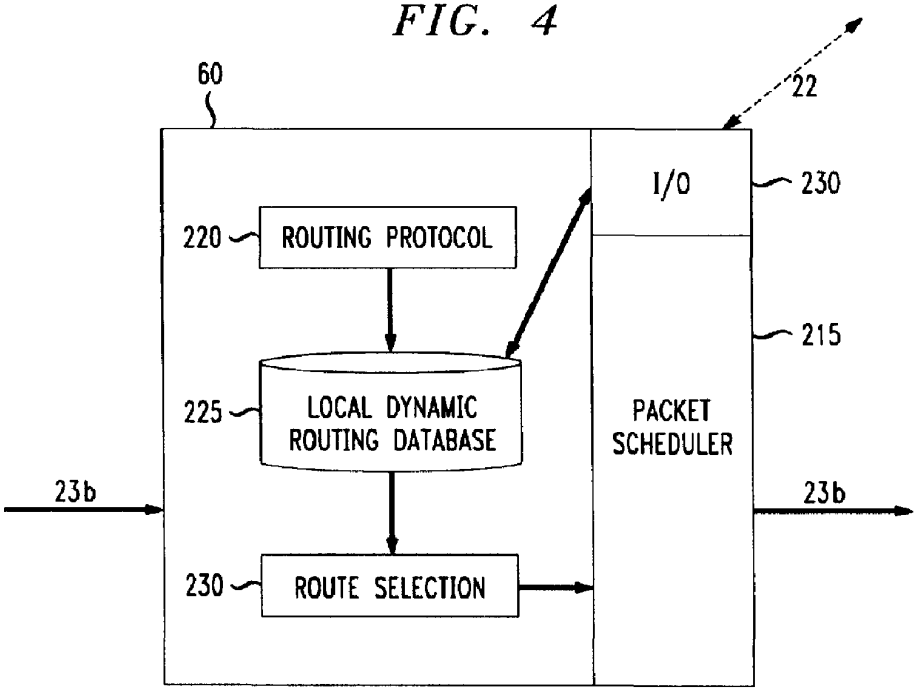

130
21
I/O
190
ROUTING TABLE
140
INVENTORY CONTROL TABLE
150
OPTIMAL ROUTING PROGRAM
160
STATISTICAL PROGRAM
170

FIG. 4


60
22
I/O
230
220
ROUTING PROTOCOL
215
225
LOCAL DYNAMIC
ROUTING DATABASE
PACKET
SCHEDULER
23b
23b
230
ROUTE SELECTION

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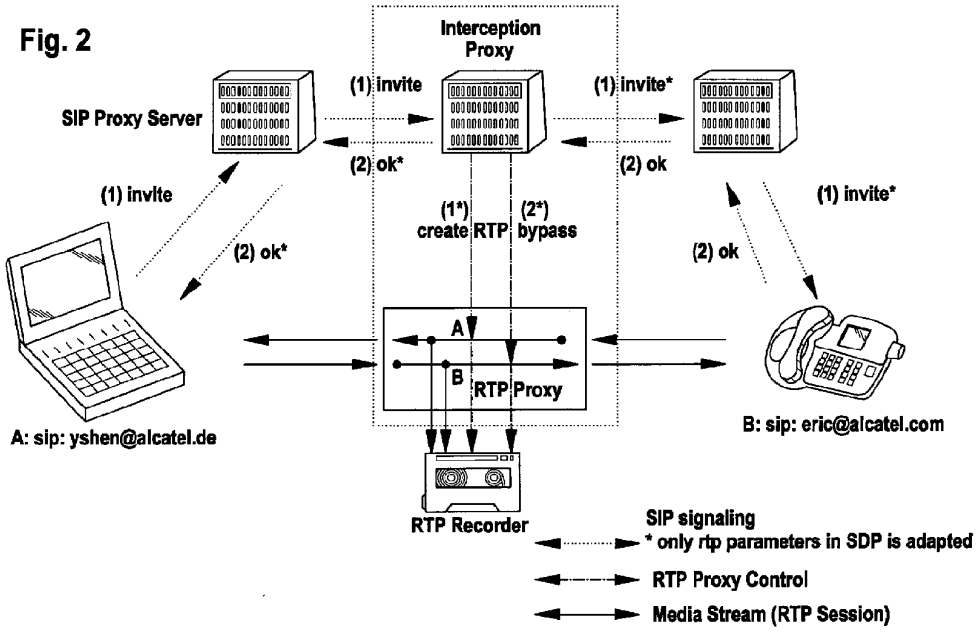
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(54) Lawful interception for VoIP calls in IP based networks

(57) The lawful interception device to monitor media streams of two IP parties includes a SIP (Session Initiation Protocol) proxy server or a MGC (Media Gateway Controller) to detect information in the signalling information being transmitted between the two IP (Internet Protocol) parties and to generate instructions out of the detected signalling information for instructing a RTP (Real-time Transport Protocol) proxy server to create channels to bypass a media stream to be intercepted via an intermediate storage medium. Due to adaptation of connection parameters in the SDP part of the SIP messages sent to the IP parties the interception is transparent to the IP parties.

Fig. 2



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Description

TECHNICAL FIELD OF THE INVENTION

[0001] This invention is related in general to the field of telecommunications systems. More particularly, the invention is related to a lawful interception device for media streams, in particular VoIP calls in IP based networks.

BACKGROUND OF THE INVENTION

[0002] Current lawful interceptions are deployed in class4/class5 switches of PSTN/PLMN networks. In 3G/UMTS or next generation networks, a connection may be IP end to end. No traffics will go through class 5/class4 switches. That means current lawful interception solutions cannot be used here. One solution may undertake an analysis of IP packets in a related network node, but it's difficult to know which route a call (media stream) will take through the network.

SUMMARY OF THE INVENTION

[0003] It is an object of the invention to provide a lawful interception device for VoIP calls in IP based networks.

[0004] The inventive lawful interception device detects information in the signalling information being transmitted between two IP parties and generates instructions out of the detected signalling information for instructing a RTP (Real-time Transport Protocol) proxy server to create channels to bypass a VoIP call to be intercepted via an intermediate storage medium. Instead of voice each media stream could be intercepted, e.g. data, internet access, e-mail, video, real-time pictures, etc.

[0005] In a SIP (Session Initiation Protocol) interception proxy server, where interception should be controlled, applications for interception are running to chose calls for interception. If a call should be monitored, the SIP proxy server has first to hold the invite message from A party. There are listening information in SDP (session description protocol) part of invite message.

[0006] SIP proxy server then instructs a RTP proxy server via a RTP proxy control interface to allocate a bypass channel for monitoring the media stream (A channel: sending to A party). The RTP information of this bypass channel (listening part: ip and port) is included in SDP part in the SIP invite message and passed to its destination.

[0007] When SIP proxy server has received a response of B party, he instructs RTP proxy via RTP proxy control interface to allocate another bypass channel for monitoring the media stream (B channel: sending to B party). The RTP information of this second bypass channel (listening part: ip and port) is included in SDP part in SIP ok message and send to its origination (A party).

[0008] After session setup, both parties will start RTP connections to RTP proxy server depending on connection parameters in its received SIP messages. But those are transparent to A and B. They do not know they are connected to a RTP proxy.

[0009] The RTP proxy can start record both media channels (A and B). At the end of this call, e.g. a media file with two sound tracks will be created by RTP proxy.

[0010] Advantages:

- centralized network node to intercept media streams,
- low cost of deployment,
- transparent to end users,
- the RTP proxy can also be used in the same way as above in a media gateway control (MEGACO, H.248) based network or H.323 network.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] For a better understanding of the present invention, reference is made to the accompanying drawings, in which:

FIG. 1 is a simplified block diagram of a portion of an exemplary telecommunications network according to the teachings of the prior art;

FIG. 2 is a simplified block diagram of a portion of an exemplary telecommunications network according to the teachings of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

[0012] FIG. 1 shows a portion of an exemplary telecommunications network according to the teachings of the prior art.

[0013] Two IP parties, e.g. yshen@ alcatel.de and eric@alcatel.com, are interconnected via two networks: a SIP signaling network and a transmission network. Via the SIP signaling network signaling is performed, e.g. a connection is established between the two IP parties. Via the transmission network the information to be transmitted, e.g. voice, data, etc. is transmitted in media streams (RTP session).

[0014] In the SIP based network, each SIP proxy server is responsible for signaling and session monitoring. The media stream will go from one IP endpoint to another IP endpoint. There is no need of a centralized media path like in PSTN network. A lawful interception of media stream could be done only in the network layer.

[0015] Recording media stream by analyzing network traffics for lawful interception is very expensive, due to the packet route through the IP network could change. Therefor the recording could only be done very closely to the endpoints. Additionaly a resembling of recorded packets is needed. A playing in real time will be difficult.

[0016] In the following definition and background information is provided regarding SIP, proxy server, RTP, SDP, etc.

SIP:

[0017] The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants. These sessions include Internet multimedia conferences, Internet telephone calls and multimedia distribution. Members in a session can communicate via multicast or via a mesh of unicast relations, or a combination of these.

[0018] SIP invitations used to create sessions carry session descriptions which allow participants to agree on a set of compatible media types. SIP supports user mobility by proxying and redirecting requests to the user's current location. Users can register their current location. SIP is not tied to any particular conference control protocol. SIP is designed to be independent of the lower-layer transport protocol and can be extended with additional capabilities.

[0019] The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, distance learning, Internet telephony and similar applications. SIP can invite both persons and "robots", such as a media storage service. SIP can invite parties to both unicast and multicast sessions; the initiator does not necessarily have to be a member of the session to which it is inviting. Media and participants can be added to an existing session.

[0020] SIP can be used to initiate sessions as well as invite members to sessions that have been advertised and established by other means. Sessions can be advertised using multicast protocols such as electronic mail, news groups, web pages or directories (LDAP), among others.

[0021] SIP transparently supports name mapping and redirection services, allowing the implementation of ISDN and Intelligent Network telephony subscriber services. These facilities also enable personal mobility. In the parlance of telecommunications intelligent network services, this is defined as: "Personal mobility is the ability of end users to originate and receive calls and access subscribed telecommunication services on any terminal in any location, and the ability of the network to identify end users as they move. Personal mobility is based on the use of a unique personal identity (i.e., personal number)." Personal mobility complements terminal mobility, i.e., the ability to maintain communications when moving a single end system from one subnet to another.

[0022] SIP supports five facets of establishing and terminating multimedia communications:

User location: determination of the end system to be used for communication;

User capabilities: determination of the media and media parameters to be used;

User availability: determination of the willingness of the called party to engage in communications;

Call setup: "ringing", establishment of call parameters at both called and calling party;

Call handling: including transfer and termination of calls.

[0023] SIP can also initiate multi-party calls using a multipoint control unit (MCU) or fully-meshed interconnection instead of multicast.
Internet telephony gateways that connect Public Switched Telephone Network (PSTN) parties can also use SIP to set up calls between them.

[0024] SIP is designed as part of the overall IETF multimedia data and control architecture currently incorporating protocols such as the real-time transport protocol (RTP) for transporting real-time data and providing QoS feedback.

[0025] A request and a response form together a transaction. SIP uses e.g. invite and ack messages to build up connections. Other messages used are e.g. ok, bye, options, register, cancel. SIP parties are identified via a SIP-ULR, e.g.: sip:cfientname@hostaddress. Each client may transmit requests to a proxy server or directly to an IP address.

[0026] An establishment of a connection is perfomed is three steps: sending an invite (request) message from a first IP party to a second IP party, sending an ok (response) message from the second IP party to the first IP party, sending an ack (response) message from the frist IP party to the second IP party. The invite message includes as much information as needed to allow the second IP party to judge whether a connection is wanted or not. The ack message is an acknowledgement, which serves to increase savety of the connection. SIP is thus not dependent on TCP or UDP.

[0027] The SIP according to the invention is the SIP currently standardized and modifications thereof and equivalents thereof.

RTP:

[0028] The Audio/Video Transport Working Group of IETF was formed to specify a protocol for real-time transmission of audio and video over UDP and IP multicast. This is the Real-time Transport Protocol, RTP, together with its associated profile for audio/video conferences and payload format documents. The payload formats currently under discussion include a number of media specific formats (MPEG-4, DTMF, PureVoice) and FEC techniques applicable to multiple formats (par-

ity FEC, Reed-Solomon coding). RTP is used to replace a normal circuit-switched trunk between two nodes.

[0029] The real-time transport protocol (RTP) is a payload format to be used for e.g. Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) encoded speech signals. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of service for real-time services. The data transport is e.g. augmented by the control protocol RTCP (Real-time Transport Control Protocol) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and mixers. The data transported by RTP in a packet, for example audio samples or compressed video data. A data packet includes e.g. the fixed RTP header, a possibly empty list of contributing sources, and the payload data.

[0030] The RTP according to the invention is the RTP currently under discussion and modifications thereof and equivalents thereof. RTP may be a protocol for both audio and video, or audio only, or video only, or audio, video and data, or audio and data, etc. One modification of RTP is e.g. RTP/I, an application level real-time protocol for distributed interactive media. Typical examples of distributed interactive media are shared whiteboards, networked computer games and distributed virtual environments. RTP/I defines a standardized framing for the transmission of data and provides mechanisms that are universally needed for this media class. Thereby RTP/I enables the development of reusable functionality and generic services that can be employed for multiple distributed interactive media. Examples for this kind of functionality are the ability to record sessions, to support late coming participants, and to provide security services. PTP/I is a protocol that follows the ideas of application level framing and integrated layer processing. It has been designed to be independent of the underlying network and transport layers. Thus RTP/I as a modified RTP protocol that reuses many aspects of RTP while it is thoroughly adapted to the specific needs of distributed interactive media.

Proxy, proxy server:

[0031] An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets, and, if necessary, rewrites a request message before forwarding it.

Server:

[0032] A server is an application program that accepts requests in order to service requests and sends back responses to those requests. Servers are either proxy, redirect or user agent servers or registrars.

User agent client (UAC), calling user agent:

[0033] A user agent client is a client application that initiates the SIP request.

SDP:

[0034] The Session Description Protocol (SDP) is intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.

[0035] The purpose of SDP is to convey information about media streams in multimedia sessions to allow the recipients of a session description to participate in the session. SDP is primarily intended for use in an internetwork, although it is sufficiently general that it can describe conferences in other network environments.

[0036] A multimedia session, for these purposes, is defined as a set of media streams that exist for some duration of time. Media streams can be many-to-many. The times during which the session is active need not be continuous.

[0037] Thus far, multicast based sessions on the Internet have differed from many other forms of conferencing in that anyone receiving the traffic can join the session (unless the session traffic is encrypted). In such an environment, SDP serves two primary purposes. It is a means to communicate the existence of a session, and is a means to convey sufficient information to enable joining and participating in the session. In a unicast environment, only the latter purpose is likely to be relevant.

[0038] Thus SDP includes:

- o Session name and purpose
- o Time(s) the session is active
- o The media comprising the session
- o Information to receive those media (addresses, ports, formats and so on)

[0039] As resources necessary to participate in a session may be limited, some additional information may also be desirable:

- o Information about the bandwidth to be used by the conference
- o Contact information for the person responsible for the session

[0040] In general, SDP must convey sufficient information to be able to join a session (with the possible

exception of encryption keys) and to announce the resources to be used to non-participants that may need to know.

[0041] SDP includes:

- o The type of media (video, audio, etc)
- o The transport protocol (RTP/UDP/IP, H.320, etc)
- o The format of the media (H.261 video, MPEG video, etc)

[0042] For an IP multicast session, the following are also conveyed:

- o Multicast address for media
- o Transport Port for media

[0043] This address and port are the destination address and destination port of the multicast stream, whether being sent, received, or both.

[0044] For an IP unicast session, the following are conveyed:

- o Remote address for media
- o Transport port for contact address

[0045] The semantics of this address and port depend on the media and transport protocol defined. By default, this is the remote address and remote port to which data is sent, and the remote address and local port on which to receive data. However, some media may define to use these to establish a control channel for the actual media flow.

[0046] The SDP according to the invention is the SDP currently standardized and modifications thereof and equivalents thereof.

[0047] FIG. 2 shows a portion of an exemplary telecommunications network according to the teachings of the present invention.

[0048] Like in fig. 1 two IP parties, e.g. yshen@alcatel.de and eric@alcatel.com, are interconnected via two networks: a SIP signaling network and a transmission network. Via the SIP signaling network signaling is performed, e.g. a connection is established between the two IP parties. Via the transmission network the information to be transmitted, e.g. voice, data, etc. is transmitted in media streams (RTP session).

[0049] Different from fig. 1 a lawful interception device is included in fig. 2. The lawful interception device is e. g. a processor with particular software. The processor is e.g. a digital signal processor, a controller, a microprocessor or the like. Instead of one processor two or more processors could be used. Two or more processors could be located at different sites. One processor could be used to perform SIP proxy server operations and another processor could be used to perform RTP proxy server operations. In general, one, two or more hardwares could be used to run one, two, or more softwares. Each software could in addition be run in parts on different hardware.

[0050] The lawful interception device includes a SIP (Session Initiation Protocol) proxy server or a MGC (Media Gateway Controller) to detect information in the signalling information being transmitted between two IP (Internet Protocol) parties and to generate instructions out of the detected signalling information for instructing a RTP (Real-time Transport Protocol) proxy server to create channels to bypass a media stream to be intercepted via an intermediate storage medium. Media streams are e.g. VoIP, data, internet access, e-mail, video, real-time pictures, music, video clips, video games, etc. The storage medium could be a compact disk, a magnetic storage medium, a read access memory, or the like.

[0051] The method for performing SIP signaling for a media stream includes the following steps:

receiving a SIP invite message of a first IP party,

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP invite message,

transmitting the adapted SIP invite message to a second IP party,

receiving a SIP response message of the seond IP party,

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP response message,

transmitting the adapted SIP response message to the first IP party.

[0052] At least one RTP parameter includes information about a bypass channel, an address, or a port. The RTP parameters sent to both IP parties differ from each other.

[0053] After receipt of the SIP invite message of the first IP party the SIP interception proxy server sends a request to the RTP interception proxy server to assign at least two channels for bothway communication. The interface used to communicate between SIP interception proxy server and RTP interception proxy server is a XML based API. The number of channels to be assigned may vary dependent of the amount of data to be transmitted, of the bandwith requested, of the quality of service requested, of the kind of information to be transmitted, e.g. voice, voice and data, voice and video, etc. At least one channel is assigned to transmit information between the RTP interception proxy server and the terminal of the first IP party. The terminal could be a phone, a laptop, a personal computer, a screenphone, a mobile phone, etc. At least one other channel is assigned to transmit information between the RTP interception proxy server and the terminal of the second IP party.

[0054] Assume channel A at the RTP interception proxy server is assigned to transmit information between the second IP terminal and the terminal of the first IP party, and channel B is assigned to transmit information between the the terminal of the first IP party and the second IP terminal. Then the RTP interception proxy server sends information about the assignment of channels A and B to the SIP interception proxy server. The SIP interception proxy server includes information about channel A in the invite message to be send to the second IP party. The information about channel A is advantageously included in the connection parameter information to be included in the SDP of the SIP invite message. After receipt of the SIP response message of the seond IP party, which corresponds to an ok message stating that a connection to the first IP party is desired, the SIP interception proxy server exchanges the connection parameter included in the SDP part of the ok message by the information about channel B. The modified ok message including the information about channel B is send to the first IP party.

[0055] Thus the first IP party will send data to channel B and receive data via channel A of the RTP interception proxy server. The second IP party will send data to channel A and receive data via channel B of the RTP interception proxy server. Within the lawful interception device the intermediate storage medium is connected to both channel A and B. Thus the information flow between both IP parties will transfer the intermediate storage medium and thus interception is enabled. The first party is not aware on which channel the second party is sending, and the second party is not aware on which channel the first party is sending. Thus interception is transparent regarding the two IP parties.

[0056] A computer program for performing at least part of the steps of the inventive method could be used as an upgrade software, which is sold e.g. to service providers, which will upgrade one or more SIP proxy server thus enabling a usual SIP proxy server having the functionality of an SIP interception proxy server. The computer program includes at least the following steps:

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP invite message,

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP response message.

The computer program could also be programmed to perform all steps of the method as described above.

[0057] Within an IP network one, two, or more SIP proxy servers could be used, one, two, or more SIP interception proxy servers could be used, one, two, or more RTP proxy servers could be used, and one, two, or more RTP interception proxy servers could be used.

[0058] The IP network could be a wireline network, a wireless network, or a combination of both.

List of abbreviations:

[0059]

3G	Third Generation
API	Application Programmer Interface
AMR	Adaptive Multi-Rate
AMR-WB	AMR-Wideband
DTMF	Dual-Tone Multi-Frequency
FEC	Forward Error Correction
H248	ITU standard
H261	ITU standard
H320	ITU standard
H323	ITU standard
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
LDAP	Lightweight Directory Access Protocol
MEGACO	Media Gateway Controller
MCU	Multipoint Control Unit
MPEG	Motion Picture Expert Group
MGC	Media Gateway Controller
NGN	Next Generation Network
PSTN	Public Switched Telephone Network
PLMN	Public Land Mobile Network
QoS	Quality of Service
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
UAC	User Agent Client
UDP	User Datagram Protocol
UMTS	Universal Mobile Transmission System
VoIP	Voice over IP
XML	extensible Markup Language

Claims

1. Lawful interception device including a SIP (Session Initiation Protocol) proxy server or a MGC (Media Gateway Controller) to detect information in the signalling information being transmitted between two IP (Internet Protocol) parties and to generate instructions out of the detected signalling information for instructing a RTP (Real-time Transport Protocol) proxy server to create channels to bypass a media stream to be intercepted via an intermediate storage medium.

2. SIP interception proxy server to detect information in the signalling information being transmitted between two IP (Internet Protocol) parties and to generate instructions out of the detected signalling information for instructing a RTP (Real-time Trans-

port Protocol) proxy server to create channels to bypass a media stream to be intercepted via an intermediate storage medium.

3. Interception MGC to detect information in the signalling information being transmitted between two IP (Internet Protocol) parties and to generate instructions out of the detected signalling information for instructing a RTP (Real-time Transport Protocol) proxy server to create channels to bypass a media stream to be intercepted via an intermediate storage medium.

4. Method for performing SIP signaling for a media stream, including the following steps:

receiving a SIP invite message of a first IP party,

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP invite message,

transmitting the adapted SIP invite message to a second IP party,

receiving a SIP response message of the seond IP party,

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP response message,

transmitting the adapted SIP response message to the first IP party.

5. Method according to claim 4, wherein at least one connection parameter includes information about a bypass channel, an address, or a port.

6. Method according to claim 4, wherein the connection parameters sent to both IP parties differ from each other.

7. Computer program for performing at least part of the steps of the method according to claim 4, including the following steps:

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP invite message,

adapting at least one connection parameter in the SDP (Session Description Protocol) of the received SIP response message.

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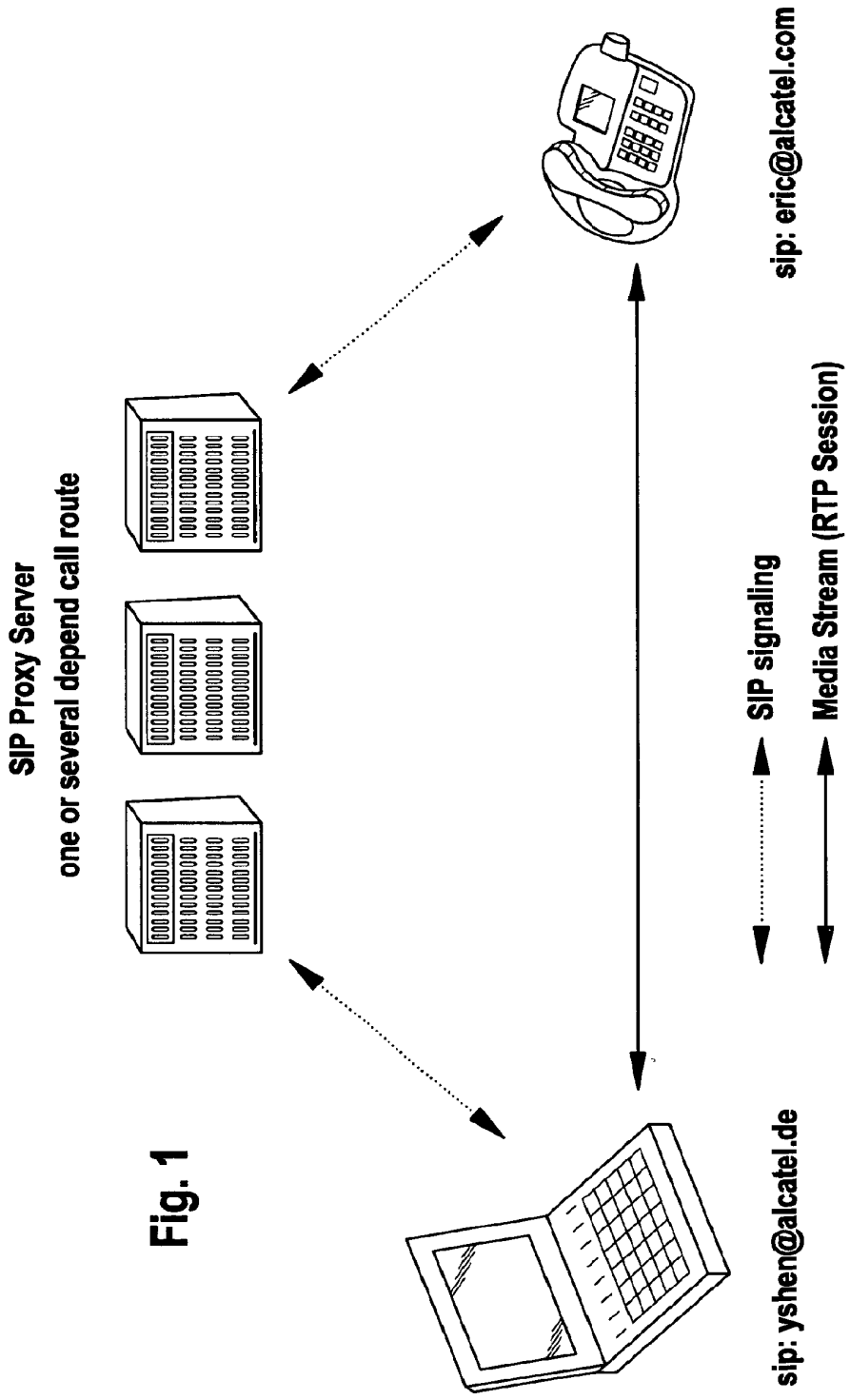

SIP Proxy Server
one or several depend call route
sip: eric@alcatel.com
SIP signaling
Media Stream (RTP Session)
sip: yshen@alcatel.de

Fig. 1

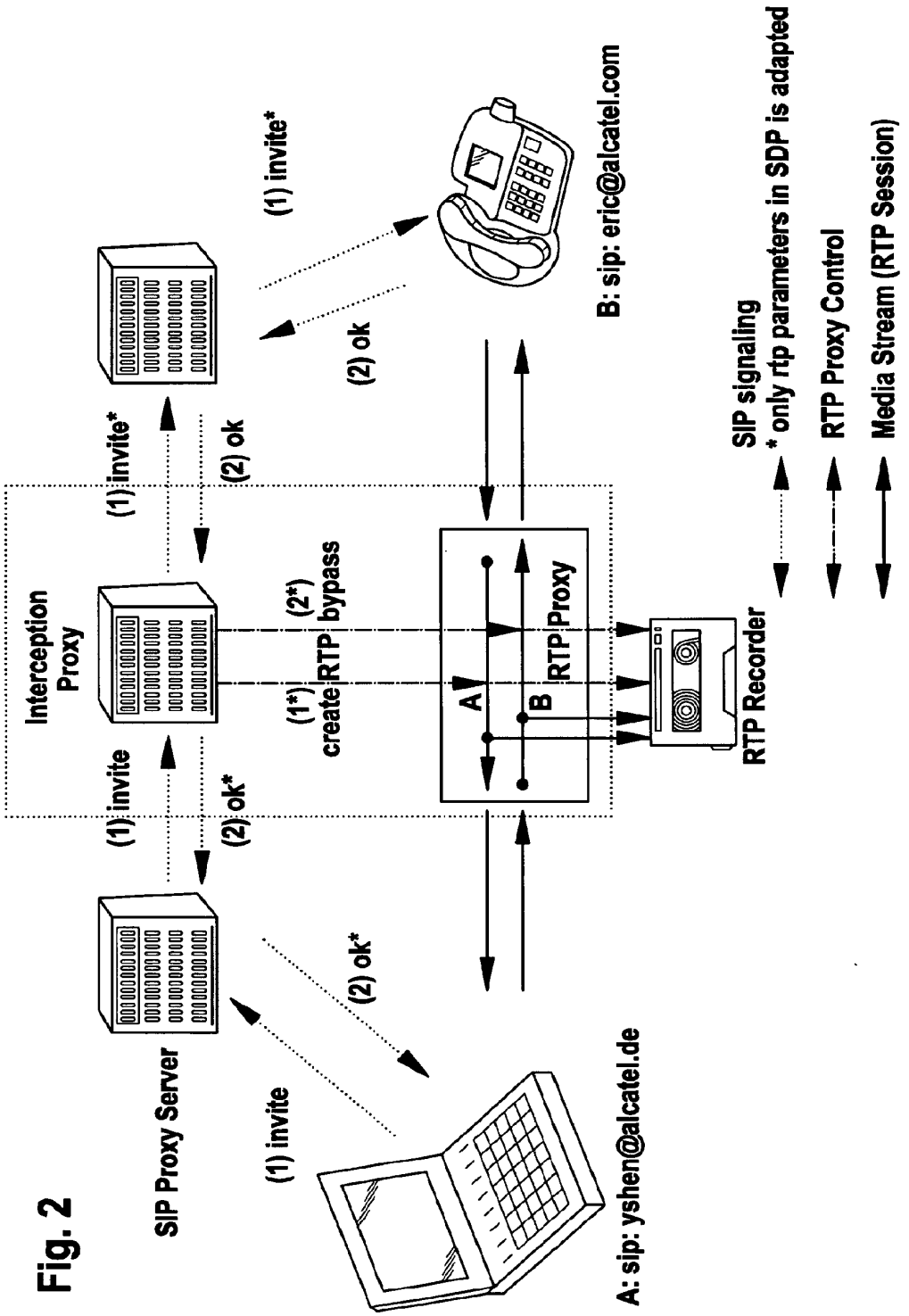

(1) invite*
(2) ok
B: sip: eric@alcatel.com
(1) invite*
(2) ok
Interception Proxy
(1) invite
(2) ok*
(1*) create RTP
(2*) bypass
A
B
RTP Proxy
RTP Recorder
SIP signaling
* only rtp parameters in SDP is adapted
RTP Proxy Control
Media Stream (RTP Session)
SIP Proxy Server
(1) invite
(2) ok*
A: sip: yshen@alcatel.de

Fig. 2

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 02 36 0235

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Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
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The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	16 January 2003	Karavassilis, N

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
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T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

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ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 02 36 0235

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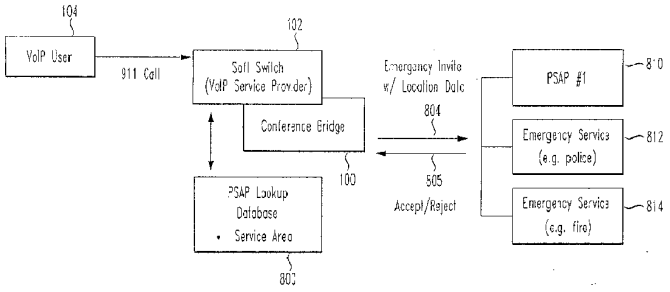
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(54) Title: VOICE OVER INTERNET PROTOCOL (VOIP) LOCATION BASED 911 CONFERENCING



(57) Abstract: Voice Over Internet Protocol (VoIP) emergency calls to an Emergency Response Center (ERC) are handled through a VoIP conference bridge on a VoIP service provider's soft switch. The soft switch works with a VoIP positioning center (VPC) to obtain location information, which is compared against a PSAP database to find an initial best-appropriate PSAP for the location of the emergency caller. The PSAP is issued an Invite message to join the conference, establishing an emergency call. Third parties such as police, ambulance may be issued Invite messages to join the conference. Cold transfers are avoided by Inviting participants to join a single emergency conference rather than passing an emergency call from party to party (e.g., from PSAP to police to ambulance, etc.) The PSAP, other emergency responders, and even the initial VoIP emergency caller may leave and rejoin the VoIP conference without dropping the conference between the others.

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VOICE OVER INTERNET PROTOCOL (VoIP) LOCATION BASED 911 CONFERENCING

This application is related to and claims priority from a co-pending U.S. Provisional Application No. 60/723,960, entitled "Voice Over Internet Protocol (VoIP) Location Based Conferencing", filed on October 6, 2005; U.S. Provisional Application No. 60/733,789, entitled "Voice Over Internet Protocol (VoIP) Multi-User Conferencing", filed on November 7, 2005; and U.S. Provisional Application No. 60/723,961, entitled "Voice Over Internet Protocol (VoIP) Location Based 911 Conferencing", filed on October 6, 2005; the entirety of all three of which are expressly incorporated herein by reference.

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates generally to Voice Over Internet (VoIP) protocols and architectures. More particularly, it relates to location based services for the provision of 911 emergency services using VoIP protocols and architectures.

2. Background of the Related Art

911 is a phone number widely recognized in North America as an emergency phone number that is used by emergency dispatch personnel, among other things, to determine a location of a caller. Enhanced 911 (E911) is defined by the transmission of callback number and location information. E911 may be implemented for landline and/or wireless devices.

A Public Safety Answering Point (PSAP) is a dispatch office that receives 9-1-1 calls from the public. A PSAP may be a local, fire or police department, an ambulance service or a regional office covering all services. A 9-1-1 ("911") service becomes E-9-1-1 ("E911") when automatic number identification and automatic location information from a communications device (e.g. wireless phone, VoIP Phone, etc.) is provided to the 911 operator.

Voice-Over-Internet Protocol (VoIP) is a technology that emulates a phone call, but instead of using a circuit based system such as the telephone network, utilizes packetized data transmission techniques most notably implemented in the Internet. 911 calls made using VoIP technology must reach the correct PSAP, but there currently is no uniform interface to the various PSAPs for call delivery because the technology for connecting calls varies. For instance, not all PSAPs are Internet Protocol (IP) capable. Some PSAPs are accessed via ordinary public switched telephone network (PSTN) telephone lines. Some PSAPs are accessed through selective routing such as direct trunks. Still other PSAPs are accessed using IP connections. There is no uniformity among the thousands of different PSAPs.

Moreover, some Public Safety Access Points (PSAPs) are not enhanced, and thus do not receive the callback or location information at all from any phone, landline or wireless.

The use of VoIP technology is growing quickly. As people adopt voice-over-IP (VoIP) technology for routine communications, the inventors herein recognize that there is a growing need to access E911 services including provision of location information from a VoIP device.

The existing E911 infrastructure is built upon copper wire line voice technology and is not fully compatible with VoIP. Given VoIP technology, there are at least three VoIP scenarios:

1. A VoIP UA that is physically connected to a static data cable at a "home" address. For instance, an Analog Telephone Adapter (ATA) that is connected to the "home" data cable and uses traditional telephone devices.
2. A VoIP UA that is physically connected to a data cable at a location different than its "home" address. For instance, a laptop computer device utilized away from home as a VoIP software telephone would be a VoIP 'visitor' device as described by this scenario.
3. A VoIP UA that is wireleless, physically disconnected from any data cable. In this situation, the VoIP UA connects to the VoIP service provider via

either a wide-are wireless technology (e.g., cellular, PCS, WiMAX) or via a local-area wireless technology (e.g., Wireless Fidelity (WiFi), UWB, etc.) using a laptop computer or handheld device.

VoIP phone calls are routed to a VoIP voice gateway, from which they are passed on to their destination. A VoIP voice gateway or soft switch is a programmable network switch that can process the signaling for all types of packet protocols. Also known as a 'media gateway controller,' 'call agent,' or 'call server, such devices are used by carriers that support converged communications services by integrating SS7 telephone signaling with packet networks. Softswitches can support, e.g., IP, DSL, ATM and frame relay.

The challenges evident with respect to determining the location of a calling VoIP telephone is perhaps most evident with respect to its use to make an emergency call (e.g., a 911 call). Nevertheless, VoIP telephone technology is quickly replacing conventional switched telephone technology. However, because VoIP is Internet Protocol (IP) based, call related information such as CallerID type services may not be available or accurate. A location of a given VoIP device may be provisioned to be at a given geographic location, or queried from a home location register (HLR) in a mobile system.

In addition, some Public Safety Access Points (PSAPs) are not enhanced, and thus do not receive the callback or location information at all from any phone; landline, cellular or VoIP.

Moreover, there is complexity in public access to Public Safety Answering Points due to lack of a Session Initiation Protocol (SIP) Uniform Resource Identifier (URI) for all PSAPs. (SIP is the IP-based protocol defined in IETF RFCs 3261 and 2543.) SIP is one of two dominant protocols used by the VoIP industry. URI is the addressing technology for identifying resources on the Internet or a private intranet. URIs were originally defined as two types: Uniform Resource Locators (URLs) which are addresses with network location, and Uniform Resource Names (URNs) which are persistent names that are address independent. Today, a URI is defined by its purpose rather than the URL vs. URN classification.) Some PSAPs are accessed only by conventional telephone

line, others only by direct telephone trunk lines. Not all PSAPs are accessible via the Internet.

Fig. 5 shows basic conventional VoIP elements required to interconnect a VoIP emergency E911 caller to a relevant public safety access point (PSAP).

In particular, as shown in Fig. 5, VoIP telephone devices **102a, 102b, 102c** (collectively referred to as **102**) are connected to respective VoIP Service Provider (VSP) soft switches **104a, 104b, 104c** (collectively referred to as **104**) using an Internet Protocol (IP) connection, most commonly over the Internet. The VoIP service provider's soft switch **104** in turn communicates with a respective VoIP Positioning Center (VPC) **106a, 106b, 106c** (collectively referred to as **106**) using an appropriate IP connection. Each VSP requires use of their own VPC, as depicted in Fig. 5.

Fig. 6 shows in more detail conventional VoIP elements required by a VPC to interconnect a VoIP emergency E911 caller to a relevant public safety access point (PSAP).

In particular, as shown in Fig. 6, each VPC **106** comprises its own respective route determination module **404**, call delivery module **406**, and provisioning list **408**.

A respective location information server (LIS) **108** services each of the VPCs **106**. The LIS **108** is responsible for storing and providing access to the subscriber location information needed for E9-1-1 call processing (as defined by the NENA VoIP Location Working Group).

A conventional VoIP Positioning Center (VPC) **106** is a system that attempts to determine the appropriate or correct PSAP **114** that a VoIP emergency E911 call should be routed to based on the VoIP subscriber's position. The conventional VPC **106** also returns associated routing instructions to the VoIP network. The conventional VPC **106** additionally provides the caller's location and the callback number to the relevant PSAP through the automatic location identifier (ALI) (The ALI is a database that accepts a PSAP query, and using that relates a specific telephone number to a street address. In the case of

an Emergency Services Query Key (ESQK), the ALI database steers the query to the appropriate VPC and steers the response back to the PSAP. An ALI is typically owned by a LEC or a PSAP.)

Further as shown in Fig. 6, each VSP route the emergency 9-1-1 call, without location object added, to their VPC **106.** The VPC must determine the correct PSAP **114** (collectively represented by PSAP **114a**, **114b** and **114c**) and route to it using the appropriate technology.

In a first scenario, the VPC **106** passes the 9-1-1 call to the PSAP **114a** using an INVITE telephone number message, via a media gateway **110** that translates between the IP protocol of the INVITE message and a telephone line interface, and interfaces with the public switched telephone network (PSTN) **112.**

In a second scenario, the VPC **106** passes the 9-1-1 call to the PSAP **114b** using an INVITE S/R message, via an ESGW **120** and selective router **122.** In this scenario, the selective router **122** is connected to the relevant PSAP **114b** via direct trunks.

In a third scenario, the VPC **106** passes the 9-1-1 call to the PSAP **114c** using an INVITE PSAP message, via IP, to the PSAP **114c.**

In the second and third scenario, the ALI **126** must be inter-connected with each VPC **106** (a,b,c). Furthermore, each VPC is burdened with supporting all the various ALI protocols: ve2, e2, PAM, legacy NENA, etc.

Thus, as can be appreciated, an Emergency call (e.g., 911, E911) may require the involvement of one or more Response Centers (RCs), e.g., Public Safety Access Point (PSAP) in addition to the RC that initially receives the emergency call. This is because there is a possibility that the emergency call is received by a PSAP other than that which is assigned to the geographic region that the caller is currently located in.

Accordingly, the PSAP that initially answers the call may need to transfer the emergency call to the correct PSAP. During transfer of the emergency VoIP call, the original RC may or may not remain on the line, but for safety purposes will not likely want to disconnect or cold transfer the emergency

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call. This is because errors may occur in the transfer, resulting in valuable time lost. One cause of a faulty transfer of the E911 call would be that the VoIP user has not updated the location stored by the VPC, or quite simply that bad routing has occurred. Another cause would be that the nature of the emergency requires multiple parties to be involved (e.g., fire/police, police/FBI, ambulance/CDC, etc.).

Conventional solutions are based on tools that can be used to find the phone numbers of other emergency response centers. The ERC receiving the call initially will perform a look-up for the correct response center, and may dial the identified correct response center, agency, etc., and transfer the call via direct dial/public switched telephone network (PSTN.

One exemplary conventional solution is called an Intelligent Emergency Network (IEN), available from Intrado Inc. of Longmont, Colorado. However, such conventional solutions typically require the emergency response center to know the direct dial lines of every PSAP, ESP, ERC, etc. nationally. Moreover, those lines may not always be staffed. Other potential problems would be caused if no automatic location identification (ALI) information is accessible or available.

There is a need for an architecture and methodology that both simplifies the complexity of a VoIP call transfers with respect to an emergency response center such as a public safety access point (PSAP).

SUMMARY OF THE INVENTION

In accordance with the principles of the present invention, a method of connecting an emergency caller with an emergency response center comprises establishing an emergency call conference. The emergency caller is added to the established emergency call conference, and the emergency response center is added to the emergency call conference. The emergency call is established after the emergency caller and the emergency response center are both added to the emergency call conference.

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BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 shows an exemplary architecture of a VoIP emergency call conference bridge application operating in a VoIP soft switch of a VoIP provider to provide VoIP emergency call conferencing, in accordance with the principles of the present invention.

Fig. 2 shows an exemplary message flow diagram of VoIP location based 911 conferencing, in accordance with the principles of the present invention.

Fig. 3 shows an exemplary architecture of a VoIP conference bridge application operating in a VoIP soft switch of a VoIP provider to provide VoIP emergency call conferencing, in accordance with the principles of the present invention.

Fig. 4 shows an exemplary message flow diagram for establishing a VoIP location based conference, in accordance with the principles of the present invention.

Fig. 5 shows basic conventional VoIP elements required to interconnect a VoIP emergency E911 caller to a relevant public safety access point (PSAP).

Fig. 6 shows in more detail conventional VoIP elements required to interconnect a VoIP emergency E911 caller to a relevant public safety access point (PSAP).

DETAILED DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

The present invention handles emergency calls through the use of a conference bridge on a VoIP service provider's soft switch. The soft switch works with a VoIP positioning center (VPC) to obtain location information, which may be gathered or confirmed by the initial recipient of the call, to ensure that appropriate participants to the emergency conference call are Invited to join the call. With the present invention in place, any number of emergency calls can be made, including any number of ERCs, PSAPs, ERPs, etc., (limited only by the number of conference bridges that can be established in provisioned equipment,

e.g., in the VoIP service provider's soft switch). Cold transfers can be avoided by Inviting participants to join a single emergency conference rather than passing an original call from party to party (e.g., from PSAP to police to ambulance, etc.) Moreover, the emergency call can survive as long as a participant remains in the emergency conference call, even after the original emergency caller hangs up.

Fig. 1 shows an exemplary architecture of a VoIP emergency call conference bridge application operating in a VoIP soft switch of a VoIP provider to provide VoIP emergency call conferencing, in accordance with the principles of the present invention.

In particular, as shown in Fig. 1, a user of a VoIP communications device **104** makes an emergency call (e.g., a 911 call). The VoIP service provider of the VoIP communications device **104** receives the 911 call, and assigns it to an available VoIP emergency conference call bridge **100**. The soft switch **102** obtains location information relating to the VoIP communications device **104**, either directly from the VoIP communications device **104** itself (e.g., if it includes a GPS device) or from a VoIP positioning center (VPC). The VoIP soft switch **102** compares the location information in a PSAP lookup database **800** to determine an initial PSAP for the service area responsible for the location of the VoIP communications device **104**. The PSAP lookup database provides an appropriate URL or other address information of the initial PSAP to the VoIP soft switch **102**, which in turn addresses an Invite message **804** (preferably including location information relating to the location of the VoIP communications device **104**). The PSAP **810**, in response, sends either an Accept message or a Reject message to the soft switch **102** in response to the Invite message **804**. Additional emergency services departments (e.g., police **812**, fire **814**, etc.) may be subsequently sent an Invite message to join the same VoIP emergency conference call.

Thus, the VoIP communication device **104** dials the appropriate emergency number (e.g., 911), and in response the VoIP service provider's soft switch **102** otherwise responsible for routing the user's calls instead establishes a

VoIP conference bridge **100** and places the incoming emergency call into the VoIP conference bridge **100**.

Although the initial emergency VoIP communication device **104** is a VoIP device, the soft switch **102** may additionally include interfaces to the Public Switched Telephone Network (PSTN) to permit non-VoIP emergency service provider's to join into the VoIP conference bridge.

Alternatively, instead of automatically placing the initial VoIP emergency caller **104** into the established VoIP conference bridge **100**, the VoIP soft switch **102** may instead Invite the initial VoIP emergency caller **104** to join the conference call via the VoIP conference bridge **100**. In response, the initial VoIP emergency caller **104** presumably accepts the Invite message and joins the VoIP conference bridge **100**.

At this point, the soft switch **102** may confirm location with the initial VoIP emergency caller **104** (if location information was provided with the initial call from the VoIP communication device **104**), or determines location from the subscriber's VPC, and captures the Location Object (LO).

The initial VoIP emergency caller **104** sends the LO and a 911 Invite message with an RC type (e.g., Fire Department, Homeland Security, etc.) to the soft switch **102** managing the VoIP conference bridge **100**.

The soft switch **102** sends the LO and Invite information to the VPC, which identifies the proper additional conference participant(s) (e.g., a PSAP, RC, first responder, other interested party, etc.) and corresponding contact information, and invites the proper participants to join the call.

The invited participant(s) can also invite other entities to join the VoIP emergency conference. While it is presumed that all participants in the VoIP emergency conference call may participate in the call, it is possible to include 'listen only' participants. For instance, a voice and/or data recording line may be invited to the VoIP emergency conference call to record any data and/or voice conversation.

Fig. 2 shows an exemplary message flow diagram of VoIP location based 911 conferencing, in accordance with the principles of the present invention.

In particular, as shown in Fig. 2, an emergency call **712** (e.g., 911) is placed from VoIP communications device **104.**

In response, the VoIP soft switch establishing the VoIP emergency conference call bridge transmits an emergency VoIP conference call Invite message (with or without a location object) **714** (or other location request) to the VoIP Positioning Center (VPC) **701**. Based on the location of the initiating VoIP emergency caller **104**, the VPC pass at least one Invite message using Internet Protocol (e.g., over the Internet) to interested third parties such as an initially contacted RC-1/PSAP **702**, PSAP-2 **703**, PSAP-n **704**, etc. The first emergency center contacted (RC-1/PSAP **702**) responds by verifying the location object and passing the same, along with the Invite RC Type, to the soft switch **718**.

As the emergency call progresses, other emergency responders may be brought into the VoIP emergency conference call. For instance, the soft switch that manages the VoIP conference call bridge **100** initiates an Invite message with location object to the VPC **701**, which in turn transmits an Invite message **722** to a subsequent emergency response center (e.g., PSAP-2 **703**). That subsequent emergency response center **703** responds by verifying/modifying the location object, and the Invite RC Type, as shown in message **724.**

The VoIP soft switch **102** may continue to invite additional emergency responders (or other parties) by passing an Invite message with location object through the VPC **701**, which passes an Invite with location object to the relevant other emergency responders **704.**

As an example to explain advantages of the present invention, the scenario is given where an emergency 9-1-1 call is routed to a PSAP based on a presumed or default location of the VoIP caller, but in fact it turns out that the PSAP that receives the VoIP call is not the correct entity to handle emergency calls from the particular location that the VoIP caller is currently at. Such errors

may occur, e.g., due to the user not updating the SLDB, bad routing, etc. In this scenario, the initial VoIP communications device dials 9-1-1, a conference line is initiated by the soft switch, an initially determined PSAP receives an Invite message to join the VoIP emergency conference bridge. The PSAP confirms/determines the user's location, and in the given scenario would determine that another PSAP is needed instead of or in addition to the PSAP on the line. In particular, the initial PSAP captures the Location Object (LO) and either rejects the Invite to join the VoIP emergency conference call (and is then removed from the conference bridge) or continues to participate in the VoIP emergency conference call (and so then stays on the conference bridge). Either way, a 911 emergency call Invite message is sent with the LO to the soft switch managing the VoIP emergency conference bridge. The VoIP soft switch sends the LO to the VPC, which then identifies the proper PSAP based on the LO and initiates an Invite message addressed over IP to the proper PSAP to join into the VoIP emergency conference call through the soft switch.

The VoIP conference bridge then joins the proper PSAP to the VoIP emergency conference call with the initial VoIP emergency caller (and with the initially contacted PSAP, if the initially contacted PSAP continues to participate in the call). In this manner, the initial VoIP emergency caller is kept on the line throughout the process, with preferably no additional manual action or key entry required from the initial emergency caller.

At the conclusion of the VoIP emergency call, the VoIP conference bridge is closed.

In cases where the initial routing of the VoIP emergency call was correct, the VoIP conference bridge would still be used, and the initial two parties would participate in the VoIP emergency conference call (e.g., the initial VoIP emergency caller and the initially Invited RC or PSAP). If no other parties are invited, additional queries to the VoIP Positioning Center (VPC) would not be necessary. If additional parties are invited, the soft switch would use location information and RC Type information from the initial RC or PSAP to determine the identity of other relevant RCs and/or PSAPs.

In general principle, Fig. 3 shows an exemplary architecture of a VoIP conference bridge application operating in a VoIP soft switch of a VoIP provider to provide VoIP call conferencing, in accordance with the principles of the present invention.

In particular, as shown in Fig. 3, a VoIP communications device **104** is serviced by their service provider's soft switch **102**. A positioning center **106** provides location data upon request from the soft switch **102**. Other VoIP users **110, 112, 114** etc. are potential members of any given conference.

Conference bridges **100** are implemented on the VoIP soft switch **102** located, e.g., at the VoIP service provider's VoIP network.

While the VoIP soft switch **102** is preferably capable of being provisioned with as many VoIP conference bridges **100** as are required in any particular application, only one conference bridge **100** is shown in Fig. 3 for simplicity of explanation.

Also, while the conference bridge **100** is shown implemented in the soft switch **102**, it can be embodied within another suitable network element having an Internet Protocol (IP) type connection (e.g., TCP/IP) with the initial user **104** as well as with the potential conferees **110, 112, 114**.

In accordance with the principles of the present invention, location information relating to the initial VoIP user **104** is passed to the VoIP conference bridge **100**, either from the user's VoIP communication device **104** or from their respective location server **106**. The location information is then compared by the VoIP soft switch **102** to find an initial desired PSAP.

The VoIP soft switch **102** makes use of the location information and other existing data or user input (e.g., existing preferences on file on the Soft Switch **102**, user entry through the keypad of the communications device **104**, or voice response). Based on the location and user input, the VoIP conference bridge **100** identifies the desired PSAP to be asked or Invited to join the conference currently established by the initial VoIP user **104** on the conference bridge **100**, and outputs an Invite or request message **204** to join that conference **100** to the specific URL(s), phone number(s) and/or other identifying address

information relating to VoIP communications equipment **110**, **112**, **114** of the relevant PSAP.

The soft switch **102** may also maintain the attributes and rules from other VoIP communication devices **110**, **112**, **114** etc. for receiving conference bridge calls, as well as the fixed location (e.g., a place of business) or the ability to query for a current location (e.g., for mobile communication devices such as mobile phones) for each device. Based on this information, with or without other user input (e.g., to select or prioritize among a list of available third parties), the soft switch **102** invites one or more other communication devices **110**, **112**, **114**, etc. to join the conference bridge. This creates a voice link between the first user **104** and the other third parties **110**, **112**, **114** without requiring the first user **104** to know the contact information or name of the third parties **110**, **112**, **114**.

Fig. 4 shows an exemplary message flow diagram for establishing a VoIP location based conference, in accordance with the principles of the present invention.

In particular, as shown in Fig. 4, the initial VoIP user **104** sends a request for conference bridge call to the soft switch **102**. Preferably the initial VoIP user **104** includes location information with the conference request call **201**. However, as depicted in Fig. 3, location information can be obtained from an appropriate positioning server **106** if not available from the initial VoIP user **104**.

Subsequent to the incoming conference call **201**, a suitable PSAP (and/or other emergency services, including a recorder line) is determined and invited with respective invite messages **204**, **206**.

In operation, the user's VoIP communication device **104** dials a pre-determined phone number (or URL) of the emergency service (e.g., 911) to initiate a VoIP emergency conference bridge **100** on the relevant VoIP soft switch **102**.

Fig. 3 shows use of a VoIP positioning center (VPC) **106**. The VoIP soft switch **102** may receive the user's location information either from each of the VoIP communication devices **104**, **110**, **112**, **114** etc., or from the VPC **106**.

The VoIP soft switch **102** preferably uses both the location information of the initiating VoIP user **104**, together with any profile criteria set for a given conference bridge **100**, to determine a suitable PSAP or other emergency services entity to be sent INVITE messages inviting them to join the established VoIP emergency conference bridge **100**.

The VoIP soft switch **102** invites one or more other VoIP communication devices **110, 112, 114**, (relating to emergency services) to join the VoIP emergency conference bridge **100**. This creates a voice link between the initiating VoIP user **104** that initially called into the VoIP emergency conference bridge **100**, and the other potential, third party conferees **110, 112, 114**, etc., without requiring the initiating VoIP user **104** to know the name or even the contact information of the other potential, third party emergency conferees **110, 112, 114**, etc.

Upon receipt of an invite to a VoIP conference bridge **204, 206**, the potential other VoIP users **110, 112, 114**, etc. (PSAPs) are preferably notified similar to an incoming telephone call, e.g. with a ring signal, though it may be customized to be distinguished from the sound of an otherwise ordinary incoming phone call. For instance, a given unique phone tone may be activated upon receipt of an invite **204, 206** to a conference bridge **100**.

In accordance with the principles of the present invention, the VoIP communication device(s) **110, 112, 114** receiving invitations to join a VoIP emergency call conference **100** may be provided with a filter that automatically rejects any/all invite requests not meeting their own specific criteria (e.g., the first invited participant to accept the Invite message) maintained on their VoIP devices **110, 112, 114** themselves, though such filtering may alternatively be performed at a network level, e.g., at the VoIP soft switch **102** or other centralized location.

Benefits of the invention include that there is no effective limit to the number of participants in the VoIP emergency conference call, there are no cold transfers of a call as VoIP invitees enter or leave the conference bridge **100**, and

there is the ability to continue the conference call even after the initial VoIP user **104** making the emergency call disconnects.

The present invention has particular applicability with any/all VoIP users, VoIP service providers, and Public Safety Access Points (PSAPs).

The invited VoIP users **110, 112, 114** may include a filter allowing through only acceptable Invite messages based on criteria established by or on the receiving VoIP communication devices **110, 112, 114.**

The present invention allows VoIP users to efficiently and quickly find and invite their most appropriate responder to their emergency, with minimal user interaction. This is particularly helpful for mobile VoIP users (e.g., while driving, walking, etc.) Moreover, there is no effective limit to the number of participants in the conference call (within network hardware limits of the conference bridge itself). There is also no risk of cold transfers of a VoIP telephone call as participants aren't handled in point-to-point connections that are transferred but rather join or exit an established conference at will. Furthermore, emergency personnel from various departments and locations in the conference call can continue in the conference even after the initial emergency caller disconnects.

Potential markets for the present invention include VoIP service providers who may implement the inventive VoIP emergency conference calling as a value added services for users. VoIP location based conferencing in accordance with the principles of the present invention has particular applicability with any/all VoIP users, VoIP service providers, and Public Safety Access Points (PSAPs).

While the invention has been described with reference to the exemplary embodiments thereof, those skilled in the art will be able to make various modifications to the described embodiments of the invention without departing from the true spirit and scope of the invention.

CLAIMS

What is claimed is:

1. A method of connecting an emergency caller with an emergency response center, comprising:

establishing an emergency call conference;

adding said emergency caller to said established emergency call conference; and

adding said emergency response center to said emergency call conference;

wherein said emergency call is established after said emergency caller and said emergency response center are both added to said emergency call conference.

2. The method of connecting an emergency caller with an emergency response center according to claim 1, further comprising:

adding a third party to said emergency call conference;

3. The method of connecting an emergency caller with an emergency response center according to claim 1, wherein:

at least three parties are present in said emergency call conference at least at a beginning of said emergency call.

4. The method of connecting an emergency caller with an emergency response center according to claim 1, wherein said emergency response center comprises:

a public safety access point (PSAP).

5. The method of connecting an emergency caller with an emergency response center according to claim 2, wherein said third party comprises:

a police dispatcher.

6. The method of connecting an emergency caller with an emergency response center according to claim 2, wherein said third party comprises:

a fire department.

7. The method of connecting an emergency caller with an emergency response center according to claim 2, wherein said third party comprises:

an ambulance company.

8. The method of connecting an emergency caller with an emergency response center according to claim 1, wherein:

said emergency caller is added to said emergency call conference after said emergency response center is added to said emergency call conference.

9. The method of connecting an emergency caller with an emergency response center according to claim 1, wherein:

said emergency response center is added to said emergency call conference after said emergency caller is added to said emergency call conference.

10. Apparatus for connecting an emergency caller with an emergency response center, comprising:

means for establishing an emergency call conference;

means for adding said emergency caller to said established emergency call conference; and

means for adding said emergency response center to said emergency call conference;

wherein said emergency call is established after said emergency caller and said emergency response center are both added to said emergency call conference.

11. Apparatus for connecting an emergency caller with an emergency response center according to claim 10, further comprising:

means for adding a third party to said emergency call conference;

12. The apparatus for connecting an emergency caller with an emergency response center according to claim 10, wherein:

at least three parties are present in said emergency call conference at least at a beginning of said emergency call.

13. The apparatus for connecting an emergency caller with an emergency response center according to claim 10, wherein said emergency response center comprises:

a public safety access point (PSAP).

14. The apparatus for connecting an emergency caller with an emergency response center according to claim 11, wherein said third party comprises:

a police dispatcher.

15. The apparatus for connecting an emergency caller with an emergency response center according to claim 11, wherein said third party comprises:

a fire department.

16. The apparatus for connecting an emergency caller with an emergency response center according to claim 11, wherein said third party comprises:

an ambulance company.

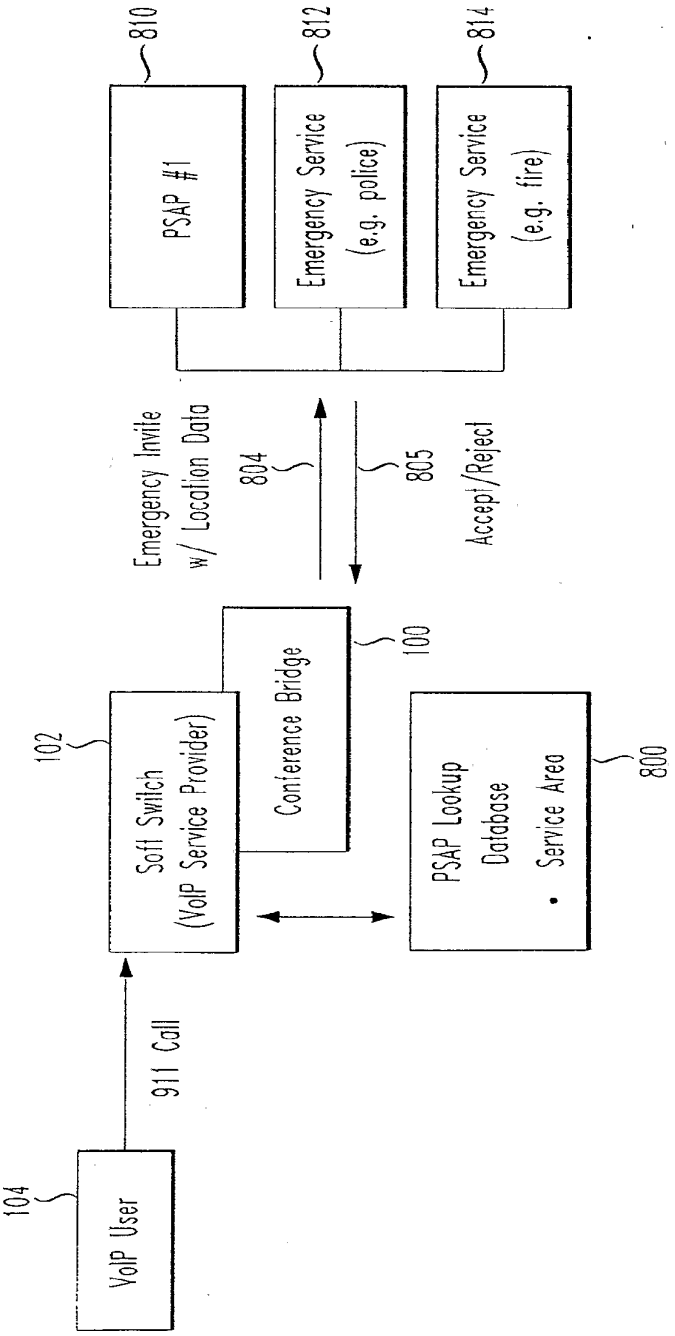
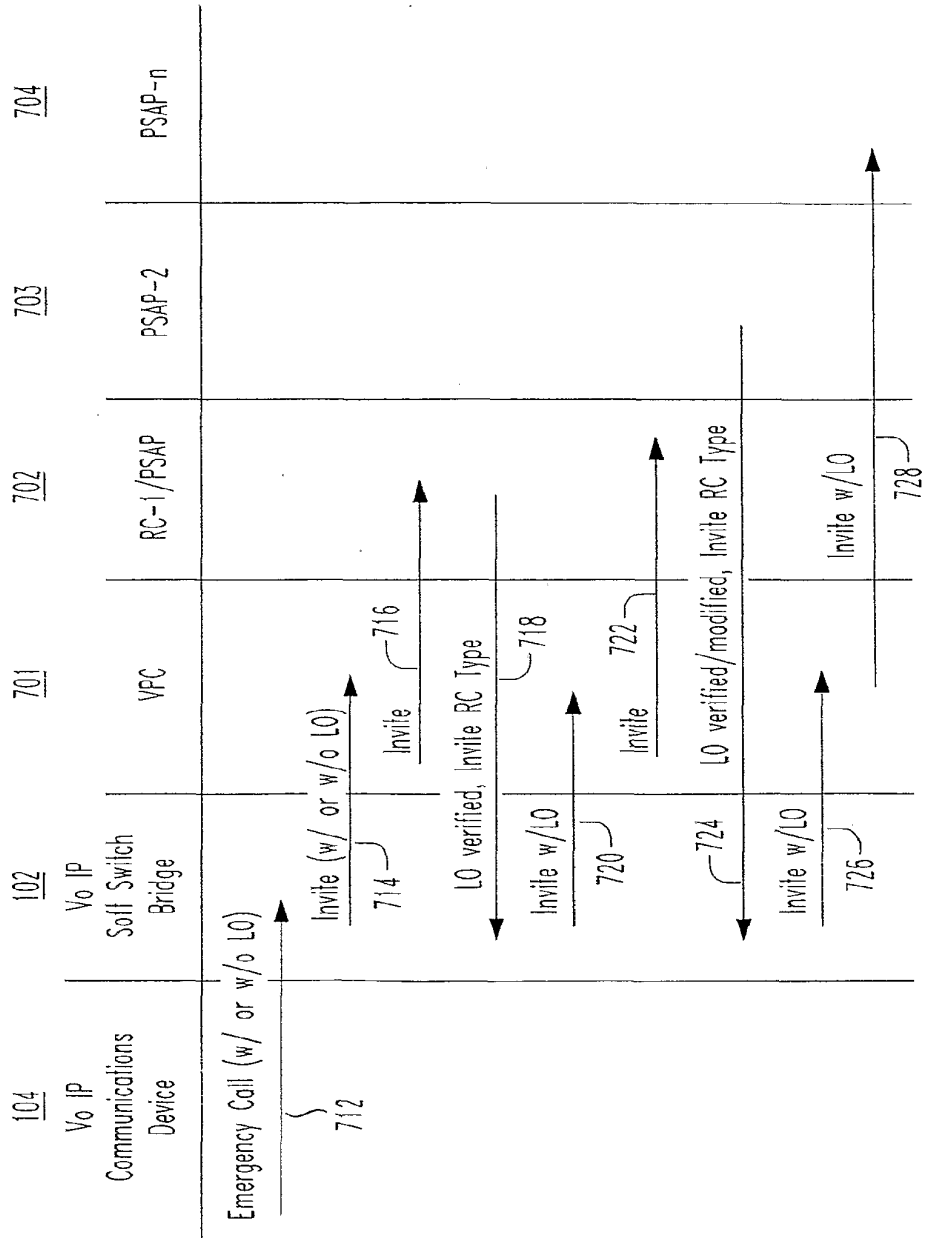

810
PSAP #1
812
Emergency Service
(e.g. police)
814
Emergency Service
(e.g. fire)
Emergency Invite
w/ Location Data
804
805
Accept/Reject
102
Soft Switch
(VoIP Service Provider)
Conference Bridge
100
PSAP Lookup
Database
• Service Area
800
911 Call
104
VoIP User

FIG. 1

FIG. 2


104
Vo IP Communications Device
102
Vo IP Soft Switch Bridge
701
VPC
702
RC-1/PSAP
703
PSAP-2
704
PSAP-n
Emergency Call (w/ or w/o LO)
712
Invite (w/ or w/o LO)
714
Invite
716
LO verified, Invite RC Type
718
Invite w/LO
720
Invite
722
LO verified/modified, Invite RC Type
724
Invite w/LO
726
Invite w/LO
728

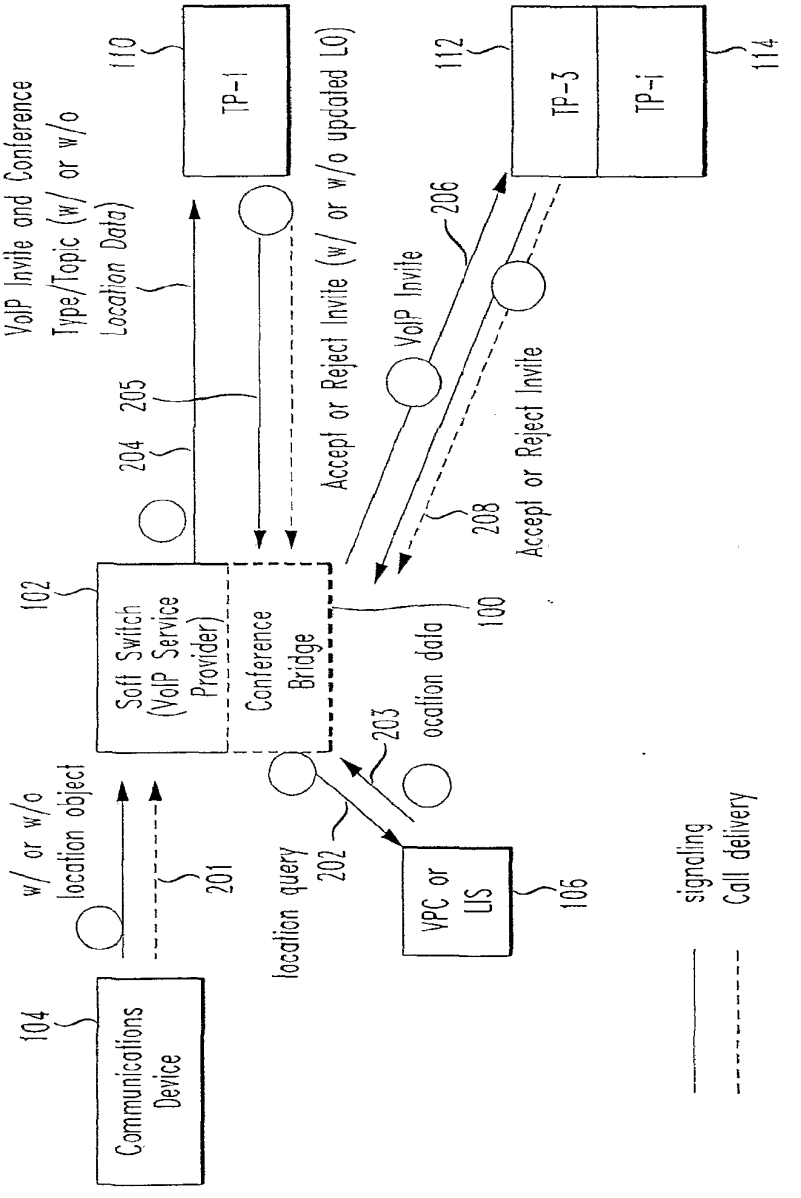


FIG. 3

META 1002
META V. VOIP-PAL

FIG. 4

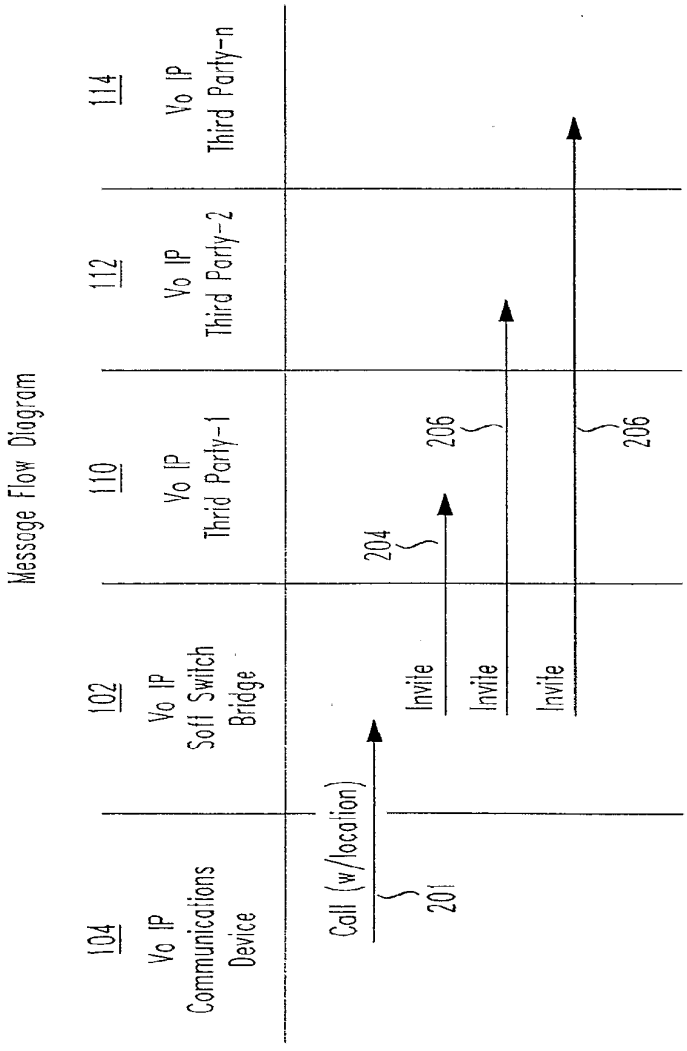
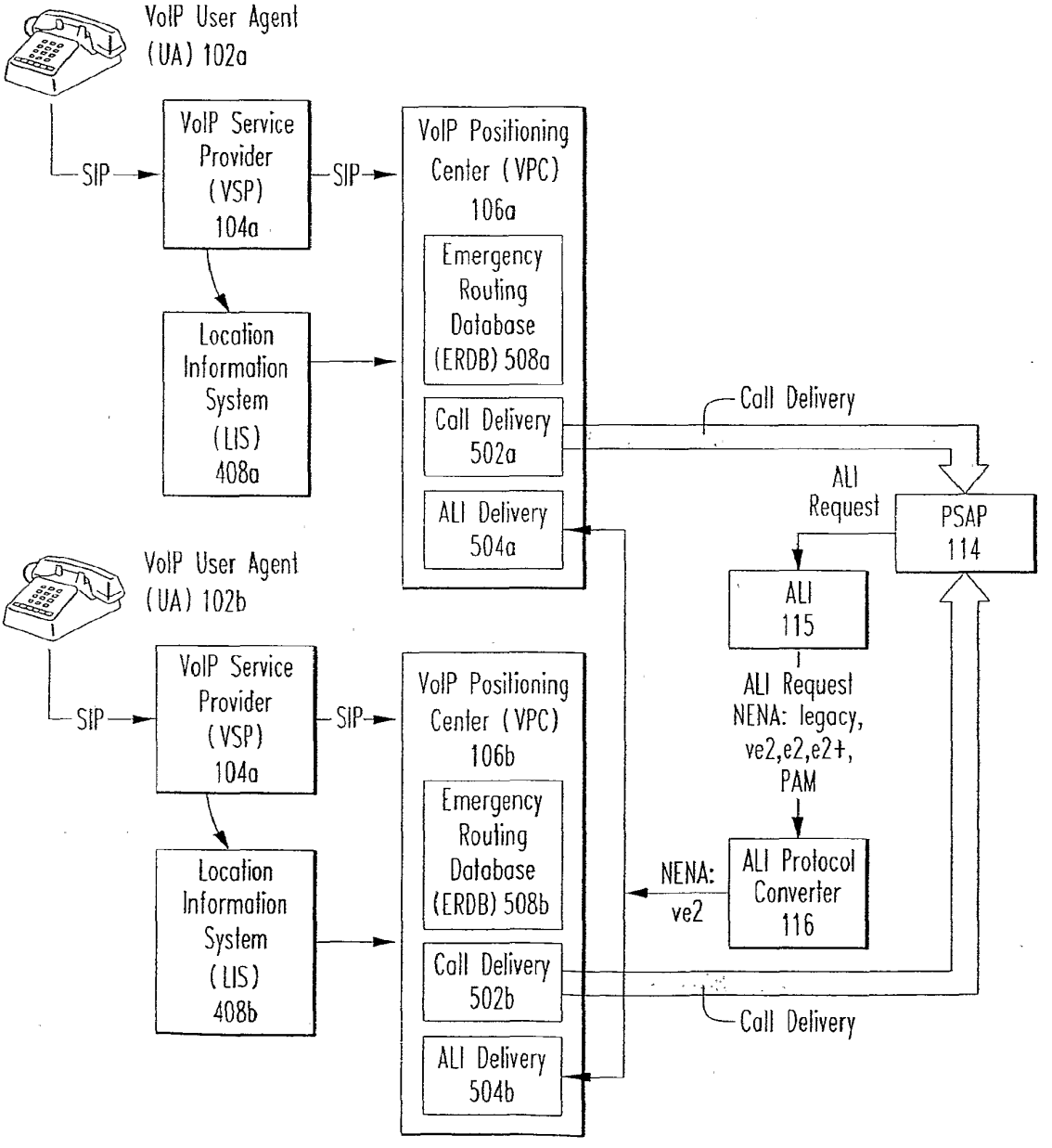

Message Flow Diagram
104
Vo IP
Communications
Device
102
Vo IP
Soft Switch
Bridge
110
Vo IP
Third Party-1
112
Vo IP
Third Party-2
114
Vo IP
Third Party-n
Call (w/location)
201
Invite
204
206
Invite
206
Invite

FIG. 5
(PRIOR ART)



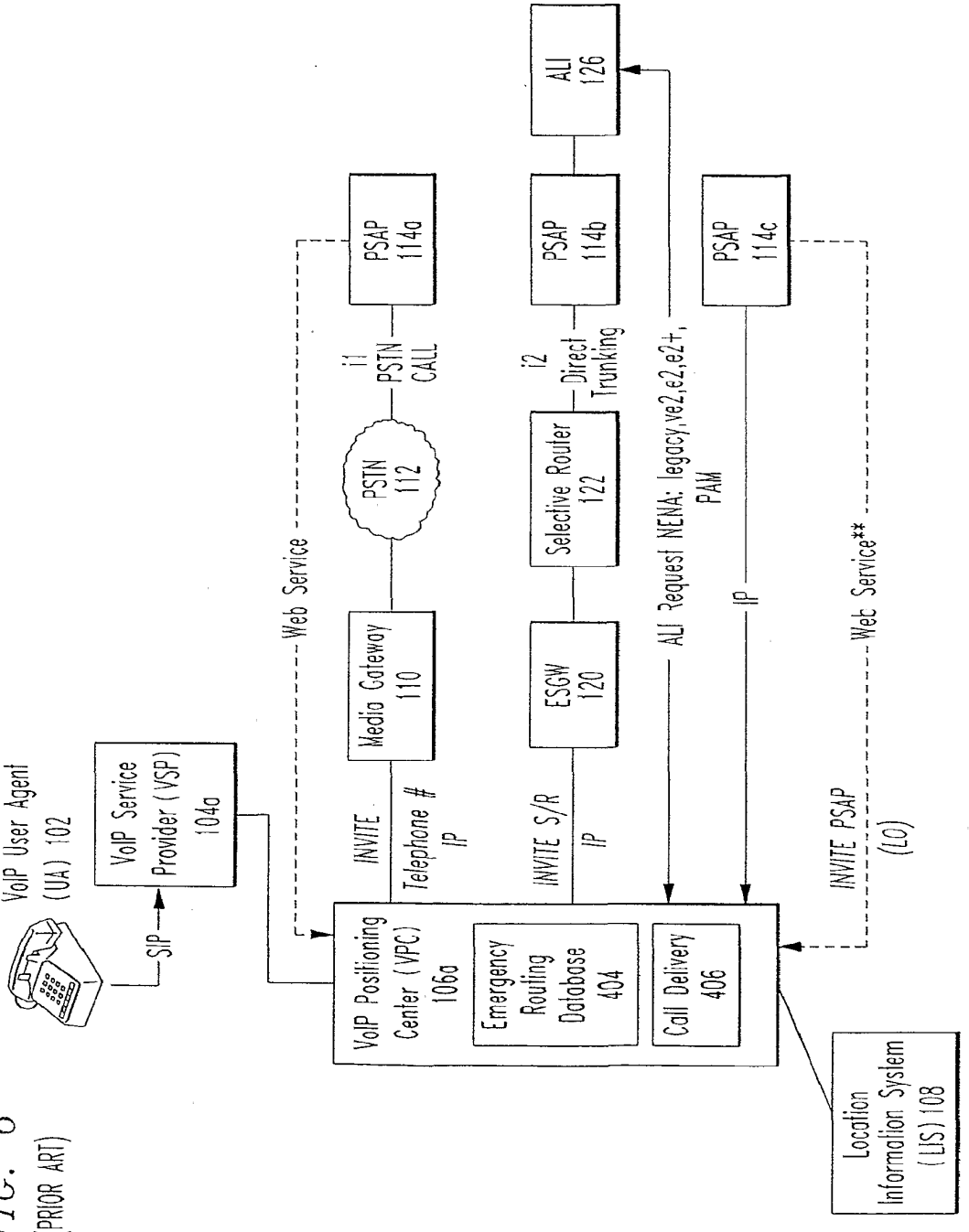


FIG. 6
(PRIOR ART)

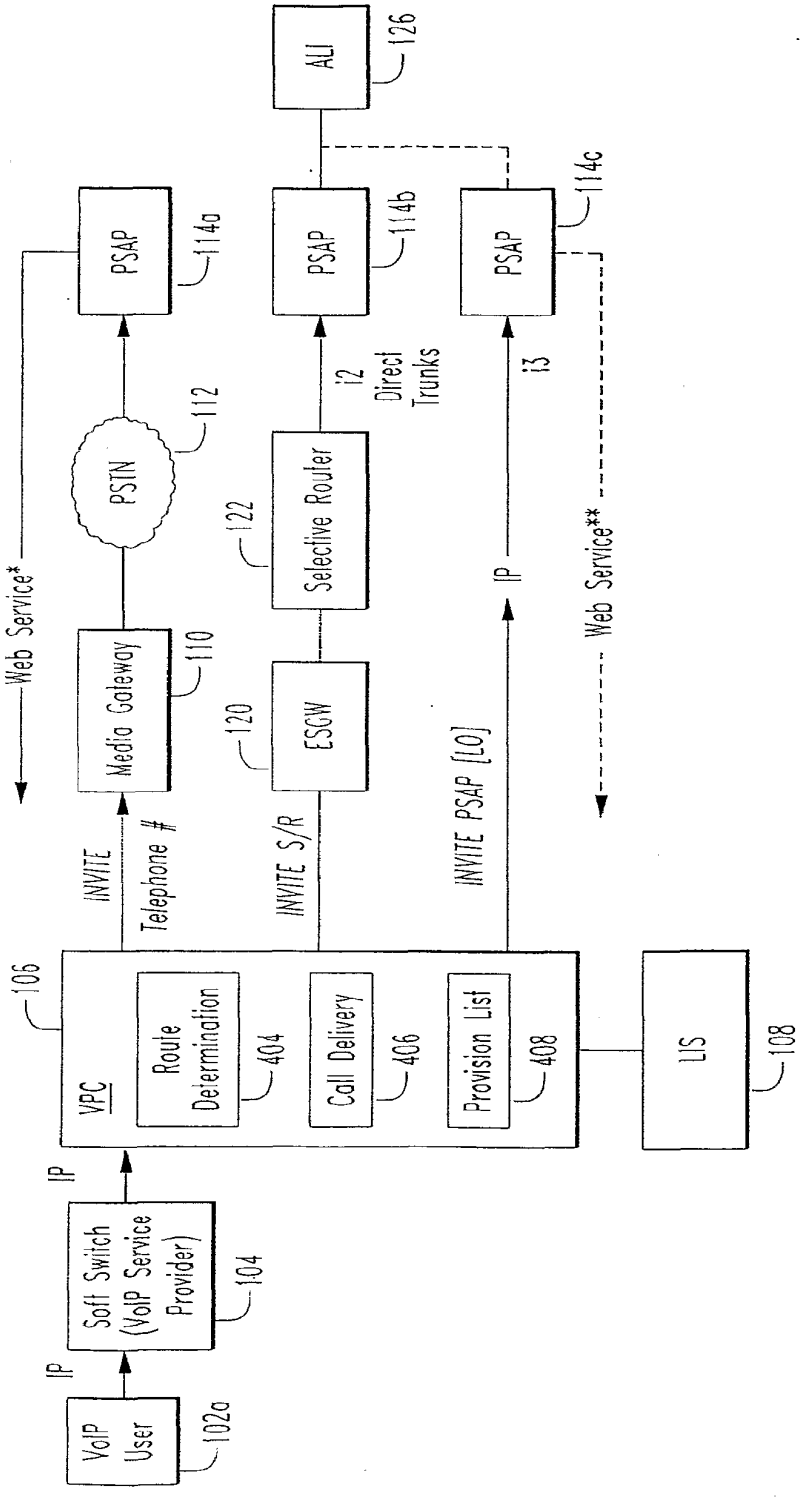


FIG. 7
(PRIOR ART)

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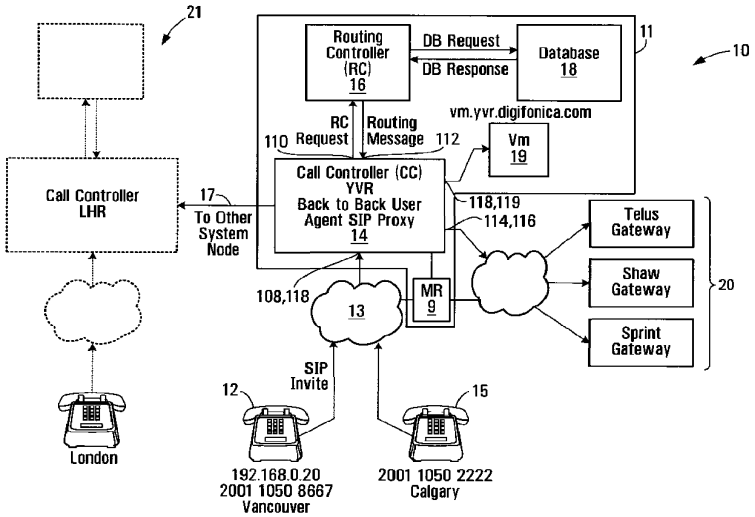
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[Continued on next page]

(54) Title: PRODUCING ROUTING MESSAGES FOR VOICE OVER IP COMMUNICATIONS



(57) Abstract: A process and apparatus to facilitate communication between callers and callees in a system comprising a plurality of nodes with which callers and callees are associated is disclosed. In response to initiation of a call by a calling subscriber, a caller identifier and a callee identifier are received. Call classification criteria associated with the caller identifier are used to classify the call as a public network call or a private network call. A routing message identifying an address, on the private network, associated with the callee is produced when the call is classified as a private network call and a routing message identifying a gateway to the public network is produced when the call is classified as a public network call.

WO 2008/052340 A1

PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

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PRODUCING ROUTING MESSAGES FOR VOICE OVER IP COMMUNICATIONS

BACKGROUND OF THE INVENTION

1. Field of Invention

This invention relates to voice over IP communications and methods and apparatus for routing and billing.

2. Description of Related Art

Internet protocol (IP) telephones are typically personal computer (PC) based telephones connected within an IP network, such as the public Internet or a private network of a large organization. These IP telephones have installed "voice-over-IP" (VoIP) software enabling them to make and receive voice calls and send and receive information in data and video formats.

IP telephony switches installed within the IP network enable voice calls to be made within or between IP networks, and between an IP network and a switched circuit network (SCN), such as the public switched telephone network (PSTN). If the IP switch supports the Signaling System 7 (SS7) protocol, the IP telephone can also access PSTN databases.

The PSTN network typically includes complex network nodes that contain all information about a local calling service area including user authentication and call routing. The PSTN network typically aggregates all information and traffic into a single location or node, processes it locally and then passes it on to other network nodes, as necessary, by maintaining route tables at the node. PSTN nodes are redundant by design and thus provide reliable service, but if a node should fail due to an earthquake or other natural disaster, significant, if not complete service outages can occur, with no other nodes being able to take up the load.

Existing VoIP systems do not allow for high availability and resiliency in delivering Voice Over IP based Session Initiation Protocol (SIP) Protocol service over a geographically dispersed area such as a city, region or continent. Most resiliency originates from the provision of IP based telephone services to one location or a small number of locations such as a single office or network of branch offices.

SUMMARY OF THE INVENTION

In accordance with one aspect of the invention, there is provided a process for operating a call routing controller to facilitate communication between callers and callees in a system comprising a plurality of nodes with which callers and callees are associated. The process involves, in response to initiation of a call by a calling subscriber, receiving a caller identifier and a callee identifier. The process also involves using call classification criteria associated with the caller identifier to classify the call as a public network call or a private network call. The process further involves producing a routing message identifying an address, on the private network, associated with the callee when the call is classified as a private network call. The process also involves producing a routing message identifying a gateway to the public network when the call is classified as a public network call.

The process may involve receiving a request to establish a call, from a call controller in communication with a caller identified by the callee identifier.

Using the call classification criteria may involve searching a database to locate a record identifying calling attributes associated with a caller identified by the caller identifier.

Locating a record may involve locating a caller dialing profile comprising a username associated with the caller, a domain associated with the caller, and at least one calling attribute.

Using the call classification criteria may involve comparing calling attributes associated with the caller dialing profile with aspects of the callee identifier.

Comparing may involve determining whether the callee identifier includes a portion that matches an IDD associated with the caller dialing profile.

Comparing may involve determining whether the callee identifier includes a portion that matches an NDD associated with the caller dialing profile.

Comparing may involve determining whether the callee identifier includes a portion that matches an area code associated with the caller dialing profile.

Comparing may involve determining whether the callee identifier has a length within a range specified in the caller dialing profile.

The process may involve formatting the callee identifier into a pre-defined digit format to produce a re-formatted callee identifier.

Formatting may involve removing an international dialing digit from the callee identifier, when the callee identifier begins with a digit matching an international dialing digit specified by the caller dialing profile associated with the caller.

Formatting may involve removing a national dialing digit from the callee identifier and prepending a caller country code to the callee identifier when the callee identifier begins with a national dialing digit.

Formatting may involve prepending a caller country code to the callee identifier when the callee identifier begins with digits identifying an area code specified by the caller dialing profile.

Formatting may involve prepending a caller country code and an area code to the callee identifier when the callee identifier has a length that matches a caller dialing number format specified by the caller dialing profile and only one area code is specified as being associated with the caller in the caller dialing profile.

The process may involve classifying the call as a private network call when the re-formatted callee identifier identifies a subscriber to the private network.

The process may involve determining whether the callee identifier complies with a pre-defined username format and if so, classifying the call as a private network call.

The process may involve causing a database of records to be searched to locate a direct in dial (DID) bank table record associating a public telephone number with the reformatted callee identifier and if the DID bank table record is found, classifying the call as a private network call and if a DID bank table record is not found, classifying the call as a public network call.

Producing the routing message identifying a node on the private network may involve setting a callee identifier in response to a username associated with the DID bank table record.

Producing the routing message may involve determining whether a node associated with the reformatted callee identifier is the same as a node associated the caller identifier.

Determining whether a node associated with the reformatted callee identifier is the same as a node associated the caller identifier may involve determining whether a prefix of the re-formatted callee identifier matches a corresponding prefix of a username associated with the caller dialing profile.

When the node associated with the caller is not the same as the node associated with the callee, the process involves producing a routing message including the caller identifier, the reformatted callee identifier and an identification of a private network node associated with the callee and communicating the routing message to a call controller.

When the node associated with the caller is the same as the node associated with the callee, the process involves determining whether to perform at least one of the following: forward the call to another party, block the call and direct the caller to a voicemail server associated with the callee.

Producing the routing message may involve producing a routing message having an identification of at least one of the callee identifier, an identification of a party to whom the call should be forwarded and an identification of a voicemail server associated with the callee.

The process may involve communicating the routing message to a call controller.

Producing a routing message identifying a gateway to the public network may involve searching a database of route records associating route identifiers with dialing codes to find a route record having a dialing code having a number pattern matching at least a portion of the reformatted callee identifier.

The process may involve searching a database of supplier records associating supplier identifiers with the route identifiers to locate at least one supplier record associated with the route identifier associated with the route record having a dialing code having a number pattern matching at least a portion of the reformatted callee identifier.

The process may involve loading a routing message buffer with the reformatted callee identifier and an identification of specific routes associated

respective ones of the supplier records associated with the route record and loading the routing message buffer with a time value and a timeout value.

The process may involve communicating a routing message involving the contents of the routing message buffer to a call controller.

The process may involve causing the dialing profile to include a maximum concurrent call value and a concurrent call count value and causing the concurrent call count value to be incremented when the user associated with the dialing profile initiates a call and causing the concurrent call count value to be decremented when a call with the user associated with the dialing profile is ended.

In accordance with another aspect of the invention, there is provided a call routing apparatus for facilitating communications between callers and callees in a system comprising a plurality of nodes with which callers and callees are associated. The apparatus includes receiving provisions for receiving a caller identifier and a callee identifier, in response to initiation of a call by a calling subscriber. The apparatus also includes classifying provisions for classifying the call as a private network cal or a public network call according to call classification criteria associated with the caller identifier. The apparatus further includes provisions for producing a routing message identifying an address, on the private network, associated with the callee when the call is classified as a private network call. The apparatus also includes provisions for producing a routing message identifying a gateway to the public network when the call is classified as a public network call.

The receiving provisions may be operably configured to receive a request to establish a call, from a call controller in communication with a caller identified by the callee identifier.

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The apparatus may further include searching provisions for searching a database including records associating calling attributes with subscribers to the private network to locate a record identifying calling attributes associated with a caller identified by the caller identifier.

The records may include dialing profiles each including a username associated with the subscriber, an identification of a domain associated with the subscriber, and an identification of at least one calling attribute associated with the subscriber.

The call classification provisions may be operably configured to compare calling attributes associated with the caller dialing profile with aspects of the callee identifier.

The calling attributes may include an international dialing digit and call classification provisions may be operably configured to determine whether the callee identifier includes a portion that matches an IDD associated with the caller dialing profile.

The calling attributes may include an national dialing digit and the call classification provisions may be operably configured to determine whether the callee identifier includes a portion that matches an NDD associated with the caller dialing profile.

The calling attributes may include an area code and the call classification provisions may be operably configured to determine whether the callee identifier includes a portion that matches an area code associated with the caller dialing profile.

The calling attribute may include a number length range and the call classification provisions may be operably configured to determine whether the

callee identifier has a length within a number length range specified in the caller dialing profile.

The apparatus may further include formatting provisions for formatting the callee identifier into a pre-defined digit format to produce a re-formatted callee identifier.

The formatting provisions may be operably configured to remove an international dialing digit from the callee identifier, when the callee identifier begins with a digit matching an international dialing digit specified by the caller dialing profile associated with the caller.

The formatting provisions may be operably configured to remove a national dialing digit from the callee identifier and prepend a caller country code to the callee identifier when the callee identifier begins with a national dialing digit.

The formatting provisions may be operably configured to prepend a caller country code to the callee identifier when the callee identifier begins with digits identifying an area code specified by the caller dialing profile.

The formatting provisions may be operably configured to prepend a caller country code and area code to the callee identifier when the callee identifier has a length that matches a caller dialing number format specified by the caller dialing profile and only one area code is specified as being associated with the caller in the caller dialing profile.

The classifying provisions may be operably configured to classify the call as a private network call when the re-formatted callee identifier identifies a subscriber to the private network.

The classifying provisions may be operably configured to classify the call as a private network call when the callee identifier complies with a pre-defined username format.

The apparatus may further include searching provisions for searching a database of records to locate a direct in dial (DID) bank table record associating a public telephone number with the reformatted callee identifier and the classifying provisions may be operably configured to classify the call as a private network call when the DID bank table record is found and to classify the call as a public network call when a DID bank table record is not found

The private network routing message producing provisions may be operably configured to produce a routing message having a callee identifier set according to a username associated with the DID bank table record.

The private network routing message producing provisions may be operably configured to determine whether a node associated with the reformatted callee identifier is the same as a node associated the caller identifier.

The private network routing provisions may include provisions for determining whether a prefix of the re-formatted callee identifier matches a corresponding prefix of a username associated with the caller dialing profile.

The private network routing message producing provisions may be operably configured to produce a routing message including the caller identifier, the reformatted callee identifier and an identification of a private network node associated with the callee and to communicate the routing message to a call controller.

The private network routing message producing provisions may be operably configured to perform at least one of the following forward the call to another

party, block the call and direct the caller to a voicemail server associated with the callee, when the node associated with the caller is the same as the node associated with the callee.

The provisions for producing the private network routing message may be operably configured to produce a routing message having an identification of at least one of the callee identifier, an identification of a party to whom the call should be forwarded and an identification of a voicemail server associated with the callee.

The apparatus further includes provisions for communicating the routing message to a call controller.

The provisions for producing a public network routing message identifying a gateway to the public network may include provisions for searching a database of route records associating route identifiers with dialing codes to find a route record having a dialing code having a number pattern matching at least a portion of the reformatted callee identifier.

The apparatus further includes provisions for searching a database of supplier records associating supplier identifiers with the route identifiers to locate at least one supplier record associated with the route identifier associated with the route record having a dialing code having a number pattern matching at least a portion of the reformatted callee identifier.

The apparatus further includes a routing message buffer and provisions for loading the routing message buffer with the reformatted callee identifier and an identification of specific routes associated respective ones of the supplier records associated with the route record and loading the routing message buffer with a time value and a timeout value.

The apparatus further includes provisions for communicating a routing message including the contents of the routing message buffer to a call controller.

The apparatus further includes means for causing said dialing profile to include a maximum concurrent call value and a concurrent call count value and for causing said concurrent call count value to be incremented when the user associated with said dialing profile initiates a call and for causing said concurrent call count value to be decremented when a call with said user associated with said dialing profile is ended.

In accordance with another aspect of the invention, there is provided a data structure for access by an apparatus for producing a routing message for use by a call routing controller in a communications system. The data structure includes dialing profile records comprising fields for associating with respective subscribers to the system, a subscriber user name, direct-in-dial records comprising fields for associating with respective subscriber usernames, a user domain and a direct-in-dial number, prefix to node records comprising fields for associating with at least a portion of the respective subscriber usernames, a node address of a node in the system, whereby a subscriber name can be used to find a user domain, at least a portion of the a subscriber name can be used to find a node with which the subscriber identified by the subscriber name is associated, and a user domain and subscriber name can be located in response to a direct-in-dial number.

In accordance with another aspect of the invention, there is provided a data structure for access by an apparatus for producing a routing message for use by a call routing controller in a communications system. The data structure includes master list records comprising fields for associating a dialing code with respective master list identifiers and supplier list records linked to master list records by the master list identifiers, said supplier list records comprising fields for associating with a communications services supplier, a supplier id, a

master list id, a route identifier and a billing rate code, whereby communications services suppliers are associated with dialing codes, such that dialing codes can be used to locate suppliers capable of providing a communications link associated with a given dialing code.

In accordance with another aspect of the invention, there is provided a method for determining a time to permit a communication session to be conducted. The method involves calculating a cost per unit time, calculating a first time value as a sum of a free time attributed to a participant in the communication session and the quotient of a funds balance held by the participant to the cost per unit time value and producing a second time value in response to the first time value and a billing pattern associated with the participant, the billing pattern including first and second billing intervals and the second time value being the time to permit a communication session to be conducted.

Calculating the first time value may involve retrieving a record associated with the participant and obtaining from the record at least one of the free time and the funds balance.

Producing the second time value may involve producing a remainder value representing a portion of the second billing interval remaining after dividing the second billing interval into a difference between the first time value and the first billing interval.

Producing the second time value may involve setting a difference between the first time value and the remainder as the second time value.

The method may further involve setting the second time value to zero when the remainder is greater than zero and the first time value is less than the free time associated with the participant.

Calculating the cost per unit time may involve locating a record in a database, the record comprising a markup type indicator, a markup value and a billing pattern and setting a reseller rate equal to the sum of the markup value and the buffer rate.

Locating the record in a database may involve locating at least one of a record associated with a reseller and a route associated with the reseller, a record associated with the reseller and a default reseller markup record.

Calculating the cost per unit time value further may involve locating at least one of an override record specifying a route cost per unit time amount associated with a route associated with the communication session, a reseller record associated with a reseller of the communications session, the reseller record specifying a reseller cost per unit time associated with the reseller for the communication session, a default operator markup record specifying a default cost per unit time.

The method may further involve setting as the cost per unit time the sum of the reseller rate and at least one of the route cost per unit time, the reseller cost per unit time and the default cost per unit time.

The method may further involve receiving a communication session time representing a duration of the communication session and incrementing a reseller balance by the product of the reseller rate and the communication session time.

The method may further involve receiving a communication session time representing a duration of the communication session and incrementing a system operator balance by a product of the buffer rate and the communication session time.

In accordance with another aspect of the invention, there is provided an apparatus for determining a time to permit a communication session to be conducted. The apparatus includes a processor circuit, a computer readable medium coupled to the processor circuit and encoded with instructions for directing the processor circuit to calculate a cost per unit time for the communication session, calculate a first time value as a sum of a free time attributed to a participant in the communication session and the quotient of a funds balance held by the participant to the cost per unit time value and produce a second time value in response to the first time value and a billing pattern associated with the participant, the billing pattern including first and second billing intervals and the second time value being the time to permit a communication session to be conducted.

The instructions may include instructions for directing the processor circuit to retrieve a record associated with the participant and obtain from the record at least one of the free time and the funds balance.

The instructions may include instructions for directing the processor circuit to produce the second time value by producing a remainder value representing a portion of the second billing interval remaining after dividing the second billing interval into a difference between the first time value and the first billing interval.

The instructions may include instructions for directing the processor circuit to produce the second time value comprises setting a difference between the first time value and the remainder as the second time value.

The instructions may include instructions for directing the processor circuit to set the second time value to zero when the remainder is greater than zero and the first time value is less than the free time associated with the participant.

The instructions for directing the processor circuit to calculate the cost per unit time may include instructions for directing the processor circuit to locate a record in a database, the record comprising a markup type indicator, a markup value and a billing pattern and set a reseller rate equal to the sum of the markup value and the buffer rate.

The instructions for directing the processor circuit to locate the record in a database may include instructions for directing the processor circuit to locate at least one of a record associated with a reseller and a route associated with the reseller, a record associated with the reseller, and a default reseller markup record. The instructions for directing the processor circuit to calculate the cost per unit time value may further include instructions for directing the processor circuit to locate at least one of an override record specifying a route cost per unit time amount associated with a route associated with the communication session, a reseller record associated with a reseller of the communications session, the reseller record specifying a reseller cost per unit time associated with the reseller for the communication session, a default operator markup record specifying a default cost per unit time.

The instructions may include instructions for directing the processor circuit to set as the cost per unit time the sum of the reseller rate and at least one of the route cost per unit time, the reseller cost per unit time and the default cost per unit time.

The instructions may include instructions for directing the processor circuit to receive a communication session time representing a duration of the communication session and increment a reseller balance by the product of the reseller rate and the communication session time.

The instructions may include instructions for directing the processor circuit to receive a communication session time representing a duration of the

communication session and increment a system operator balance by a product of the buffer rate and the communication session time.

In accordance with another aspect of the invention, there is provided a process for attributing charges for communications services. The process involves determining a first chargeable time in response to a communication session time and a pre-defined billing pattern, determining a user cost value in response to the first chargeable time and a free time value associated with a user of the communications services, changing an account balance associated with the user in response to a user cost per unit time. The process may further involve changing an account balance associated with a reseller of the communications services in response to a reseller cost per unit time and the communication session time and changing an account balance associated with an operator of the communications services in response to an operator cost per unit time and the communication session time.

Determining the first chargeable time may involve locating at least one of an override record specifying a route cost per unit time and billing pattern associated with a route associated with the communication session, a reseller record associated with a reseller of the communications session, the reseller record specifying a reseller cost per unit time and billing pattern associated with the reseller for the communication session and a default record specifying a default cost per unit time and billing pattern and setting as the pre-defined billing pattern the billing pattern of the record located. The billing pattern of the record located may involve a first billing interval and a second billing interval.

Determining the first chargeable time may involve setting the first chargeable time equal to the first billing interval when the communication session time is less than or equal to the first billing interval.

Determining the first chargeable time may involve producing a remainder value representing a portion of the second billing interval remaining after dividing the second billing interval into a difference between communication session time and the first interval when the communication session time is greater than the communication session time and setting the first chargeable time to a difference between the communication session time and the remainder when the remainder is greater than zero and setting the first chargeable time to the communication session time when the remainder is not greater than zero.

The process may further involve determining a second chargeable time in response to the first chargeable time and the free time value associated with the user of the communications services when the first chargeable time is greater than or equal to the free time value associated with the user of the communications services.

Determining the second chargeable time may involve setting the second chargeable time to a difference between the first chargeable time.

The process may further involve resetting the free time value associated with the user to zero when the first chargeable time is greater than or equal to the free time value associated with the user of the communications services.

Changing an account balance associated with the user may involve calculating a user cost value in response to the second chargeable time and the user cost per unit time.

The process may further involve changing a user free cost balance in response to the user cost value.

The process may further involve setting the user cost to zero when the first chargeable time is less than the free time value associated with the user.

The process may further involve changing a user free time balance in response to the first chargeable time.

In accordance with another aspect of the invention, there is provided an apparatus for attributing charges for communications services. The apparatus includes a processor circuit, a computer readable medium in communication with the processor circuit and encoded with instructions for directing the processor circuit to determine a first chargeable time in response to a communication session time and a pre-defined billing pattern, determine a user cost value in response to the first chargeable time and a free time value associated with a user of the communications services, change an account balance associated with the user in response to a user cost per unit time.

The instructions may further include instructions for changing an account balance associated with a reseller of the communications services in response to a reseller cost per unit time and the communication session time and changing an account balance associated with an operator of the communications services in response to an operator cost per unit time and the communication session time.

The instructions for directing the processor circuit to determine the first chargeable time may further include instructions for causing the processor circuit to communicate with a database to locate at least one of an override record specifying a route cost per unit time and billing pattern associated with a route associated with the communication session, a reseller record associated with a reseller of the communications session, the reseller record specifying a reseller cost per unit time and billing pattern associated with the reseller for the communication session and a default record specifying a default cost per unit time and billing pattern and instructions for setting as the pre-defined billing pattern the billing pattern of the record located. The billing

pattern of the record located may include a first billing interval and a second billing interval.

The instructions for causing the processor circuit to determine the first chargeable time may include instructions for directing the processor circuit to set the first chargeable time equal to the first billing interval when the communication session time is less than or equal to the first billing interval.

The instructions for causing the processor circuit to determine the first chargeable time may include instructions for producing a remainder value representing a portion of the second billing interval remaining after dividing the second billing interval into a difference between communication session time and the first interval when the communication session time is greater than the communication session time and instructions for causing the processor circuit to set the first chargeable time to a difference between the communication session time and the remainder when the remainder is greater than zero and instructions for causing the processor circuit to set the first chargeable time to the communication session time when the remainder is not greater than zero.

The instructions may further include instructions for causing the processor circuit to determine a second chargeable time in response to the first chargeable time and the free time value associated with the user of the communications services when the first chargeable time is greater than or equal to the free time value associated with the user of the communications services.

The instructions for causing the processor circuit to determine the second chargeable time may include instructions for causing the processor circuit to set the second chargeable time to a difference between the first chargeable time.

The instructions may further include instructions for causing the processor circuit to reset the free time value associated with the user to zero when the first chargeable time is greater than or equal to the free time value associated with the user of the communications services.

The instructions for causing the processor circuit to change an account balance associated with the user may include instructions for causing the processor circuit to calculate a user cost value in response to the second chargeable time and the user cost per unit time.

The instructions may further include instructions for causing the processor circuit to change a user free cost balance in response to the user cost value.

The instructions may further include instructions for causing the processor circuit to set the user cost to zero when the first chargeable time is less than the free time value associated with the user.

The instructions may further include instructions for causing the processor circuit to change a user free time balance in response to the first chargeable time.

In accordance with another aspect of the invention, there is provided a computer readable medium encoded with codes for directing a processor circuit to execute one or more of the methods described above and/or variants thereof.

Other aspects and features of the present invention will become apparent to those ordinarily skilled in the art upon review of the following description of specific embodiments of the invention in conjunction with the accompanying figures.

BRIEF DESCRIPTION OF THE DRAWINGS

In drawings which illustrate embodiments of the invention,

Figure **1** is a block diagram of a system according to a first embodiment of the invention;

Figure **2** is a block diagram of a caller telephone according to the first embodiment of the invention;

Figure **3** is a schematic representation of a SIP invite message transmitted between the caller telephone and a controller shown in Figure **1**;

Figure **4** is a block diagram of a call controller shown in Figure **1**;

Figure **5** is a flowchart of a process executed by the call controller shown in Figure **1**;

Figure **6** is a schematic representation of a routing, billing and rating (RC) request message produced by the call controller shown in Figure **1**;

Figure **7** is a block diagram of a processor circuit of a routing, billing, rating element of the system shown in Figure **1**;

Figures **8A-8D** is a flowchart of a RC request message handler executed by the RC processor circuit shown in Figure **7**;

Figure **9** is a tabular representation of a dialing profile stored in a database accessible by the RC shown in Figure **1**;

Figure **10** is a tabular representation of a dialing profile for a caller using the caller telephone shown in Figure **1**;

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Figure **11** is a tabular representation of a callee profile for a callee located in Calgary;

Figure **12** is a tabular representation of a callee profile for a callee located in London;

Figure **13** is a tabular representation of a Direct-in-Dial (DID) bank table record stored in the database shown in Figure **1**;

Figure **14** is a tabular representation of an exemplary DID bank table record for the Calgary callee referenced in Figure **11**;

Figure **15** is a tabular representation of a routing message transmitted from the RC to the call controller shown in Figure **1**;

Figure **16** is a schematic representation of a routing message buffer holding a routing message for routing a call to the Calgary callee referenced in Figure **11**;

Figure **17** is a tabular representation of a prefix to supernode table record stored in the database shown in Figure **1**;

Figure **18** is a tabular representation of a prefix to supernode table record that would be used for the Calgary callee referenced in Figure **11**;

Figure **19** is a tabular representation of a master list record stored in a master list table in the database shown in Figure **1**;

Figure **20** is a tabular representation of a populated master list record;

Figure **21** is a tabular representation of a suppliers list record stored in the database shown in Figure **1**;

Figure **22** is a tabular representation of a specific supplier list record for a first supplier;

Figure **23** is a tabular representation of a specific supplier list record for a second supplier;

Figure **24** is a tabular representation of a specific supplier list record for a third supplier;

Figure **25** is a schematic representation of a routing message, held in a routing message buffer, identifying to the controller a plurality of possible suppliers that may carry the call;

Figure **26** is a tabular representation of a call block table record;

Figure **27** is a tabular representation of a call block table record for the Calgary callee;

Figure **28** is a tabular representation of a call forwarding table record;

Figure **29** is a tabular representation of a call forwarding table record specific for the Calgary callee;

Figure **30** is a tabular representation of a voicemail table record specifying voicemail parameters to enable the caller to leave a voicemail message for the callee;

Figure **31** is a tabular representation of a voicemail table record specific to the Calgary callee;

Figure **32** is a schematic representation of an exemplary routing message, held in a routing message buffer, indicating call forwarding numbers and a voicemail server identifier;

Figures **33**A and **33**B are respective portions of a flowchart of a process executed by the RC processor for determining a time to live value;

Figure **34** is a tabular representation of a subscriber bundle table record;

Figure **35** is a tabular representation of a subscriber bundle record for the Vancouver caller;

Figure **36** is a tabular representation of a bundle override table record;

Figure **37** is a tabular representation of bundle override record for a located master list ID;

Figure **38** is a tabular representation of a subscriber account table record;

Figure **39** is a tabular representation of a subscriber account record for the Vancouver caller;

Figure **40** is a flowchart of a process for producing a second time value executed by the RC processor circuit shown in Figure **7**;

Figure **41** is a flowchart for calculating a call cost per unit time;

Figure **42** is a tabular representation of a system operator special rates table record;

Figure **43** is a tabular representation of a system operator special rates table record for a reseller named Klondike;

Figure **44** is a tabular representation of a system operator mark-up table record;

Figure **45** is a tabular representation of a system operator mark-up table record for the reseller Klondike;

Figure **46** is a tabular representation of a default system operator mark-up table record;

Figure **47** is a tabular representation of a reseller special destinations table record;

Figure **48** is a tabular representation of a reseller special destinations table record for the reseller Klondike;

Figure **49** is a tabular representation of a reseller global mark-up table record;

Figure **50** is a tabular representation of a reseller global mark-up table record for the reseller Klondike;

Figure **51** is a tabular representation of a SIP bye message transmitted from either of the telephones shown in Figure **1** to the call controller;

Figure **52** is a tabular representation of a SIP bye message sent to the controller from the Calgary callee;

Figure **53** is a flowchart of a process executed by the call controller for producing a RC stop message in response to receipt of a SIP bye message;

Figure **54** is a tabular representation of an exemplary RC call stop message;

Figure **55** is a tabular representation of an RC call stop message for the Calgary callee;

Figures **56**A and **56**B are respective portions of a flowchart of a RC call stop message handling routine executed by the RC shown in Figure **1**;

Figure **57** is a tabular representation of a reseller accounts table record;

Figure **58** is a tabular representation of a reseller accounts table record for the reseller Klondike;

Figure **59** is a tabular representation of a system operator accounts table record; and

Figure **60** is a tabular representation of a system operator accounts record for the system operator described herein.

DETAILED DESCRIPTION

Referring to Figure **1**, a system for making voice over IP telephone/videophone calls is shown generally at **10**. The system includes a first super node shown generally at **11** and a second super node shown generally at **21**. The first super node **11** is located in geographical area, such as Vancouver, B.C., Canada for example and the second super node **21** is located in London, England, for example. Different super nodes may be located in different geographical regions throughout the world to provide

telephone/videophone service to subscribers in respective regions. These super nodes may be in communication with each other by high speed/ high data throughput links including optical fiber, satellite and/or cable links, forming a backbone to the system. These super nodes may alternatively or, in addition, be in communication with each other through conventional internet services.

In the embodiment shown, the Vancouver supernode **11** provides telephone/videophone service to western Canadian customers from Vancouver Island to Ontario. Another node (not shown) may be located in Eastern Canada to provide services to subscribers in that area.

Other nodes of the type shown may also be employed within the geographical area serviced by a supernode, to provide for call load sharing, for example within a region of the geographical area serviced by the supernode. However, in general, all nodes are similar and have the properties described below in connection with the Vancouver supernode **11**.

In this embodiment, the Vancouver supernode includes a call controller (C) **14**, a routing controller (RC) **16,** a database **18** and a voicemail server **19** and a media relay **9**. Each of these may be implemented as separate modules on a common computer system or by separate computers, for example. The voicemail server **19** need not be included in the node and can be provided by an outside service provider.

Subscribers such as a subscriber in Vancouver and a subscriber in Calgary communicate with the Vancouver supernode using their own internet service providers which route internet traffic from these subscribers over the internet shown generally at **13** in Figure **1**. To these subscribers the Vancouver supernode is accessible at a pre-determined internet protocol (IP) address or a fully qualified domain name that can be accessed in the usual way through a subscriber's internet service provider. The subscriber in Vancouver uses a

telephone **12** that is capable of communicating with the Vancouver supernode **11** using Session Initiation Protocol (SIP) messages and the Calgary subscriber uses a similar telephone **15**, in Calgary AB.

It should be noted that throughout the description of the embodiments of this invention, the IP/UDP addresses of all elements such as the caller and callee telephones, call controller, media relay, and any others, will be assumed to be valid IP/UDP addresses directly accessible via the Internet or a private IP network, for example, depending on the specific implementation of the system. As such, it will be assumed, for example, that the caller and callee telephones will have IP/UDP addresses directly accessible by the call controllers and the media relays on their respective supernodes, and those addresses will not be obscured by Network Address Translation (NAT) or similar mechanisms. In other words, the IP/UDP information contained in SIP messages (for example the SIP Invite message or the RC Request message which will be described below) will match the IP/UDP addresses of the IP packets carrying these SIP messages.

It will be appreciated that in many situations, the IP addresses assigned to various elements of the system may be in a private IP address space, and thus not directly accessible from other elements. Furthermore, it will also be appreciated that NAT is commonly used to share a "public" IP address between multiple devices, for example between home PCs and IP telephones sharing a single Internet connection. For example, a home PC may be assigned an IP address such as **192.168.0.101** and a Voice over IP telephone may be assigned an IP address of **192.168.0.103**. These addresses are located in so called "non-routable" (IP) address space and cannot be accessed directly from the Internet. In order for these devices to communicate with other computers located on the Internet, these IP addresses have to be converted into a "public" IP address, for example **24.10.10.123** assigned by the Internet Service Provider to the subscriber, by a device performing NAT, typically a home router. In addition to translating the IP addresses, NAT

typically also translates UDP port numbers, for example an audio path originating at a VoIP telephone and using a UDP port **12378** at its private IP address, may have be translated to a UDP port **23465** associated with the public IP address of the NAT device. In other words, when a packet originating from the above VoIP telephone arrives at an Internet-based supernode, the source IP/UDP address contained in the IP packet header will be **24.10.10.1:23465**, whereas the source IP/UDP address information contained in the SIP message inside this IP packet will be **192.168.0.103:12378**. The mismatch in the IP/UDP addresses may cause a problem for SIP-based VoIP systems because, for example, a supernode will attempt to send messages to a private address of a telephone but the messages will never get there.

Referring to Figure **1**, in an attempt to make a call by the Vancouver telephone/videophone **12** to the Calgary telephone/videophone **15**, the Vancouver telephone/videophone sends a SIP invite message to the Vancouver supernode **11** and in response, the call controller **14** sends an RC request message to the RC **16** which makes various enquiries of the database **18** to produce a routing message which is sent back to the call controller **14**. The call controller **14** then communicates with the media relay **9** to cause a communications link including an audio path and a videophone (if a videopath call) to be established through the media relay to the same node, a different node or to a communications supplier gateway as shown generally at **20** to carry audio, and where applicable, video traffic to the call recipient or callee.

Generally, the RC **16** executes a process to facilitate communication between callers and callees. The process involves, in response to initiation of a call by a calling subscriber, receiving a callee identifier from the calling subscriber, using call classification criteria associated with the calling subscriber to classify the call as a public network call or a private network call and producing a routing message identifying an address on the private network,

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associated with the callee when the call is classified as a private network call and producing a routing message identifying a gateway to the public network when the call is classified as a public network call.

Subscriber Telephone

In greater detail, referring to Figure **2**, in this embodiment, the telephone/videophone **12** includes a processor circuit shown generally at **30** comprising a microprocessor **32**, program memory **34**, an input/output (I/O) port **36**, parameter memory **38** and temporary memory **40**. The program memory **34**, I/O port **36**, parameter memory **38** and temporary memory **40** are all in communication with the microprocessor **32**. The I/O port **36** has a dial input **42** for receiving a dialled telephone/videophone number from a keypad, for example, or from a voice recognition unit or from pre-stored telephone/videophone numbers stored in the parameter memory **38**, for example. For simplicity, in Figure **2** a box labelled dialing functions **44** represents any device capable of informing the microprocessor **32** of a callee identifier, e.g., a callee telephone/videophone number.

The processor **32** stores the callee identifier in a dialled number buffer **45**. In this case, assume the dialled number is **2001 1050 2222** and that it is a number associated with the Calgary subscriber. The I/O port **36** also has a handset interface **46** for receiving and producing signals from and to a handset that the user may place to his ear. This interface **46** may include a BLUETOOTH™ wireless interface, a wired interface or speaker phone, for example. The handset acts as a termination point for an audio path (not shown) which will be appreciated later. The I/O port **36** also has an internet connection **48** which is preferably a high speed internet connection and is operable to connect the telephone/videophone to an internet service provider. The internet connection **48** also acts as a part of the voice path, as will be appreciated later. It will be appreciated that where the subscriber device is a videophone, a separate video path is established in the same way an audio path is established. For simplicity, the following description refers to a

telephone call, but it is to be understood that a videophone call is handled similarly, with the call controller causing the media relay to facilitate both an audio path and a video path instead of only an audio path.

The parameter memory **38** has a username field **50**, a password field **52** an IP address field **53** and a SIP proxy address field **54**, for example. The user name field **50** is operable to hold a user name, which in this case is **2001 1050 8667**. The user name is assigned upon subscription or registration into the system and, in this embodiment, includes a twelve digit number having a continent code **61**, a country code **63**, a dealer code **70** and a unique number code **74**. The continent code **61** is comprised of the first or left-most digit of the user name in this embodiment. The country code **63** is comprised of the next three digits. The dealer code **70** is comprised of the next four digits and the unique number code **74** is comprised of the last four digits. The password field **52** holds a password of up to **512** characters, in this example. The IP address field **53** stores an IP address of the telephone, which for this explanation is **192.168.0.20**. The SIP proxy address field **54** holds an IP protocol compatible proxy address which may be provided to the telephone through the internet connection **48** as part of a registration procedure.

The program memory **34** stores blocks of codes for directing the processor **32** to carry out the functions of the telephone, one of which includes a firewall block **56** which provides firewall functions to the telephone, to prevent access by unauthorized persons to the microprocessor **32** and memories **34**, **38** and **40** through the internet connection **48**. The program memory **34** also stores codes **57** for establishing a call ID. The call ID codes **57** direct the processor **32** to produce a call identifier having a format comprising a hexadecimal string at an IP address, the IP address being the IP address of the telephone. Thus, an exemplary call identifier might be FF**10@192.168.0.20**.

Generally, in response to picking up the handset interface **46** and activating a dialing function **44**, the microprocessor **32** produces and sends a SIP invite

message as shown in Figure **3**, to the routing controller **16** shown in Figure **1**. This SIP invite message is essentially to initiate a call by a calling subscriber.

Referring to Figure **3**, the SIP invite message includes a caller ID field **60**, a callee identifier field **62**, a digest parameters field **64**, a call ID field **65** an IP address field **67** and a caller UDP port field **69**. In this embodiment, the caller ID field **60** includes the user name **2001 1050 8667** that is the Vancouver user name stored in the user name field **50** of the parameter memory **38** in the telephone **12** shown in Figure **2**. In addition, referring back to Figure **3**, the callee identifier field **62** includes a callee identifier which in this embodiment is the user name **2001 1050 2222** that is the dialled number of the Calgary subscriber stored in the dialled number buffer **45** shown in Figure **2**. The digest parameters field **64** includes digest parameters and the call ID field **65** includes a code comprising a generated prefix code (FF**10**) and a suffix which is the Internet Protocol (IP) address of the telephone **12** stored in the IP address field **53** of the telephone. The IP address field **67** holds the IP address assigned to the telephone, in this embodiment **192.168.0.20**, and the caller UDP port field **69** includes a UDP port identifier identifying a UDP port at which the audio path will be terminated at the caller's telephone.

Call Controller

Referring to Figure **4**, a call controller circuit of the call controller **14** (Figure **1**) is shown in greater detail at **100**. The call controller circuit **100** includes a microprocessor **102**, program memory **104** and an I/O port **106**. The circuit **100** may include a plurality of microprocessors, a plurality of program memories and a plurality of I/O ports to be able to handle a large volume of calls. However, for simplicity, the call controller circuit **100** will be described as having only one microprocessor **102**, program memory **104** and I/O port **106**, it being understood that there may be more.

Generally, the I/O port **106** includes an input **108** for receiving messages such as the SIP invite message shown in Figure **3**, from the telephone shown in

Figure **2**. The I/O port **106** also has an RC request message output **110** for transmitting an RC request message to the RC **16** of Figure **1**, an RC message input **112** for receiving routing messages from the RC **16**, a gateway output **114** for transmitting messages to one of the gateways **20** shown in Figure **1** to advise the gateway to establish an audio path, for example, and a gateway input **116** for receiving messages from the gateway. The I/O port **106** further includes a SIP output **118** for transmitting messages to the telephone **12** to advise the telephone of the IP addresses of the gateways which will establish the audio path. The I/O port **106** further includes a voicemail server input and output **117**, **119** respectively for communicating with the voicemail server **19** shown in Figure **1**.

While certain inputs and outputs have been shown as separate, it will be appreciated that some may be a single IP address and IP port. For example, the messages sent to the RC **16** and received from the RC **16** may be transmitted and received on the same single IP port.

The program memory **104** includes blocks of code for directing the microprocessor **102** to carry out various functions of the call controller **14**. For example, these blocks of code include a first block **120** for causing the call controller circuit **100** to execute a SIP invite to RC request process to produce an RC request message in response to a received SIP invite message. In addition, there is a routing message to gateway message block **122** which causes the call controller circuit **100** to produce a gateway query message in response to a received routing message from the RC **16**.

Referring to Figure **5**, the SIP invite to RC request process is shown in more detail at **120**. On receipt of a SIP invite message of the type shown in Figure **3**, block **122** of Figure **5** directs the call controller circuit **100** of Figure **4** to authenticate the user. This may be done, for example, by prompting the user for a password, by sending a message back to the telephone **12** which is interpreted at the telephone as a request for a password entry or the

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password may automatically be sent to the call controller **14** from the telephone, in response to the message. The call controller **14** may then make enquiries of databases to which it has access, to determine whether or not the user's password matches a password stored in the database. Various functions may be used to pass encryption keys or hash codes back and forth to ensure that the transmittal of passwords is secure.

Should the authentication process fail, the call controller circuit **100** is directed to an error handling routine **124** which causes messages to be displayed at the telephone **12** to indicate there was an authentication problem. If the authentication procedure is passed, block **121** directs the call controller circuit **100** to determine whether or not the contents of the caller ID field **60** of the SIP invite message received from the telephone is an IP address. If it is an IP address, then block **123** directs the call controller circuit **100** to set the contents of a type field variable maintained by the microprocessor **102** to a code representing that the call type is a third party invite. If at block **121** the caller ID field contents do not identify an IP address, then block **125** directs the microprocessor to set the contents of the type field to a code indicating that the call is being made by a system subscriber. Then, block **126** directs the call controller circuit to read the call identifier **65** provided in the SIP invite message from the telephone **12**, and at block **128** the processor is directed to produce an RC request message that includes that call ID. Block **129** then directs the call controller circuit **100** to send the RC request to the RC **16**.

Referring to Figure **6**, an RC request message is shown generally at **150** and includes a caller field **152**, a callee field **154**, a digest field **156**, a call ID field **158** and a type field **160**. The caller, callee, digest call ID fields **152**, **154**, **156** and **158** contain copies of the caller, callee, digest parameters and call ID fields **60**, **62**, **64** and **65** of the SIP invite message shown in Figure **3**. The type field **160** contains the type code established at blocks **123** or **125** of Figure **5** to indicate whether the call is from a third party or system subscriber,

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respectively. The caller identifier field may include a PSTN number or a system subscriber username as shown, for example.

Routing Controller (RC)

Referring to Figure **7**, the RC **16** is shown in greater detail and includes an RC processor circuit shown generally at **200**. The RC processor circuit **200** includes a processor **202**, program memory **204**, a table memory **206**, buffer memory **207**, and an I/O port **208**, all in communication with the processor **202**. (As earlier indicated, there may be a plurality of processor circuits (**202**), memories (**204**), etc.)

The buffer memory **207** includes a caller id buffer **209** and a callee id buffer **211**.

The I/O port **208** includes a database request port **210** through which a request to the database (**18** shown in Figure **1**) can be made and includes a database response port **212** for receiving a reply from the database **18**. The I/O port **208** further includes an RC request message input **214** for receiving the RC request message from the call controller (**14** shown in Figure **1**) and includes a routing message output **216** for sending a routing message back to the call controller **14**. The I/O port **208** thus acts to receive caller identifier and a callee identifier contained in the RC request message from the call controller, the RC request message being received in response to initiation of a call by a calling subscriber.

The program memory **204** includes blocks of codes for directing the processor **202** to carry out various functions of the RC (**16**). One of these blocks includes an RC request message handler **250** which directs the RC to produce a routing message in response to a received RC request message. The RC request message handler process is shown in greater detail at **250** in Figures **8**A through **8**D.

RC Request Message Handler

Referring to Figure **8**A, the RC request message handler begins with a first block **252** that directs the RC processor circuit (**200**) to store the contents of the RC request message (**150**) in buffers in the buffer memory **207** of Figure **7**, one of which includes the caller ID buffer **209** of Figure **7** for separately storing the contents of the callee field **154** of the RC request message. Block **254** then directs the RC processor circuit to use the contents of the caller field **152** in the RC request message shown in Figure **6**, to locate and retrieve from the database **18** a record associating calling attributes with the calling subscriber. The located record may be referred to as a dialing profile for the caller. The retrieved dialing profile may then be stored in the buffer memory **207**, for example.

Referring to Figure **9**, an exemplary data structure for a dialing profile is shown generally at **253** and includes a user name field **258**, a domain field **260**, and calling attributes comprising a national dialing digits (NDD) field **262**, an international dialing digits (IDD) field **264**, a country code field **266**, a local area codes field **267**, a caller minimum local length field **268**, a caller maximum local length field **270**, a reseller field **273**, a maximum number of concurrent calls field **275** and a current number of concurrent calls field **277**. Effectively the dialing profile is a record identifying calling attributes of the caller identified by the caller identifier. More generally, dialing profiles represent calling attributes of respective subscribers.

An exemplary caller profile for the Vancouver subscriber is shown generally at **276** in Figure **10** and indicates that the user name field **258** includes the user name (**2001 1050 8667**) that has been assigned to the subscriber and is stored in the user name field **50** in the telephone as shown in Figure **2**.

Referring back to Figure **10**, the domain field **260** includes a domain name as shown at **282**, including a node type identifier **284**, a location code identifier **286**, a system provider identifier **288** and a domain portion **290**. The domain

field **260** effectively identifies a domain or node associated with the user identified by the contents of the user name field **258**.

In this embodiment, the node type identifier **284** includes the code "sp" identifying a supernode and the location identifier **286** identifies the supernode as being in Vancouver (YVR). The system provider identifier **288** identifies the company supplying the service and the domain portion **290** identifies the "com" domain.

The national dialled digit field **262** in this embodiment includes the digit "**1**" and, in general, includes a number specified by the International Telecommunications Union (ITU) Telecommunications Standardization Sector (ITU-T) E. **164** Recommendation which assigns national dialing digits to countries.

The international dialing digit field **264** includes a code also assigned according to the ITU-T according to the country or location of the user.

The country code field **266** also includes the digit "**1**" and, in general, includes a number assigned according to the ITU-T to represent the country in which the user is located.

The local area codes field **267** includes a list of area codes that have been assigned by the ITU-T to the geographical area in which the subscriber is located. The caller minimum and maximum local number length fields **268** and **270** hold numbers representing minimum and maximum local number lengths permitted in the area code(s) specified by the contents of the local area codes field **267**. The reseller field **273** is optional and holds a code identifying a retailer of the services, in this embodiment "Klondike". The maximum number of concurrent calls field **275** holds a code identifying the maximum number of concurrent calls that the user is entitled to cause to concurrently exist. This permits more than one call to occur concurrently while all calls for the user are

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billed to the same account. The current number of concurrent calls field **277** is initially **0** and is incremented each time a concurrent call associated with the user is initiated and is decremented when a concurrent call is terminated.

The area codes associated with the user are the area codes associated with the location code identifier **286** of the contents of the domain field **260**.

A dialing profile of the type shown in Figure **9** is produced whenever a user registers with the system or agrees to become a subscriber to the system. Thus, for example, a user wishing to subscribe to the system may contact an office maintained by a system operator and personnel in the office may ask the user certain questions about his location and service preferences, whereupon tables can be used to provide office personnel with appropriate information to be entered into the user name **258**, domain **260**, NDD **262**, IDD **264**, country code **266**, local area codes **267**, caller minimum and maximum local length fields **268** and **270** reseller field **273** and concurrent call fields **275** and **277** to establish a dialing profile for the user.

Referring to Figures **11** and **12**, callee dialing profiles for users in Calgary and London, respectively for example, are shown.

In addition to creating dialing profiles when a user registers with the system, a direct-in-dial (DID) record of the type shown at **278** in Figure **13** is added to a direct-in-dial bank table in the database (**18** in Figure **1**) to associate the username and a host name of the supernode with which the user is associated, with an E.**164** number associated with the user on the PSTN network.

An exemplary DID table record entry for the Calgary callee is shown generally at **300** in Figure **14**. The user name field **281** and user domain field **272** are analogous to the user name and user domain fields **258** and **260** of the caller dialing profile shown in Figure **10**. The contents of the DID field **274** include a

E.**164** public telephone number including a country code **283**, an area code **285**, an exchange code **287** and a number **289**. If the user has multiple telephone numbers, then multiple records of the type shown at **300** would be included in the DID bank table, each having the same user name and user domain, but different DID field **274** contents reflecting the different telephone numbers associated with that user.

In addition to creating dialing profiles as shown in Figure **9** and DID records as shown in Figure **13** when a user registers with the system, call blocking records of the type shown in Figure **26**, call forwarding records of the type shown in Figure **28** and voicemail records of the type shown in Figure **30** may be added to the database **18** when a new subscriber is added to the system.

Referring back to Figure **8**A, after retrieving a dialing profile for the caller, such as shown at **276** in Figure **10**, the RC processor circuit **200** is directed to block **256** which directs the processor circuit (**200**) to determine whether the contents of the concurrent call field **277** are less then the contents of the maximum concurrent call field **275** of the dialing profile for the caller and, if so, block **271** directs the processor circuit to increment the contents of the concurrent call field **277**. If the contents of concurrent call field **277** are equal to or greater than the contents of the maximum concurrent call field **275**, block **259** directs the processor circuit **200** to send an error message back to the call controller (**14**) to cause the call controller to notify the caller that the maximum number of concurrent calls has been reached and no further calls can exist concurrently, including the presently requested call.

Assuming block **256** allows the call to proceed, the RC processor circuit **200** is directed to perform certain checks on the callee identifier provided by the contents of the callee field **154** in Figure **6**, of the RC request message **150**. These checks are shown in greater detail in Figure **8**B.

Referring to Figure **8**B, the processor (**202** in Figure **7**) is directed to a first block **257** that causes it to determine whether a digit pattern of the callee identifier (**154**) provided in the RC request message (**150**) includes a pattern that matches the contents of the international dialing digits (IDD) field **264** in the caller profile shown in Figure **10**. If so, then block **259** directs the processor (**202**) to set a call type code identifier variable maintained by the processor to indicate that the call is an international call and block **261** directs the processor to produce a reformatted callee identifier by reformatting the callee identifier into a predefined digit format. In this embodiment, this is done by removing the pattern of digits matching the IDD field contents **264** of the caller dialing profile to effectively shorten the callee identifier. Then, block **263** directs the processor **202** to determine whether or not the callee identifier has a length which meets criteria establishing it as a number compliant with the E.**164** Standard set by the ITU. If the length does not meet this criteria, block **265** directs the processor **202** to send back to the call controller (**14**) a message indicating the length is not correct. The process is then ended. At the call controller **14**, routines (not shown) stored in the program memory **104** may direct the processor (**102** of Figure **4**) to respond to the incorrect length message by transmitting a message back to the telephone (**12** shown in Figure **1**) to indicate that an invalid number has been dialled.

Still referring to Figure **8**B, if the length of the amended callee identifier meets the criteria set forth at block **263**, block **269** directs the processor (**202** of Figure **7**) to make a database request to determine whether or not the amended callee identifier is found in a record in the direct-in-dial bank (DID) table. Referring back to Figure **8**B, at block **269**, if the processor **202** receives a response from the database indicating that the reformatted callee identifier produced at block **261** is found in a record in the DID bank table, then the callee is a subscriber to the system and the call is classified as a private network call by directing the processor to block **279** which directs the processor to copy the contents of the corresponding user name field (**281** in Figure **14**) from the callee DID bank table record (**300** in Figure **14**) into the

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callee ID buffer (**211** in Figure **7**). Thus, the processor **202** locates a subscriber user name associated with the reformatted callee identifier. The processor **202** is then directed to point B in Figure **8**A.

Subscriber to Subscriber Calls Between Different Nodes

Referring to Figure **8**A, block **280** directs the processor (**202** of Figure **7**) to execute a process to determine whether or not the node associated with the reformatted callee identifier is the same node that is associated with the caller identifier. To do this, the processor **202** determines whether or not a prefix (e.g., continent code **61**) of the callee name held in the callee ID buffer (**211** in Figure **7**), is the same as the corresponding prefix of the caller name held in the username field **258** of the caller dialing profile shown in Figure **10**. If the corresponding prefixes are not the same, block **302** in Figure **8**A directs the processor (**202** in Figure **7**) to set a call type flag in the buffer memory (**207** in Figure **7**) to indicate the call is a cross-domain call. Then, block **350** of Figure **8**A directs the processor (**202** of Figure **7**) to produce a routing message identifying an address on the private network with which the callee identified by the contents of the callee ID buffer is associated and to set a time to live for the call at a maximum value of **99999**, for example.

Thus the routing message includes a caller identifier, a call identifier set according to a username associated with the located DID bank table record and includes an identifier of a node on the private network with which the callee is associated.

The node in the system with which the callee is associated is determined by using the callee identifier to address a supernode table having records of the type as shown at **370** in Figure **17**. Each record **370** has a prefix field **372** and a supernode address field **374**. The prefix field **372** includes the first n digits of the callee identifier. In this embodiment n=**2**. The supernode address field **374** holds a code representing the IP address or a fully qualified domain name of the node associated with the code stored in the callee identifier prefix field

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372. Referring to Figure 18, for example, if the prefix is 20, the supernode address associated with that prefix is sp.yvr.digifonica.com.

Referring to Figure 15, a generic routing message is shown generally at 352 and includes an optional supplier prefix field 354, and optional delimiter field 356, a callee user name field 358, at least one route field 360, a time to live field 362 and other fields 364. The optional supplier prefix field 354 holds a code for identifying supplier traffic. The optional delimiter field 356 holds a symbol that delimits the supplier prefix code from the callee user name field 358. In this embodiment, the symbol is a number sign (#). The route field 360 holds a domain name or IP address of a gateway or node that is to carry the call, and the time to live field 362 holds a value representing the number of seconds the call is permitted to be active, based on subscriber available minutes and other billing parameters.

Referring to Figure 8A and Figure 16, an example of a routing message produced by the processor at block 350 for a caller associated with a different node than the caller is shown generally at 366 and includes only a callee field 359, a route field 361 and a time to live field 362.

Referring to Figure 8A, having produced a routing message as shown in Figure 16, block 381 directs the processor (202 of Figure 7) to send the routing message shown in Figure 16 to the call controller 14 shown in Figure 1.

Referring back to Figure 8B, if at block 257, the callee identifier stored in the callee id buffer (211 in Figure 7) does not begin with an international dialing digit, block 380 directs the processor (202) to determine whether or not the callee identifier begins with the same national dial digit code as assigned to the caller. To do this, the processor (202) is directed to refer to the retrieved caller dialing profile as shown in Figure 10. In Figure 10, the national dialing

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digit code **262** is the number **1**. Thus, if the callee identifier begins with the number **1**, then the processor (**202**) is directed to block **382** in Figure **8**B.

Block **382** directs the processor (**202** of Figure **7**) to examine the callee identifier to determine whether or not the digits following the NDD digit identify an area code that is the same as any of the area codes identified in the local area codes field **267** of the caller dialing profile **276** shown in Figure **10**. If not, block **384** of Figure **8**B directs the processor **202** to set the call type flag to indicate that the call is a national call. If the digits following the NDD digit identify an area code that is the same as a local area code associated with the caller as indicated by the caller dialing profile, block **386** directs the processor **202** to set the call type flag to indicate a local call, national style. After executing blocks **384** or **386**, block **388** directs the processor **202** to format the callee identifier into a pre-defined digit format to produce a re-formatted callee identifier by removing the national dialled digit and prepending a caller country code identified by the country code field **266** of the caller dialing profile shown in Figure **10**. The processor (**202**) is then directed to block **263** of Figure **8**B to perform other processing as already described above.

If at block **380**, the callee identifier does not begin with a national dialled digit, block **390** directs the processor (**202**) to determine whether the callee identifier begins with digits that identify the same area code as the caller. Again, the reference for this is the retrieved caller dialing profile shown in Figure **10**. The processor (**202**) determines whether or not the first few digits of the callee identifier identify an area code corresponding to the local area code field **267** of the retrieved caller dialing profile. If so, then block **392** directs the processor **202** to set the call type flag to indicate that the call is a local call and block **394** directs the processor (**202**) to format the callee identifier into a pre-defined digit format to produce a reformatted callee identifier by prepending the caller country code to the callee identifier, the caller country code being determined from the country code field **266** of the

retrieved caller dialing profile shown in Figure **10**. The processor (**202**) is then directed to block **263** for further processing as described above.

Referring back to Figure **8B**, at block **390**, the callee identifier does not start with the same area code as the caller, block **396** directs the processor (**202** of Figure **7**) to determine whether the number of digits in the callee identifier, i.e. the length of the callee identifier, is within the range of digits indicated by the caller minimum local number length field **268** and the caller maximum local number length field **270** of the retrieved caller dialing profile shown in Figure **10**. If so, then block **398** directs the processor (**202**) to set the call type flag to indicate a local call and block **400** directs the processor (**202**) to format the callee identifier into a pre-defined digit format to produce a reformatted callee identifier by prepending to the callee identifier the caller country code (as indicated by the country code field **266** of the retrieved caller dialing profile shown in Figure **10)** followed by the caller area code (as indicated by the local area code field **267** of the caller profile shown in Figure **10**). The processor (**202**) is then directed to block **263** of Figure **8B** for further processing as described above.

Referring back to Figure **8B**, if at block **396**, the callee identifier has a length that does not fall within the range specified by the caller minimum local number length field (**268** in Figure **10**) and the caller maximum local number length field (**270** in Figure **10**), block **402** directs the processor **202** of Figure **7** to determine whether or not the callee identifier identifies a valid user name. To do this, the processor **202** searches through the database (**18** of Figure **10** of dialing profiles to find a dialing profile having user name field contents (**258** in Figure **10**) that match the callee identifier. If no match is found, block **404** directs the processor (**202**) to send an error message back to the call controller (**14**). If at block **402**, a dialing profile having a user name field **258** that matches the callee identifier is found, block **406** directs the processor **202** to set the call type flag to indicate that the call is a private network call and then the processor is directed to block **280** of Figure **8A**. Thus, the call is

classified as a private network call when the callee identifier identifies a subscriber to the private network.

From Figure **8B**, it will be appreciated that there are certain groups of blocks of codes that direct the processor **202** in Figure **7** to determine whether the callee identifier has certain features such as an international dialing digit, a national dialing digit, an area code and a length that meet certain criteria, and cause the processor **202** to reformat the callee identifier stored in the callee id buffer **211**, as necessary into a predetermined target format including only a country code, area code, and a normal telephone number, for example, to cause the callee identifier to be compatible with the E.**164** number plan standard in this embodiment. This enables block **269** in Figure **8B** to have a consistent format of callee identifiers for use in searching through the DID bank table records of the type shown in Figure **13** to determine how to route calls for subscriber to subscriber calls on the same system. Effectively, therefore blocks **257**, **380**, **390**, **396** and **402** establish call classification criteria for classifying the call as a public network call or a private network call. Block **269** classifies the call, depending on whether or not the formatted callee identifier has a DID bank table record and this depends on how the call classification criteria are met and block **402** directs the processor **202** of Figure **7** to classify the call as a private network call when the callee identifier complies with a pre-defined format, i.e. is a valid user name and identifies a subscriber to the private network, after the callee identifier has been subjected to the classification criteria of blocks **257**, **380**, **390** and **396**.

Subscriber to Non-Subscriber Calls

Not all calls will be subscriber to subscriber calls and this will be detected by the processor **202** of Figure **7** when it executes block **269** in Figure **8B**, and does not find a DID bank table record that is associated with the callee, in the DID bank table. When this occurs, the call is classified as a public network

call by directing the processor **202** to block **408** of Figure **8**B which causes it to set the contents of the callee id buffer **211** of Figure **7** equal to the newly formatted callee identifier, i.e., a number compatible with the E.**164** standard. Then, block **410** of Figure **8**B directs the processor (**202**) to search a database of route or master list records associating route identifiers with dialing codes shown in Figure **19** to locate a router having a dialing code having a number pattern matching at least a portion of the reformatted callee identifier.

Referring to Figure **19**, a data structure for a master list or route list record is shown. Each master list record includes a master list ID field **500**, a dialing code field **502**, a country code field **504**, a national sign number field **506**, a minimum length field **508**, a maximum length field **510**, a national dialled digit field **512**, an international dialled digit field **514** and a buffer rate field **516**.

The master list ID field **500** holds a unique code such as **1019**, for example, identifying the record. The dialing code field **502** holds a predetermined number pattern that the processor **202** of Figure **7** uses at block **410** in Figure **8**B to find the master list record having a dialing code matching the first few digits of the amended callee identifier stored in the callee id buffer **211**. The country code field **504** holds a number representing the country code associated with the record and the national sign number field **506** holds a number representing the area code associated with the record. (It will be observed that the dialing code is a combination of the contents of the country code field **504** and the national sign number field **506**.) The minimum length field **508** holds a number representing the minimum length of digits associated with the record and the maximum length field **51** holds a number representing the maximum number of digits in a number with which the record may be compared. The national dialled digit (NDD) field **512** holds a number representing an access code used to make a call within the country specified by the country code, and the international dialled digit (IDD) field **514** holds a

number representing the international prefix needed to dial a call from the country indicated by the country code.

Thus, for example, a master list record may have a format as shown in Figure **20** with exemplary field contents as shown.

Referring back to Figure **8**B, using the country code and area code portions of the reformatted callee identifier stored in the callee id buffer **211**, block **410** directs the processor **202** of Figure **7** to find a master list record such as the one shown in Figure **20** having a dialing code that matches the country code (**1**) and area code (**604**) of the callee identifier. Thus, in this example, the processor (**202**) would find a master list record having an ID field containing the number **1019**. This number may be referred to as a route ID. Thus, a route ID number is found in the master list record associated with a predetermined number pattern in the reformatted callee identifier.

After executing block **410** in Figure **8**B, the process continues as shown in Figure **8**D. Referring to Figure **8**D, block **412** directs the processor **202** of Figure **7** to use the route ID number to search a database of supplier records associating supplier identifiers with route identifiers to locate at least one supplier record associated with the route identifier to identify at least one supplier operable to supply a communications link for the route.

Referring to Figure **21**, a data structure for a supplier list record is shown. Supplier list records include a supplier ID field **540**, a master list ID field **542**, an optional prefix field **544**, a specific route identifier field **546**, a NDD/IDD rewrite field **548**, a rate field **550**, and a timeout field **551**. The supplier ID field **540** holds a code identifying the name of the supplier and the master list ID field **542** holds a code for associating the supplier record with a master list record. The prefix field **544** holds a string used to identify the supplier traffic and the specific route identifier field **546** holds an IP address of a gateway operated by the supplier indicated by the supplier ID field **540**. The NDD/IDD

rewrite field **548** holds a code representing a rewritten value of the NDD/IDD associated with this route for this supplier, and the rate field **550** holds a code indicating the cost per second to the system operator to use the route provided by the gateway specified by the contents of the route identifier field **546**. The timeout field **551** holds a code indicating a time that the call controller should wait for a response from the associated gateway before giving up and trying the next gateway. This time value may be in seconds, for example. Exemplary supplier records are shown in Figures **22**, **23** and **24** for the exemplary suppliers shown at **20** in Figure **1**, namely Telus, Shaw and Sprint.

Referring back to Figure **8**D, at block **412** the processor **202** finds all supplier records that identify the master list ID found at block **410** of Figure **8**B.

Referring back to Figure **8**D, block **560** directs the processor **202** of Figure **7** to begin to produce a routing message of the type shown in Figure **15**. To do this, the processor **202** loads a routing message buffer as shown in Figure **25** with a supplier prefix of the least costly supplier where the least costly supplier is determined from the rate fields **550** of Figure **21** of the records associated with respective suppliers.

Referring to Figures **22-24**, in the embodiment shown, the supplier "Telus" has the lowest number in the rate field **550** and therefore the prefix **4973** associated with that supplier is loaded into the routing message buffer shown in Figure **25** first.

Block **562** in Figure **8**D directs the processor to delimit the prefix **4973** by the number sign (#) and to next load the reformatted callee identifier into the routing message buffer shown in Figure **25**. At block **563** of Figure **8**D, the contents of the route identifier field **546** of Figure **21** of the record associated with the supplier "Telus" are added by the processor **202** of Figure **7** to the routing message buffer shown in Figure **25** after an @ sign delimiter, and then

block **564** in Figure **8D** directs the processor to get a time to live value, which in one embodiment may be **3600** seconds, for example. Block **566** then directs the processor **202** to load this time to live value and the timeout value (**551**) in Figure **21** in the routing message buffer of Figure **25**. Accordingly, a first part of the routing message for the Telus gateway is shown generally at **570** in Figure **25**.

Referring back to Figure **8D**, block **571** directs the processor **202** back to block **560** and causes it to repeat blocks **560**, **562**, **563**, **564** and **566** for each successive supplier until the routing message buffer is loaded with information pertaining to each supplier identified by the processor at block **412**. Thus, a second portion of the routing message as shown at **572** in Figure **25** relates to the second supplier identified by the record shown in Figure **23**. Referring back to Figure **25**, a third portion of the routing message as shown at **574** and is associated with a third supplier as indicated by the supplier record shown in Figure **24**.

Consequently, referring to Figure **25**, the routing message buffer holds a routing message identifying a plurality of different suppliers able to provide gateways to the public telephone network (i.e. specific routes) to establish at least part of a communication link through which the caller may contact the callee. In this embodiment, each of the suppliers is identified, in succession, according to rate. Other criteria for determining the order in which suppliers are listed in the routing message may include preferred supplier priorities which may be established based on service agreements, for example.

Referring back to Figure **8D**, block **568** directs the processor **202** of Figure **7** to send the routing message shown in Figure **25** to the call controller **14** in Figure **1**.

Subscriber to Subscriber Calls Within the Same Node

Referring back to Figure **8A**, if at block **280**, the callee identifier received in the RC request message has a prefix that identifies the same node as that associated with the caller, block **600** directs the processor **202** to use the callee identifier in the callee id buffer **211** to locate and retrieve a dialing profile for the callee. The dialing profile may be of the type shown in Figure **11** or **12**, for example. Block **602** of Figure **8A** then directs the processor **202** of Figure **7** to get call block, call forward and voicemail records from the database **18** of Figure **1** based on the user name identified in the callee dialing profile retrieved by the processor at block **600**. Call block, call forward and voicemail records may be as shown in Figures **26**, **27**, **28** and **30** for example.

Referring to Figure **26**, the call block records include a user name field **604** and a block pattern field **606**. The user name field holds a user name corresponding to the user name in the user name field (**258** in Figure **10**) of the callee profile and the block pattern field **606** holds one or more E.**164**-compatible numbers or user names identifying PSTN numbers or system subscribers from whom the subscriber identified in the user name field **604** does not wish to receive calls.

Referring to Figure **8A** and Figure **27**, block **608** directs the processor **202** of Figure **7** to determine whether or not the caller identifier received in the RC request message matches a block pattern stored in the block pattern field **606** of the call block record associated with the callee identified by the contents of the user name field **604** in Figure **26**. If the caller identifier matches a block pattern, block **610** directs the processor to send a drop call or non-completion message to the call controller (**14**) and the process is ended. If the caller identifier does not match a block pattern associated with the callee, block **609** directs the processor to store the username and domain of the callee, as determined from the callee dialing profile, and a time to live value in the routing message buffer as shown at **650** in Figure **32**. Referring back to

Figure **8**A, block **612** then directs the processor **202** to determine whether or not call forwarding is required.

Referring to Figure **28**, the call forwarding records include a user name field **614**, a destination number field **616**, and a sequence number field **618**. The user name field **614** stores a code representing a user with which the record is associated. The destination number field **616** holds a user name representing a number to which the current call should be forwarded, and the sequence number field **618** holds an integer number indicating the order in which the user name associated with the corresponding destination number field **616** should be attempted for call forwarding. The call forwarding table may have a plurality of records for a given user. The processor **202** of Figure **7** uses the contents of the sequence number field **618** to place the records for a given user in order. As will be appreciated below, this enables the call forwarding numbers to be tried in an ordered sequence.

Referring to Figure **8**A and Figure **29**, if at block **612**, the call forwarding record for the callee identified by the callee identifier contains no contents in the destination number field **616** and accordingly no contents in the sequence number field **618**, there are no call forwarding entries for this callee, and the processor **202** is directed to block **620** in Figure **8**C. If there are entries in the call forwarding table **27**, block **622** in Figure **8**A directs the processor **202** to search the dialing profile table to find a dialing profile record as shown in Figure **9**, for the user identified by the destination number field **616** of the call forward record shown in Figure **28**. The processor **202** of Figure **7** is further directed to store the username and domain for that user and a time to live value in the routing message buffer as shown at **652** in Figure **32**, to produce a routing message as illustrated. This process is repeated for each call forwarding record associated with the callee identified by the callee id buffer **211** in Figure **7** to add to the routing message buffer all call forwarding usernames and domains associated with the callee.

Referring back to Figure **8**A, if at block **612** there are no call forwarding records, then at block **620** in Figure **8**C the processor **202** is directed to determine whether or not the user identified by the callee identifier has paid for voicemail service. This is done by checking to see whether or not a flag is set in a voicemail record of the type shown in Figure **30** in a voicemail table stored in the database **18** shown in Figure **1**.

Referring to Figure **30**, voicemail records in this embodiment may include a user name field **624**, a voicemail server field **626**, a seconds to voicemail field **628** and an enable field **630**. The user name field **624** stores the user name of the callee. The voicemail server field **626** holds a code identifying a domain name of a voicemail server associated with the user identified by the user name field **624**. The seconds to voicemail field **628** holds a code identifying the time to wait before engaging voicemail, and the enable field **630** holds a code representing whether or not voicemail is enabled for the user. Referring back to Figure **8**C, at block **620** if the processor **202** of Figure **7** finds a voicemail record as shown in Figure **30** having user name field **624** contents matching the callee identifier, the processor is directed to examine the contents of the enabled field **630** to determine whether or not voicemail is enabled. If voicemail is enabled, then block **640** in Figure **8**C directs the processor **202** to Figure **7** to store the contents of the voicemail server field **626** and the contents of the seconds to voicemail field **628** in the routing message buffer, as shown at **654** in Figure **32**. Block **642** then directs the processor **202** to get time to live values for each path specified by the routing message according to the cost of routing and the user's balance. These time to live values are then appended to corresponding paths already stored in the routing message buffer.

Referring back to Figure **8**C, block **644** then directs the processor **202** of Figure **7** to store the IP address of the current node in the routing message buffer as shown at **656** in Figure **32**. Block **646** then directs the processor **202** to send the routing message shown in Figure **32** to the call controller **14** in

Figure **1**. Thus in the embodiment described the routing controller will produce a routing message that will cause at least one of the following: forward the call to another party, block the call and direct the caller to a voicemail server.

Referring back to Figure **1**, the routing message whether of the type shown in Figures **16**, **25** or **32**, is received at the call controller **14** and the call controller interprets the receipt of the routing message as a request to establish a call.

Referring to Figure **4**, the program memory **104** of the call controller **14** includes a routing to gateway routine depicted generally at **122**.

Where a routing message of the type shown in Figure **32** is received by the call controller **14**, the routing to gateway routine **122** shown in Figure **4** may direct the processor **102** cause a message to be sent back through the internet **13** shown in Figure **1** to the callee telephone **15**, knowing the IP address of the callee telephone **15** from the user name.

Alternatively, if the routing message is of the type shown in Figure **16**, which identifies a domain associated with another node in the system, the call controller may send a SIP invite message along the high speed backbone **17** connected to the other node. The other node functions as explained above, in response to receipt of a SIP invite message.

If the routing message is of the type shown in Figure **25** where there are a plurality of gateway suppliers available, the call controller sends a SIP invite message to the first supplier, in this case Telus, using a dedicated line or an internet connection to determine whether or not Telus is able to handle the call. If the Telus gateway returns a message indicating it is not able to handle the call, the call controller **14** then proceeds to send a SIP invite message to the next supplier, in this case Shaw. The process is repeated until one of the suppliers responds indicating that it is available to carry the call. Once a supplier responds indicating that it is able to carry the call, the supplier sends

back to the call controller **14** an IP address for a gateway provided by the supplier through which the call or audio path of the call will be carried. This IP address is sent in a message from the call controller **14** to the media relay **9** which responds with a message indicating an IP address to which the caller telephone should send its audio/video, traffic and an IP address to which the gateway should send its audio/video for the call. The call controller conveys the IP address at which the media relay expects to receive audio/video from the caller telephone, to the caller telephone **12** in a message. The caller telephone replies to the call controller with an IP address at which it would like to receive audio/video and the call controller conveys that IP address to the media relay. The call may then be conducted between the caller and callee through the media relay and gateway.

Referring back to Figure **1**, if the call controller **14** receives a routing message of the type shown in Figure **32**, and which has at least one call forwarding number and/or a voicemail number, the call controller attempts to establish a call to the callee telephone **15** by seeking from the callee telephone a message indicating an IP address to which the media relay should send audio/video. If no such message is received from the callee telephone, no call is established. If no call is established within a pre-determined time, the call controller **14** attempts to establish a call with the next user identified in the call routing message in the same manner. This process is repeated until all call forwarding possibilities have been exhausted, in which case the call controller communicates with the voicemail server **19** identified in the routing message to obtain an IP address to which the media relay should send audio/video and the remainder of the process mentioned above for establishing IP addresses at the media relay **9** and the caller telephone is carried out to establish audio/video paths to allowing the caller to leave a voicemail message with the voicemail server.

When an audio/video path through the media relay is established, a call timer maintained by the call controller **14** logs the start date and time of the call and

logs the call ID and an identification of the route (i.e., audio/video path IP address) for later use in billing.

Time to Live

Referring to Figures **33**A and **33**B, a process for determining a time to live value for any of blocks **642** in Figure **8**C, **350** in Figure **8**A or **564** in Figure **8**D above is described. The process is executed by the processor **202** shown in Figure **7**. Generally, the process involves calculating a cost per unit time, calculating a first time value as a sum of a free time attributed to a participant in the communication session and the quotient of a funds balance held by the participant to the cost per unit time value and producing a second time value in response to the first time value and a billing pattern associated with the participant, the billing pattern including first and second billing intervals and the second time value being the time to permit a communication session to be conducted.

Referring to Figure **33**A, in this embodiment, the process begins with a first block **700** that directs the RC processor to determine whether or not the call type set at block **302** in Figure **8**A indicates the call is a network or cross-domain call. If the call is a network or cross-domain call, block **702** of Figure **33**A directs the RC processor to set the time to live equal to **99999** and the process is ended. Thus, the network or cross-domain call type has a long time to live. If at block **700** the call type is determined not to be a network or cross-domain type, block **704** directs the RC processor to get a subscriber bundle table record from the database **18** in Figure **1** and store it locally in the subscriber bundle record buffer at the RC **14**.

Referring to Figure **34**, a subscriber bundle table record is shown generally at **706**. The record includes a user name field **708** and a services field **710**. The user name field **708** holds a code identifying the subscriber user name and the services field **710** holds codes identifying service features assigned to the subscriber, such as free local calling, call blocking and voicemail, for example.

Figure **35** shows an exemplary subscriber bundle record for the Vancouver caller. In this record the user name field **708** is loaded with the user name **2001 1050 8667** and the services field **710** is loaded with codes **10**, **14** and **16** corresponding to free local calling, call blocking and voicemail, respectively. Thus, user **2001 1050 8667** has free local calling, call blocking and voicemail features.

Referring back to Figure **33A**, after having loaded a subscriber bundle record into the subscriber bundle record buffer, block **712** directs the RC processor to search the database (**18**) determine whether or not there is a bundle override table record for the master list ID value that was determined at block **410** in Figure **8B**. An exemplary bundle override table record is shown at **714** in Figure **36**. The bundle table record includes a master list ID field **716**, an override type field **718**, an override value field **720** a first interval field **722** and a second interval field **724**. The master list ID field **716** holds a master list ID code. The override type field **718** holds an override type code indicating a fixed, percent or cent amount to indicate the amount by which a fee will be increased. The override value field **720** holds a real number representing the value of the override type. The first interval field **722** holds a value indicating the minimum number of seconds for a first level of charging and the second interval field **724** holds a number representing a second level of charging.

Referring to Figure **37**, a bundle override record for the located master list ID code is shown generally at **726** and includes a master list ID field **716** holding the code **1019** which was the code located in block **410** of Figure **8B**. The override type field **718** includes a code indicating the override type is a percentage value and the override value field **720** holds the value **10.0** indicating that the override will be **10.0**% of the charged value. The first interval field **722** holds a value representing **30** seconds and the second interval field **724** holds a value representing **6** seconds. The **30** second value in the first interval field **722** indicates that charges for the route will be made at

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a first rate for **30** seconds and thereafter the charges will be made at a different rate in increments of **6** seconds, as indicated by the contents of the second interval field **724**.

Referring back to Figure **33A**, if at block **712** the processor finds a bundle override record of the type shown in Figure **37**, block **728** directs the processor to store the bundle override record in local memory. In the embodiment shown, the bundle override record shown in Figure **37** is stored in the bundle override record buffer at the RC as shown in Figure **7**. Still referring to Figure **33**A, block **730** then directs the RC processor to determine whether or not the subscriber bundle table record **706** in Figure **35** has a services field including a code identifying that the user is entitled to free local calling and also directs the processor to determine whether or not the call type is not a cross domain cell, i.e. it is a local or local/national style. If both of these conditions are satisfied, block **732** directs the processor to set the time to live equal to **99999**, giving the user a long period of time for the call. The process is then ended. If the conditions associated with block **730** are not satisfied, block **734** of Figure **33**B directs the RC processor to retrieve a subscriber account record associated with a participant in the call. This is done by copying and storing in the subscriber account record buffer a subscriber account record for the caller.

Referring to Figure **38**, an exemplary subscriber account table record is shown generally at **736**. The record includes a user name field **738**, a funds balance field **740** and a free time field **742**. The user name field **738** holds a subscriber user name, the funds balance field **740** holds a real number representing the dollar value of credit available to the subscriber and the free time field **742** holds an integer representing the number of free seconds that the user is entitled to.

An exemplary subscriber account record for the Vancouver caller is shown generally at **744** in Figure **39**, wherein the user name field **738** holds the user

name **2001 1050 8667**, the funds balance field **740** holds the value **$10.00**, and the free time field **742** holds the value **100**. The funds balance field holding the value of **$10.00** indicates the user has **$10.00** worth of credit and the free time field having the value of **100** indicates that the user has a balance of **100** free seconds of call time.

Referring back to Figure **33**B, after copying and storing the subscriber account record shown in Figure **39** from the database to the subscriber account record buffer RC, block **746** directs the processor to determine whether or not the subscriber account record funds balance field **740** or free time field **742** are greater than zero. If they are not greater than zero, block **748** directs the processor to set the time to live equal to zero and the process is ended. The RC then sends a message back to the call controller to cause the call controller to deny the call to the caller. If the conditions associated with block **746** are satisfied, block **750** directs the processor to calculate the call cost per unit time. A procedure for calculating the call cost per unit time is described below in connection with Figure **41**.

Assuming the procedure for calculating the cost per second returns a number representing the call cost per second, block **752** directs the processor **202** in Figure **7** to determine whether or not the cost per second is equal to zero. If so, block **754** directs the processor to set the time to live to **99999** to give the caller a very long length of call and the process is ended.

If at block **752** the call cost per second is not equal to zero, block **756** directs the processor **202** in Figure **7** to calculate a first time to live value as a sum of a free time attributed to the participant in the communication session and the quotient of the funds balance held by the participant to the cost per unit time value. To do this, the processor **202** of Figure **7** is directed to set a first time value or temporary time to live value equal to the sum of the free time provided in the free time field **742** of the subscriber account record shown in Figure **39** and the quotient of the contents of the funds balance field **740** in the

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subscriber account record for the call shown in Figure **39** and the cost per second determined at block **750** of Figure **33B**. Thus, for example, if at block **750** the cost per second is determined to be three cents per second and the funds balance field holds the value $**10.00**, the quotient of the funds balance and cost per second is **333** seconds and this is added to the contents of the free time field **742**, which is **100**, resulting in a time to live of **433** seconds.

Block **758** then directs the RC processor to produce a second time value in response to the first time value and the billing pattern associated with the participant as established by the bundle override record shown in Figure **37**. This process is shown in greater detail at **760** in Figure **40** and generally involves producing a remainder value representing a portion of the second billing interval remaining after dividing the second billing interval into a difference between the first time value and the first billing interval.

Referring to Figure **40**, the process for producing the second time value begins with a first block **762** that directs the processor **202** in Figure **7** to set a remainder value equal to the difference between the time to live value calculated at block **756** in Figure **33B** and the contents of the first interval field **722** of the record shown in Figure **37**, multiplied by the modulus of the contents of the second interval field **724** of Figure **37**. Thus, in the example given, the difference between the time to live field and the first interval field is **433** minus **30**, which is **403** and therefore the remainder produced by the mod of **403** divided by **6** is **0.17**. Block **764** then directs the processor to determine whether or not this remainder value is greater than zero and, if so, block **766** directs the processor to subtract the remainder from the first time value and set the difference as the second time value. To do this the processor is directed to set the time to live value equal to the current time to live of **403** minus the remainder of **1**, i.e., **402** seconds. The processor is then returned back to block **758** of Figure **33B**.

Referring back to Figure **40**, if at block **764** the remainder is not greater than zero, block **768** directs the processor **202** of Figure **7** to determine whether or not the time to live is less than the contents of the first interval field **722** in the record shown in Figure **37**. If so, then block **770** of Figure **40** directs the processor to set the time to live equal to zero. Thus, the second time value is set to zero when the remainder is greater than zero and the first time value is less than the free time associated with the participant in the call. If at block **768** the conditions of that block are not satisfied, the processor returns the first time to live value as the second time to live value.

Thus, referring to Figure **33**B, after having produced a second time to live value, block **772** directs the processor to set the time to live value for use in blocks **342**, **350** or **564**.

Cost per Second

Referring back to Figure **33**B, at block **750** it was explained that a call cost per unit time is calculated. The following explains how that call cost per unit time value is calculated.

Referring to Figure **41**, a process for calculating a cost per unit time is shown generally at **780**. The process is executed by the processor **202** in Figure **7** and generally involves locating a record in a database, the record comprising a markup type indicator, a markup value and a billing pattern and setting a reseller rate equal to the sum of the markup value and the buffer rate, locating at least one of an override record specifying a route cost per unit time amount associated with a route associated with the communication session, a reseller record associated with a reseller of the communications session, the reseller record specifying a reseller cost per unit time associated with the reseller for the communication session and a default operator markup record specifying a default cost per unit time and setting as the cost per unit time the sum of the reseller rate and at least one of the route cost per unit time, the reseller cost per unit time and the default cost per unit time.

The process begins with a first set of blocks **782**, **802** and **820** which direct the processor **202** in Figure **7** to locate at least one of a record associated with a reseller and a route associated with the reseller, a record associated with the reseller, and a default reseller mark-up record. Block **782**, in particular, directs the processor to address the database **18** to look for a record associated with a reseller and a route with the reseller by looking for a special rate record based on the master list ID established at block **410** in Figure **8**C.

Referring to Figure **42**, a system operator special rate table record is shown generally at **784**. The record includes a reseller field **786**, a master list ID field **788**, a mark-up type field **790**, a mark-up value field **792**, a first interval field **794** and a second interval field **796**. The reseller field **786** holds a reseller ID code and the master list ID field **788** holds a master list ID code. The mark-up type field **790** holds a mark-up type such as fixed percent or cents and the mark-up value field **792** holds a real number representing the value corresponding to the mark-up type. The first interval field **794** holds a number representing a first level of charging and the second interval field **796** holds a number representing a second level of charging.

An exemplary system operator special rate table for a reseller known as "Klondike" is shown at **798** in Figure **43**. In this record, the reseller field **786** holds a code indicating the retailer ID is Klondike, the master list ID field **788** holds the code **1019** to associate the record with the master list ID code **1019**. The mark-up type field **790** holds a code indicating the mark-up type is cents and the mark-up value field **792** holds a mark-up value indicating **1/10** of one cent. The first interval field **794** holds the value **30** and the second interval field **796** holds the value **6**, these two fields indicating that the operator allows **30** seconds for free and then billing is done in increments of **6** seconds after that.

Referring back to Figure **41**, if at block **782** a record such as the one shown in Figure **43** is located in the system operator special rates table, the processor is directed to block **800** in Figure **41**. If such a record is not found in the system operator special rates table, block **802** directs the processor to address the database **18** to look in a system operator mark-up table for a mark-up record associated with the reseller.

Referring to Figure **44**, an exemplary system operator mark-up table record is shown generally at **804**. The record includes a reseller field **806**, a mark-up type field **808**, a mark-up value field **810**, a first interval field **812** and a second interval field **814**. The reseller mark-up type, mark-up value, first interval and second interval fields are as described in connection with the fields by the same names in the system operator special rates table shown in Figure **42**.

Figure **45** provides an exemplary system operator mark-up table record for the reseller known as Klondike and therefore the reseller field **806** holds the value "Klondike", the mark-up type field **808** holds the value cents, the mark-up value field holds the value **0.01**, the first interval field **812** holds the value **30** and the second interval field **814** holds the value **6**. This indicates that the reseller "Klondike" charges by the cent at a rate of one cent per minute. The first **30** seconds of the call are free and billing is charged at the rate of one cent per minute in increments of **6** seconds.

Figure **46** provides an exemplary system operator mark-up table record for cases where no specific system operator mark-up table record exists for a particular reseller, i.e., a default reseller mark-up record. This record is similar to the record shown in Figure **45** and the reseller field **806** holds the value "all", the mark-up type field **808** is loaded with a code indicating mark-up is based on a percentage, the mark-up value field **810** holds the percentage by which the cost is marked up, and the first and second interval fields **812** and **814** identify first and second billing levels.

Referring back to Figure **41**, if at block **802** a specific mark-up record for the reseller identified at block **782** is not located, block **820** directs the processor to get the mark-up record shown in Figure **46**, having the "all" code in the reseller field **806**. The processor is then directed to block **800**.

Referring back to Figure **41**, at block **800**, the processor **202** of Figure **7** is directed to set a reseller rate equal to the sum of the mark-up value of the record located by blocks **782**, **802** or **820** and the buffer rate specified by the contents of the buffer rate field **516** of the master list record shown in Figure **20**. To do this, the RC processor sets a variable entitled "reseller cost per second" to a value equal to the sum of the contents of the mark-up value field (**792**, **810**) of the associated record, plus the contents of the buffer rate field (**516**) from the master list record associated with the master list ID. Then, block **822** directs the processor to set a system operator cost per second variable equal to the contents of the buffer rate field (**516**) from the master list record. Block **824** then directs the processor to determine whether the call type flag indicates the call is local or national/local style and whether the caller has free local calling. If both these conditions are met, then block **826** sets the user cost per second variable equal to zero and sets two increment variables equal to one, for use in later processing. The cost per second has thus be calculated and the process shown in Figure **41** is ended.

If at block **824** the conditions of that block are not met, the processor **202** of Figure **7** is directed to locate at least one of a bundle override table record specifying a route cost per unit time associated with a route associated with the communication session, a reseller special destinations table record associated with a reseller of the communications session, the reseller record specifying a reseller cost per unit time associated with the reseller for the communication session and a default reseller global markup record specifying a default cost per unit time.

To do this block **828** directs the processor **202** of Figure **7** to determine whether or not the bundle override record **726** in Figure **37** located at block **712** in Figure **33A** has a master list ID equal to the stored master list ID that was determined at block **410** in Figure **8B**. If not, block **830** directs the processor to find a reseller special destinations table record in a reseller special destinations table in the database (**18**), having a master list ID code equal to the master list ID code of the master list ID that was determined at block **410** in Figure **8B**. An exemplary reseller special destinations table record is shown in Figure **47** at **832**. The reseller special destinations table record includes a reseller field **834**, a master list ID field **836**, a mark-up type field **838**, a mark-up value field **840**, a first interval field **842** and a second interval field **844**. This record has the same format as the system operator special rates table record shown in Figure **42**, but is stored in a different table to allow for different mark-up types and values and time intervals to be set according to resellers' preferences. Thus, for example, an exemplary reseller special destinations table record for the reseller "Klondike" is shown at **846** in Figure **48**. The reseller field **834** holds a value indicating the reseller as the reseller "Klondike" and the master list ID field holds the code **1019**. The mark-up type field **838** holds a code indicating the mark-up type is percent and the mark-up value field **840** holds a number representing the mark-up value as **5%**. The first and second interval fields identify different billing levels used as described earlier.

Referring back to Figure **41**, the record shown in Figure **48** may be located at block **830**, for example. If at block **830** such a record is not found, then block **832** directs the processor to get a default operator global mark-up record based on the reseller ID.

Referring to Figure **49**, an exemplary default reseller global mark-up table record is shown generally at **848**. This record includes a reseller field **850**, a mark-up type field **852**, a mark-up value field **854**, a first interval field **856** and a second interval field **858**. The reseller field **850** holds a code identifying the

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reseller. The mark-up type field **852**, the mark-up value field **854** and the first and second interval fields **856** and **858** are of the same type as described in connection with fields of the same name in Figure **47**, for example. The contents of the fields of this record **860** may be set according to system operator preferences, for example.

Referring to Figure **50**, an exemplary reseller global mark-up table record is shown generally at **860**. In this record, the reseller field **850** holds a code indicating the reseller is "Klondike", the mark-up type field **852** holds a code indicating the mark-up type is percent, the mark-up value field **854** holds a value representing **10**% as the mark-up value, the first interval field **856** holds the value **30** and the second interval field **858** holds the values **30** and **6** respectively to indicate the first **30** seconds are free and billing is to be done in **6** second increments after that.

Referring back to Figure **41**, should the processor get to block **832**, the reseller global mark-up table record as shown in Figure **50** is retrieved from the database and stored locally at the RC. As seen in Figure **41**, it will be appreciated that if the conditions are met in blocks **828** or **830**, or if the processor executes block **832**, the processor is then directed to block **862** which causes it to set an override value equal to the contents of the mark-up value field of the located record, to set the first increment variable equal to the contents of the first interval field of the located record and to set the second increment variable equal to the contents of the second interval field of the located record. (The increment variables were alternatively set to specific values at block **826** in Figure **41**.)

It will be appreciated that the located record could be a bundle override record of the type shown in Figure **37** or the located record could be a reseller special destination record of the type shown in Figure **48** or the record could be a reseller global mark-up table record of the type shown in Figure **50**. After the override and first and second increment variables have been set at block

862, the processor 202 if Figure 7 is directed to set as the cost per unit time the sum of the reseller rate and at least one of the route cost per unit time, the reseller cost per unit time and the default cost per unit time, depending on which record was located. To do this, block 864 directs the processor to set the cost per unit time equal to the sum of the reseller cost set at block 800 in Figure 41, plus the contents of the override variable calculated in block 862 in Figure 41. The cost per unit time has thus been calculated and it is this cost per unit time that is used in block 752 of Figure 33B, for example.

Terminating the Call

In the event that either the caller or the callee terminates a call, the telephone of the terminating party sends a SIP bye message to the controller 14. An exemplary SIP bye message is shown at 900 in Figure 51 and includes a caller field 902, a callee field 904 and a call ID field 906. The caller field 902 holds a twelve digit user name, the callee field 904 holds a PSTN compatible number or user name, and the call ID field 906 holds a unique call identifier field of the type shown in the call ID field 65 of the SIP invite message shown in Figure 3.

Thus, for example, referring to Figure 52, a SIP bye message for the Calgary callee is shown generally at 908 and the caller field 902 holds a user name identifying the caller, in this case 2001 1050 8667, the callee field 904 holds a user name identifying the Calgary callee, in this case 2001 1050 2222, and the call ID field 906 holds the code FA10 @ 192.168.0.20, which is the call ID for the call.

The SIP bye message shown in Figure 52 is received at the call controller 14 and the call controller executes a process as shown generally at 910 in Figure 53. The process includes a first block 912 that directs the call controller processor 202 of Figure 7 to copy the caller, callee and call ID field contents from the SIP bye message received from the terminating party to corresponding fields of an RC stop message buffer (not shown). Block 914

then directs the processor to copy the call start time from the call timer and to obtain a call stop time from the call timer. Block **916** then directs the call controller to calculate a communication session time by determining the difference in time between the call start time and the call stop time. This session time is then stored in a corresponding field of the RC call stop message buffer. Block **917** then directs the processor to decrement the contents of the current concurrent call field **277** of the dialing profile for the caller as shown in Figure **10**, to indicate that there is one less concurrent call in progress. A copy of the amended dialing profile for the caller is then stored in the database **18** of Figure **1**. Block **918** then directs the processor to copy the route from the call log. An RC call stop message produced as described above is shown generally at **1000** in Figure **54**. An RC call stop message specifically associated with the call made to the Calgary callee is shown generally at **1020** in Figure **55**.

Referring to Figure **54**, the RC stop call message includes a caller field **1002**, callee field **1004**, a call ID field **1006**, an account start time field **1008**, an account stop time field **1010**, a communication session time **1012** and a route field **1014**. The caller field **1002** holds a username, the callee field **1004** holds a PSTN-compatible number or system number, the call ID field **1006** hold the unique call identifier received from the SIP invite message shown in Figure **3**, the account start time field **1008** holds the date and start time of the call, the account stop time field **1010** holds the date and time the call ended, the communication session time field **1012** holds a value representing the difference between the start time and the stop time, in seconds, and the route field **1014** holds the IP address for the communications link that was established.

Referring to Figure **55**, an exemplary RC stop call message for the Calgary callee is shown generally at **1020**. In this example the caller field **1002** holds the user name **2001 1050 8667** identifying the Vancouver-based caller and the callee field **1004** holds the user name **2001 1050 2222** identifying the

Calgary callee. The contents of the call ID field **1006** are **FA10 @ 192.168.0.20**. The contents of the account start time field **1008** are **2006-12-30 12:12:12** and the contents of the account stop time field are **2006-12-30 12:12:14**. The contents of the communication session time field **1012** are **2** to indicate **2** seconds call duration and the contents of the route field are **72.64.39.58**.

Referring back to Figure **53**, after having produced an RC call stop message, block **920** directs the processor **202** in Figure **7** to send the RC stop message compiled in the RC call stop message buffer to the RC **16** of Figure **1**. Block **922** directs the call controller **14** to send a "bye" message back to the party that did not terminate the call.

The RC **16** of Figure **1** receives the call stop message and an RC call stop message process is invoked at the RC, the process being shown at **950** in Figures **56A**, **56B** and **56C**. Referring to Figure **56A**, the RC stop message process **950** begins with a first block **952** that directs the processor **202** in Figure **7** to determine whether or not the communication session time is less than or equal to the first increment value set by the cost calculation routine shown in Figure **41**, specifically blocks **826** or **862** thereof. If this condition is met, then block **954** of Figure **56A** directs the RC processor to set a chargeable time variable equal to the first increment value set at block **826** or **862** of Figure **41**. If at block **952** of Figure **56A** the condition is not met, block **956** directs the RC processor to set a remainder variable equal to the difference between the communication session time and the first increment value mod the second increment value produced at block **826** or **862** of Figure **41**. Then, the processor is directed to block **958** of Figure **56A** which directs it to determine whether or not the remainder is greater than zero. If so, block **960** directs the RC processor to set the chargeable time variable equal to the difference between the communication session time and the remainder value. If at block **958** the remainder is not greater than zero, block **962** directs the RC processor to set the chargeable time variable equal to the contents of the

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communication session time from the RC stop message. The processor is then directed to block **964**. In addition, after executing block **954** or block **960**, the processor is directed to block **964**.

Block **964** directs the processor **202** of Figure **7** to determine whether or not the chargeable time variable is greater than or equal to the free time balance as determined from the free time field **742** of the subscriber account record shown in Figure **39**. If this condition is satisfied, block **966** of Figure **56A** directs the processor to set the free time field **742** in the record shown in Figure **39**, to zero. If the chargeable time variable is not greater than or equal to the free time balance, block **968** directs the RC processor to set a user cost variable to zero and Block **970** then decrements the free time field **742** of the subscriber account record for the caller by the chargeable time amount determined by block **954**, **960** or **962**.

If at Block **964** the processor **202** of Figure **7** was directed to Block **966** which causes the free time field (**742** of Figure **39**) to be set to zero, referring to Figure **56B**, Block **972** directs the processor to set a remaining chargeable time variable equal to the difference between the chargeable time and the contents of the free time field (**742** of Figure **39**). Block **974** then directs the processor to set the user cost variable equal to the product of the remaining chargeable time and the cost per second calculated at Block **750** in Figure **33B**. Block **976** then directs the processor to decrement the funds balance field (**740**) of the subscriber account record shown in Figure **39** by the contents of the user cost variable calculated at Block **974**.

After completing Block **976** or after completing Block **970** in Figure **56A**, block **978** of Figure **56B** directs the processor **202** of Figure **7** to calculate a reseller cost variable as the product of the reseller rate as indicated in the mark-up value field **810** of the system operator mark-up table record shown in Figure **45** and the communication session time determined at Block **916** in Figure **53**. Then, Block **980** of Figure **56B** directs the processor to add the

reseller cost to the reseller balance field **986** of a reseller account record of the type shown in Figure **57** at **982**.

The reseller account record includes a reseller ID field **984** and the aforementioned reseller balance field **986**. The reseller ID field **984** holds a reseller ID code, and the reseller balance field **986** holds an accumulated balance of charges.

Referring to Figure **58**, a specific reseller accounts record for the reseller "Klondike" is shown generally at **988**. In this record the reseller ID field **984** holds a code representing the reseller "Klondike" and the reseller balance field **986** holds a balance of $**100.02**. Thus, the contents of the reseller balance field **986** in Figure **58** are incremented by the reseller cost calculated at block **978** of Figure **56**B.

Still referring to Figure **56**B, after adding the reseller cost to the reseller balance field as indicated by Block **980**, Block **990** directs the processor to **202** of Figure **7** calculate a system operator cost as the product of the system operator cost per second, as set at block **822** in Figure **41**, and the communication session time as determined at Block **916** in Figure **53**. Block **992** then directs the processor to add the system operator cost value calculated at Block **990** to a system operator accounts table record of the type shown at **994** in Figure **59**. This record includes a system operator balance field **996** holding an accumulated charges balance. Referring to Figure **60** in the embodiment described, the system operator balance field **996** may hold the value $**1,000.02** for example, and to this value the system operator cost calculated at Block **990** is added when the processor executes Block **992** of Figure **56**B.

Ultimately, the final reseller balance **986** in Figure **58** holds a number representing an amount owed to the reseller by the system operator and the

system operator balance **996** of Figure **59** holds a number representing an amount of profit for the system operator.

While specific embodiments of the invention have been described and illustrated, such embodiments should be considered illustrative of the invention only and not as limiting the invention as construed in accordance with the accompanying claims.

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What is claimed is:

1. A process for operating a call routing controller to facilitate communication between callers and callees in a system comprising a plurality of nodes with which callers and callees are associated, the process comprising:

in response to initiation of a call by a calling subscriber, receiving a caller identifier and a callee identifier;

using call classification criteria associated with the caller identifier to classify the call as a public network call or a private network call;

producing a routing message identifying an address, on the private network, associated with the callee when the call is classified as a private network call; and

producing a routing message identifying a gateway to the public network when the call is classified as a public network call.

2. The process of claim **1** further comprising receiving a request to establish a call, from a call controller in communication with a caller identified by said callee identifier.

3. The process of claim **1** wherein using said call classification criteria comprises searching a database to locate a record identifying calling attributes associated with a caller identified by said caller identifier.

4. The process of claim **3** wherein locating a record comprises locating a caller dialing profile comprising a username associated with said caller, a domain associated with said caller, and at least one calling attribute.

5. The process of claim 4 wherein using said call classification criteria comprises comparing calling attributes associated with said caller dialing profile with aspects of said callee identifier.

6. The process of claim 4 wherein comparing comprises determining whether said callee identifier includes a portion that matches an IDD associated with said caller dialing profile.

7. The process of claim 4 wherein comparing comprises determining whether said callee identifier includes a portion that matches an NDD associated with said caller dialing profile.

8. The process of claim 4 wherein comparing comprises determining whether said callee identifier includes a portion that matches an area code associated with said caller dialing profile.

9. The process of claim 4 wherein comparing comprises determining whether said callee identifier has a length within a range specified in said caller dialing profile.

10. The process of claim 4 further comprising formatting said callee identifier into a pre-defined digit format to produce a re-formatted callee identifier.

11. The process of claim 10 wherein formatting comprises removing an international dialing digit from said callee identifier, when said callee identifier begins with a digit matching an international dialing digit specified by said caller dialing profile associated with said caller.

12. The process of claim 10 wherein formatting comprises removing a national dialing digit from said callee identifier and prepending a caller

country code to said callee identifier when said callee identifier begins with a national dialing digit.

13. The process of claim **10** wherein formatting comprises prepending a caller country code to said callee identifier when said callee identifier begins with digits identifying an area code specified by said caller dialing profile.

14. The process of claim **10** wherein formatting comprises prepending a caller country code and area code to said callee identifier when said callee identifier has a length that matches a caller dialing number format specified by said caller dialing profile and only one area code is specified as being associated with said caller in said caller dialing profile.

15. The process of claim **10** further comprising classifying said call as a private network call when said re-formatted callee identifier identifies a subscriber to the private network.

16. The process of claim **10** further comprising determining whether said callee identifier complies with a pre-defined username format and if so classifying the call as a private network call.

17. The process of claim **10** further comprising causing a database of records to be searched to locate a direct in dial (DID) bank table record associating a public telephone number with said reformatted callee identifier and if said DID bank table record is found classifying the call as a private network call and if a DID bank table record is not found classifying the call as a public network call.

18. The process of claim **17** wherein producing said routing message identifying a node on the private network comprises setting a callee

identifier in response to a username associated with said DID bank table record.

19. The process of claim **18** wherein producing said routing message comprises determining whether a node associated with the reformatted callee identifier is the same as a node associated the caller identifier.

20. The process of claim **19** wherein determining whether a node associated with the reformatted callee identifier is the same as a node associated the caller identifier comprises determining whether a prefix of said re-formatted callee identifier matches a corresponding prefix of a username associated with said caller dialing profile.

21. The process of claim **20** wherein when said node associated with said caller is not the same as the node associated with the callee, producing a routing message including said caller identifier, said reformatted callee identifier and an identification of a private network node associated with said callee and communicating said routing message to a call controller.

22. The process of claim **19** wherein when said node associated with said caller is the same as the node associated with said callee, determining whether to perform at least one of the following: forward said call to another party, block the call and direct the caller to a voicemail server associated with the callee.

23. The process of claim **22** wherein producing said routing message comprises producing a routing message having an identification of at least one of the callee identifier, an identification of a party to whom the call should be forwarded and an identification of a voicemail server associated with the callee.

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24. The process of claim **23** further comprising communicating said routing message to a call controller.

25. The process of claim **10** wherein producing a routing message identifying a gateway to the public network comprises searching a database of route records associating route identifiers with dialing codes to find a route record having a dialing code having a number pattern matching at least a portion of said reformatted callee identifier.

26. The process of claim **25** further comprising searching a database of supplier records associating supplier identifiers with said route identifiers to locate at least one supplier record associated with said route identifier associated with said route record having a dialing code having a number pattern matching at least a portion of said reformatted callee identifier.

27. The process of claim **26** further comprising loading a routing message buffer with the reformatted callee identifier and an identification of specific routes associated respective ones of the supplier records associated with said route record and loading said routing message buffer with a time value and a timeout value.

28. The process of claim **27** further comprising communicating a routing message comprising the contents of said routing message buffer to a call controller.

29. The process of claim **4** further comprising causing said dialing profile to include a maximum concurrent call value and a concurrent call count value and causing said concurrent call count value to be incremented when the user associated with said dialing profile initiates a call and causing said concurrent call count value to be decremented when a call with said user associated with said dialing profile is ended.

30. A computer readable medium encoded with codes for directing a processor to execute the method of any one of claims **1-29**.

31. A call routing apparatus for facilitating communications between callers and callees in a system comprising a plurality of nodes with which callers and callees are associated, the apparatus comprising:

receiving means for receiving a caller identifier and a callee identifier, in response to initiation of a call by a calling subscriber;

classifying means for classifying the call as a private network cal or a public network call according to call classification criteria associated with the caller identifier;

means for producing a routing message identifying an address, on the private network, associated with the callee when the call is classified as a private network call; and

means for producing a routing message identifying a gateway to the public network if the call is classified as a public network call.

32. The apparatus of claim **31** wherein said receiving means is operably configured to receive a request to establish a call, from a call controller in communication with a caller identified by said callee identifier.

33. The apparatus of claim **31** further comprising searching means for searching a database comprising records associating calling attributes with subscribers to said private network to locate a record identifying calling attributes associated with a caller identified by said caller identifier.

34. The apparatus of claim **33** wherein said records include dialing profiles each comprising a username associated with said subscriber, an identification of a domain associated with said subscriber, and an identification of at least one calling attribute associated with said subscriber.

35. The apparatus of claim **34** wherein said call classification means is operably configured to compare calling attributes associated with said caller dialing profile with aspects of said callee identifier.

36. The apparatus of claim **35** wherein said calling attributes include an international dialing digit and wherein said call classification means is operably configured to determine whether said callee identifier includes a portion that matches an IDD associated with said caller dialing profile.

37. The apparatus of claim **34** wherein said calling attributes include an national dialing digit and wherein said call classification means is operably configured to determine whether said callee identifier includes a portion that matches an NDD associated with said caller dialing profile.

38. The apparatus of claim **34** wherein said calling attributes include an area code and wherein said call classification means is operably configured to determine whether said callee identifier includes a portion that matches an area code associated with said caller dialing profile.

39. The apparatus of claim **34** wherein said calling attribute include a number length range and wherein said call classification means is operably configured to determine whether said callee identifier has a length within a range specified in said caller dialing profile.

40. The apparatus of claim **34** further comprising formatting means for formatting said callee identifier into a pre-defined digit format to produce a re-formatted callee identifier.

41. The apparatus of claim **40** wherein said formatting means is operably configured to remove an international dialing digit from said callee identifier, when said callee identifier begins with a digit matching an international dialing digit specified by said caller dialing profile associated with said caller.

42. The apparatus of claim **40** wherein said formatting means is operably configured to remove a national dialing digit from said callee identifier and prepend a caller country code to said callee identifier when said callee identifier begins with a national dialing digit.

43. The apparatus of claim **40** wherein said formatting means is operably configured to prepend a caller country code to said callee identifier when said callee identifier begins with digits identifying an area code specified by said caller dialing profile.

44. The apparatus of claim **40** wherein said formatting means is operably configured to prepend a caller country code and area code to said callee identifier when said callee identifier has a length that matches a caller dialing number format specified by said caller dialing profile and only one area code is specified as being associated with said caller in said caller dialing profile.

45. The apparatus of claim **40** wherein said classifying means is operably configured to classifying said call as a private network call when said re-formatted callee identifier identifies a subscriber to the private network.

46. The apparatus of claim **40** wherein said classifying means is operably configured to classify the call as a private network call when said callee identifier complies with a pre-defined username format.

47. The apparatus of claim **40** further comprising searching means for searching a database of records to locate a direct in dial (DID) bank table record associating a public telephone number with said reformatted callee identifier and wherein said classifying means is operably configured to classify the call as a private network call when said DID bank table record is found and to classify the call as a public network call when a DID bank table record is not found

48. The apparatus of claim **47** wherein said private network routing message producing means is operably configured to produce a routing message having a callee identifier set according to a username associated with said DID bank table record.

49. The apparatus of claim **48** wherein said private network routing message producing means is operably configured to determine whether a node associated with the reformatted callee identifier is the same as a node associated the caller identifier.

50. The apparatus of claim **49** wherein said private network routing means includes means for determining whether a prefix of said re-formatted callee identifier matches a corresponding prefix of a username associated with said caller dialing profile.

51. The apparatus of claim **50** wherein said private network routing message producing means is operably configured to produce a routing message including said caller identifier, said reformatted callee identifier and an identification of a private network node associated with

said callee and communicating said routing message to a call controller.

52. The apparatus of claim **49** wherein said private network routing message producing means is operably configured to perform at least one of the following: forward said call to another party, block the call and direct the caller to a voicemail server associated with the callee, when said node associated with said caller is the same as the node associated with said callee.

53. The apparatus of claim **52** wherein said means for producing said private network routing message is operably configured to produce a routing message having an identification of at least one of the callee identifier, an identification of a party to whom the call should be forwarded and an identification of a voicemail server associated with the callee.

54. The apparatus of claim **53** further comprising means for communicating said routing message to a call controller.

55. The apparatus of claim **40** wherein said means for producing a public network routing message identifying a gateway to the public network comprises means for searching a database of route records associating route identifiers with dialing codes to find a route record having a dialing code having a number pattern matching at least a portion of said reformatted callee identifier.

56. The apparatus of claim **55** further comprising means for searching a database of supplier records associating supplier identifiers with said route identifiers to locate at least one supplier record associated with said route identifier associated with said route record having a dialing

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code having a number pattern matching at least a portion of said reformatted callee identifier.

57. The apparatus of claim **56** further comprising a routing message buffer and means for loading said routing message buffer with the reformatted callee identifier and an identification of specific routes associated respective ones of the supplier records associated with said route record and loading said routing message buffer with a time value and a timeout value.

58. The apparatus of claim **57** further comprising means for communicating a routing message comprising the contents of said routing message buffer to a call controller.

59. The apparatus of claim **34** further comprising means for causing said dialing profile to include a maximum concurrent call value and a concurrent call count value and for causing said concurrent call count value to be incremented when the user associated with said dialing profile initiates a call and for causing said concurrent call count value to be decremented when a call with said user associated with said dialing profile is ended.

Data Structure

60. A data structure for access by an apparatus for producing a routing message for use by a call routing controller in a communications system, the data structure comprising:

dialing profile records comprising fields for associating with respective subscribers to the system:

a subscriber user name

direct-in-dial records comprising fields for associating with respective subscriber usernames:

a user domain; and

a direct-in-dial number;

prefix to node records comprising fields for associating with at least a portion of said respective subscriber usernames:

a node address of a node in said system,

whereby a subscriber name can be used to find a user domain, at least a portion of said a subscriber name can be used to find a node with which the subscriber identified by the subscriber name is associated, and a user domain and subscriber name can be located in response to a direct-in-dial number.

61. A data structure for access by an apparatus for producing a routing message for use by a call routing controller in a communications system, the data structure comprising:

master list records comprising fields for associating a dialing code with respective master list identifiers; and

supplier list records linked to master list records by said master list identifiers, aid supplier list records comprising fields for associating with a communications services supplier:

a supplier id;

a master list id;

a route identifier; and

a billing rate code,

whereby communications services suppliers are associated with dialing codes, such that dialing codes can be used to locate suppliers capable of providing a communications link associated with a given dialing code.

62. A method of determining a time to permit a communication session to be conducted, the method comprising:

calculating a cost per unit time;

calculating a first time value as a sum of a free time attributed to a participant in the communication session and the quotient of a funds balance held by said participant to said cost per unit time value; and

producing a second time value in response to said first time value and a billing pattern associated with said participant, said billing pattern including first and second billing intervals and said second time value being said time to permit a communication session to be conducted.

63. The method of claim **62** wherein calculating said first time value comprises retrieving a record associated with said participant and obtaining from said record at least one of said free time and said funds balance.

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64. The method of claim **62** wherein producing said second time value comprises producing a remainder value representing a portion of said second billing interval remaining after dividing said second billing interval into a difference between said first time value and said first billing interval.

65. The method of claim **64** wherein producing said second time value comprises setting a difference between said first time value and said remainder as said second time value.

66. The method of claim **62** further comprising setting said second time value to zero when said remainder is greater than zero and said first time value is less than said free time associated with said participant.

67. The method of claim **62** wherein calculating said cost per unit time comprises:

locating a record in a database, said record comprising a markup type indicator, a markup value and a billing pattern;

and

setting a reseller rate equal to the sum of said markup value and said buffer rate.

68. The method of claim **67** wherein locating said record in a database comprises locating at least one of:

a record associated with a reseller and a route associated with the reseller;

a record associated with the reseller; and

a default reseller markup record.

69. The method of claim **67** wherein calculating said cost per unit time value further comprises locating at least one of:

an override record specifying a route cost per unit time amount associated with a route associated with the communication session;

a reseller record associated with a reseller of said communications session, said reseller record specifying a reseller cost per unit time associated with said reseller for the communication session;

a default operator markup record specifying a default cost per unit time.

70. The method of claim **69** further comprising setting as said cost per unit time the sum of said reseller rate and at least one of said route cost per unit time, said reseller cost per unit time and said default cost per unit time.

71. The method of claim **69** further comprising receiving a communication session time representing a duration of said communication session and incrementing a reseller balance by the product of said reseller rate and said communication session time.

72. The method of claim **69** further comprising receiving a communication session time representing a duration of said communication session and incrementing a system operator balance by a product of said buffer rate and said communication session time.

73. A computer readable medium encoded with instructions for directing a processor circuit to execute the method of any one of claims **62-72**.

74. An apparatus for determining a time to permit a communication session to be conducted, the apparatus comprising:

a processor circuit;

a computer readable medium coupled to the processor circuit and encoded with instructions for directing the processor circuit to:

calculate a cost per unit time for the communication session;

calculate a first time value as a sum of a free time attributed to a participant in the communication session and the quotient of a funds balance held by said participant to said cost per unit time value; and

produce a second time value in response to said first time value and a billing pattern associated with said participant, said billing pattern including first and second billing intervals and said second time value being said time to permit a communication session to be conducted.

75. The apparatus of claim **74** wherein said instructions include instructions for directing the processor circuit to retrieve a record associated with said participant and obtain from said record at least one of said free time and said funds balance.

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76. The apparatus of claim **74** wherein said instructions include instructions for directing the processor circuit to produce said second time value by producing a remainder value representing a portion of said second billing interval remaining after dividing said second billing interval into a difference between said first time value and said first billing interval.

77. The apparatus of claim **76** wherein said instructions include instructions for directing the processor circuit to produce said second time value comprises setting a difference between said first time value and said remainder as said second time value.

78. The apparatus of claim **74** wherein said instructions include instructions for directing the processor circuit to set said second time value to zero when said remainder is greater than zero and said first time value is less than said free time associated with said participant.

79. The apparatus of claim **74** wherein said instructions for directing said processor circuit to calculate said cost per unit time comprises instructions for directing the processor circuit to:

locate a record in a database, said record comprising a markup type indicator, a markup value and a billing pattern;

and

set a reseller rate equal to the sum of said markup value and said buffer rate.

80. The apparatus of claim **79** wherein said instructions for directing the processor circuit to locate said record in a database comprises instruction for directing the processor circuit to locate at least one of:

a record associated with a reseller and a route associated with the reseller;

a record associated with the reseller;

a default reseller markup record;

81. The apparatus of claim **79** wherein said instructions for directing the processor circuit to calculate said cost per unit time value further comprises instructions for directing the processor circuit to locate at least one of:

an override record specifying a route cost per unit time amount associated with a route associated with the communication session;

a reseller record associated with a reseller of said communications session, said reseller record specifying a reseller cost per unit time associated with said reseller for the communication session;

a default operator markup record specifying a default cost per unit time.

82. The apparatus of claim **81** wherein said instructions include instructions for directing the processor circuit to set as said cost per unit time the sum of said reseller rate and at least one of said route

cost per unit time, said reseller cost per unit time and said default cost per unit time.

83. The apparatus of claim **81** wherein said instructions include instructions for directing the processor circuit to receive a communication session time representing a duration of said communication session and increment a reseller balance by the product of said reseller rate and said communication session time.

84. The apparatus of claim **81** wherein said instructions include instructions for directing the processor circuit to receive a communication session time representing a duration of said communication session and increment a system operator balance by a product of said buffer rate and said communication session time.

Attributing Charges to a User

85. A process for attributing charges for communications services, the process comprising:

determining a first chargeable time in response to a communication session time and a pre-defined billing pattern;

determining a user cost value in response to said first chargeable time and a free time value associated with a user of said communications services;

changing an account balance associated with said user in response to a user cost per unit time.

changing an account balance associated with a reseller of said communications services in response to a reseller cost per unit time and said communication session time; and

changing an account balance associated with an operator of said communications services in response to an operator cost per unit time and said communication session time.

86. The process of claim **85** wherein determining said first chargeable time comprises:

locating at least one of:

an override record specifying a route cost per unit time and billing pattern associated with a route associated with the communication session;

a reseller record associated with a reseller of said communications session, said reseller record specifying a reseller cost per unit time and billing pattern associated with said reseller for the communication session; and

a default record specifying a default cost per unit time and billing pattern; and

setting as said pre-defined billing pattern the billing pattern of the record located,

wherein the billing pattern of the record located comprises a first billing interval and a second billing interval.

87. The process of claim **85** wherein determining said first chargeable time comprises setting said first chargeable time equal to said first billing interval when said communication session time is less than or equal to said first billing interval.

88. The process of claim **86** wherein determining said first chargeable time comprises producing a remainder value representing a portion of said second billing interval remaining after dividing said second billing interval into a difference between communication session time and said first interval when said communication session time is greater than said communication session time; and

setting said first chargeable time to a difference between said communication session time and said remainder when said remainder is greater than zero; and

setting said first chargeable time to said communication session time when said remainder is not greater than zero.

89. The process of claim **88** further comprising determining a second chargeable time in response to said first chargeable time and said free time value associated with said user of said communications services when said first chargeable time is greater than or equal to said free time value associated with said user of said communications services.

90. The process of claim **89** wherein determining said second chargeable time comprises setting said second chargeable time to a difference between said first chargeable time.

91. The process of claim **89** further comprising resetting said free time value associated with the user to zero when said first chargeable time

is greater than or equal to said free time value associated with said user of said communications services.

92. The process of claim **90** wherein changing an account balance associated with the user comprises calculating a user cost value in response to said second chargeable time and said user cost per unit time.

93. The process of claim **92** further comprising changing a user free cost balance in response to said user cost value.

94. The process of claim **85** further comprising setting said user cost to zero when said first chargeable time is less than said free time value associated with the user.

95. The process of claim **85** further comprising changing a user free time balance in response to said first chargeable time.

96. A computer readable medium encoded with instructions for directing a processor circuit to execute the process of any one of claims **85-95**.

97. An apparatus for attributing charges for communications services, the apparatus comprising:

a processor circuit;

a computer readable medium in communication with the processor circuit and encoded with instructions for directing said processor circuit to;

determine a first chargeable time in response to a communication session time and a pre-defined billing pattern;

determine a user cost value in response to said first chargeable time and a free time value associated with a user of said communications services;

change an account balance associated with said user in response to a user cost per unit time.

change an account balance associated with a reseller of said communications services in response to a reseller cost per unit time and said communication session time; and

change an account balance associated with an operator of said communications services in response to an operator cost per unit time and said communication session time.

98. The apparatus of claim **97** wherein said instructions for directing the processor circuit to determine said first chargeable time comprises:

instructions for causing said processor circuit to communicate with a database to locate at least one of:

an override record specifying a route cost per unit time and billing pattern associated with a route associated with the communication session;

a reseller record associated with a reseller of said communications session, said reseller record specifying a reseller cost per unit time and billing pattern associated with said reseller for the communication session; and

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a default record specifying a default cost per unit time and billing pattern; and

instructions for setting as said pre-defined billing pattern the billing pattern of the record located,

wherein the billing pattern of the record located comprises a first billing interval and a second billing interval.

99. The apparatus of claim **97** wherein said instructions causing the processor circuit to determine said first chargeable time comprises instructions for directing the processor circuit to set said first chargeable time equal to said first billing interval when said communication session time is less than or equal to said first billing interval.

100. The apparatus of claim **98** wherein said instructions for causing the processor circuit to determine said first chargeable time comprises instructions for producing a remainder value representing a portion of said second billing interval remaining after dividing said second billing interval into a difference between communication session time and said first interval when said communication session time is greater than said communication session time; and

instructions for causing the processor circuit to set said first chargeable time to a difference between said communication session time and said remainder when said remainder is greater than zero; and

instructions for causing the processor circuit to set said first chargeable time to said communication session time when said remainder is not greater than zero.

101. The apparatus of claim **100** wherein the computer readable medium is further encoded with instructions for causing the processor circuit to determine a second chargeable time in response to said first chargeable time and said free time value associated with said user of said communications services when said first chargeable time is greater than or equal to said free time value associated with said user of said communications services.

102. The apparatus of claim **101** wherein said instructions for causing the processor circuit to determine said second chargeable time comprises instructions for causing the processor circuit to set said second chargeable time to a difference between said first chargeable time.

103. The apparatus of claim **101** wherein the computer readable medium is further encoded with instructions for causing the processor circuit to reset said free time value associated with the user to zero when said first chargeable time is greater than or equal to said free time value associated with said user of said communications services.

104. The apparatus of claim **102** wherein said instructions for causing the processor circuit to change an account balance associated with the user comprises instructions for causing the processor circuit to calculate a user cost value in response to said second chargeable time and said user cost per unit time.

105. The apparatus of claim **104** wherein the computer readable medium is further encoded with instructions for causing the processor circuit to change a user free cost balance in response to said user cost value.

106. The apparatus of claim **97** wherein the computer readable medium is further encoded with instructions for causing the processor circuit to

set said user cost to zero when said first chargeable time is less than said free time value associated with the user.

107. The apparatus of claim **97** wherein the computer readable medium is further encoded with instructions for causing the processor circuit to change a user free time balance in response to said first chargeable time.

AMENDED CLAIMS
received by the International Bureau on 18 April 2008 (18.04.08)

code having a number pattern matching at least a portion of said reformatted callee identifier.

57. The apparatus of claim 56 further comprising a routing message buffer and means for loading said routing message buffer with the reformatted callee identifier and an identification of specific routes associated respective ones of the supplier records associated with said route record and loading said routing message buffer with a time value and a timeout value.

58. The apparatus of claim 57 further comprising means for communicating a routing message comprising the contents of said routing message buffer to a call controller.

59. The apparatus of claim 34 further comprising means for causing said dialing profile to include a maximum concurrent call value and a concurrent call count value and for causing said concurrent call count value to be incremented when the user associated with said dialing profile initiates a call and for causing said concurrent call count value to be decremented when a call with said user associated with said dialing profile is ended.

Data Structure

60. A data structure for access by an apparatus for producing a routing message for use by a call routing controller in a communications system, the data structure comprising:

dialing profile records comprising fields for associating a subscriber username with respective subscribers to the system;

direct-in-dial records comprising fields for associating a user domain and a direct-in-dial number with respective subscriber usernames;

prefix to node records comprising fields for associating a node address of a node in said system with at least a portion of said respective subscriber usernames:

whereby said subscriber username can be used to find said user domain, at least a portion of said subscriber username can be used to find said node with which a subscriber identified by said subscriber user name is associated, and said user domain and said subscriber username can be located in response to said direct-in-dial number.

61. A data structure for access by an apparatus for producing a routing message for use by a call routing controller in a communications system, the data structure comprising:

master list records comprising fields for associating a dialing code with respective master list identifiers; and

supplier list records linked to said master list records by said master list identifiers, said supplier list records comprising fields for associating with a communications services supplier:

a supplier id;

a master list id;

a route identifier; and

a billing rate code,

whereby at least one communications service supplier is associated with said dialing code, such that said dialing code can be used to locate suppliers capable of providing a communications link associated with a given dialing code.

62. A method of determining a time to permit a communication session to be conducted, the method comprising:

calculating a cost per unit time;

calculating a first time value as a sum of a free time attributed to a participant in the communication session and the quotient of a funds balance held by said participant to said cost per unit time value; and

producing a second time value in response to said first time value and a billing pattern associated with said participant, said billing pattern including first and second billing intervals and said second time value being said time to permit a communication session to be conducted.

63. The method of claim 62 wherein calculating said first time value comprises retrieving a record associated with said participant and obtaining from said record at least one of said free time and said funds balance.

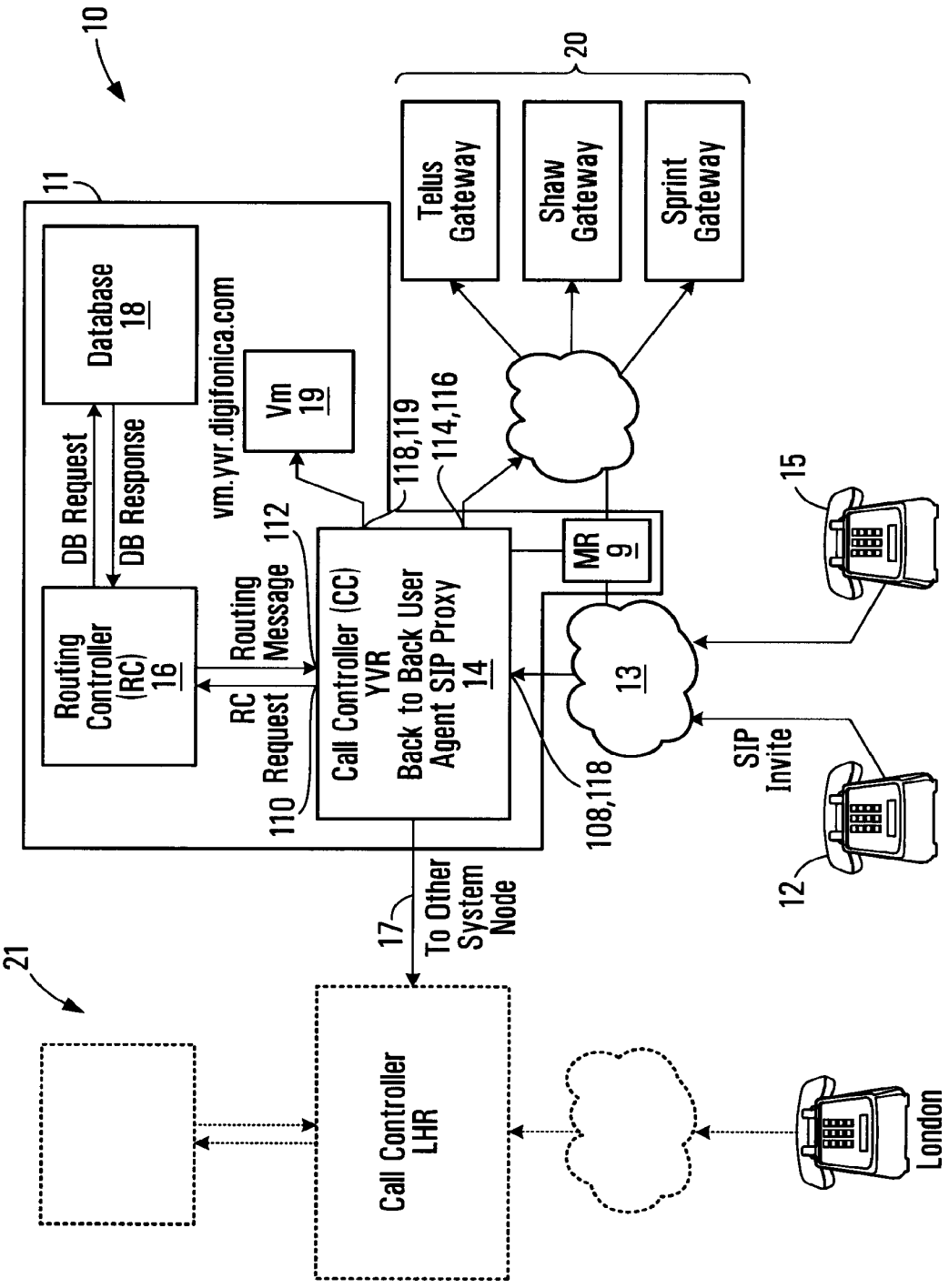

10
11
20
Telus Gateway
Shaw Gateway
Sprint Gateway
Database 18
Vm 19
vm.yvr.digifonica.com
DB Request
DB Response
Routing Controller (RC) 16
Routing Message 112
RC Request 110
Call Controller (CC) YVR Back to Back User Agent SIP Proxy 14
118,119
114,116
MR 9
13
108,118
15
2001 1050 2222 Calgary
SIP Invite
12
192.168.0.20 2001 1050 8667 Vancouver
17
To Other System Node
21
Call Controller LHR
London

FIG. 1

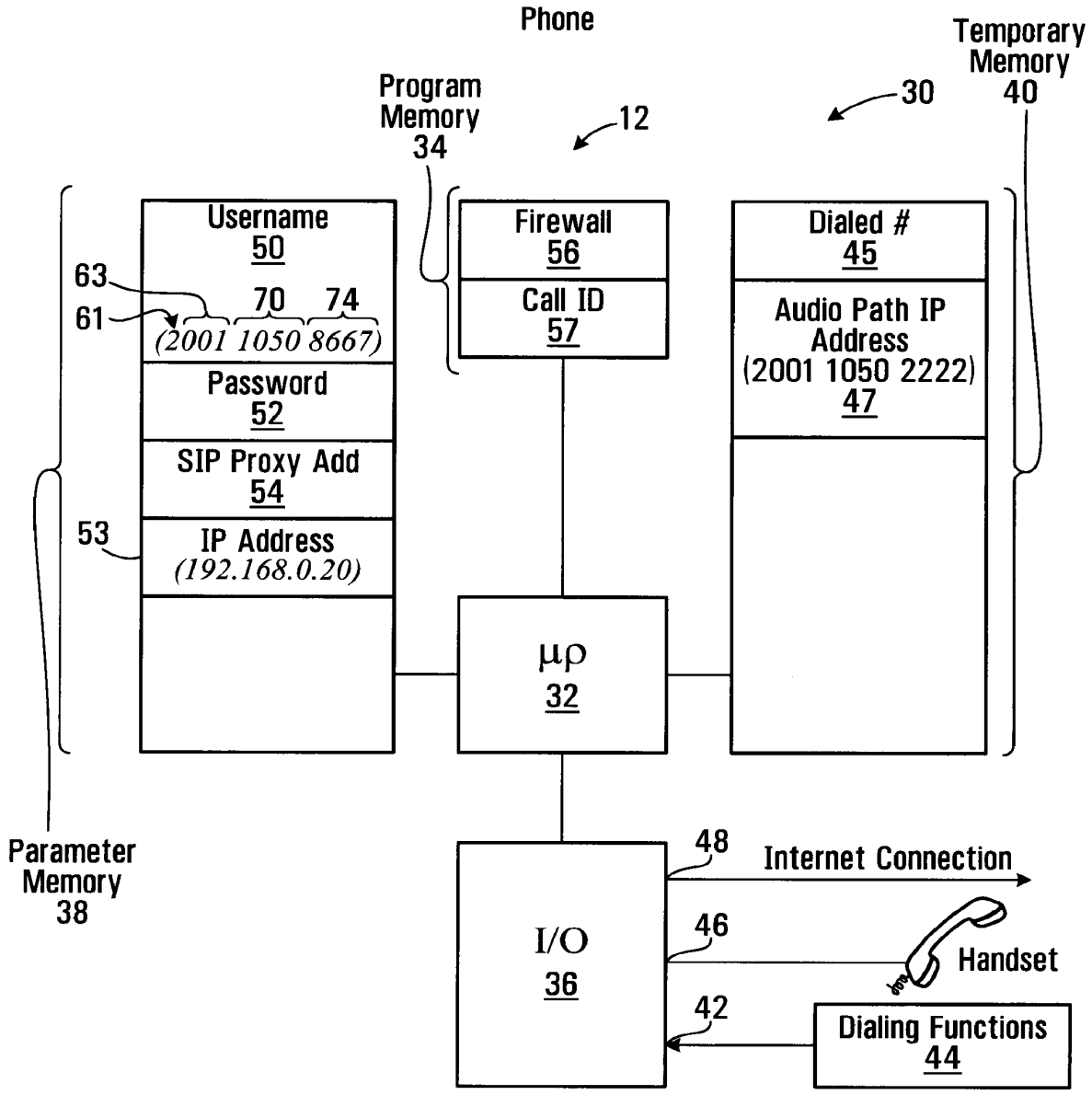

Phone
Temporary Memory 40
Program Memory 34
12
30
Username 50
63
61
70
74
(2001 1050 8667)
Password 52
SIP Proxy Add 54
53
IP Address (192.168.0.20)
Firewall 56
Call ID 57
Dialed # 45
Audio Path IP Address (2001 1050 2222) 47
μρ 32
Parameter Memory 38
I/O 36
48
Internet Connection
46
Handset
42
Dialing Functions 44

FIG. 2

SIP Invite Message

60	Caller	2001 1050 8667
62	Callee	2001 1050 2222
64	Digest Parameters	XXXXXXX
65	Call ID	FF10@ 192.168.0.20
67	IP Address	192.168.0.20
69	Caller UDP Port	1

FIG. 3

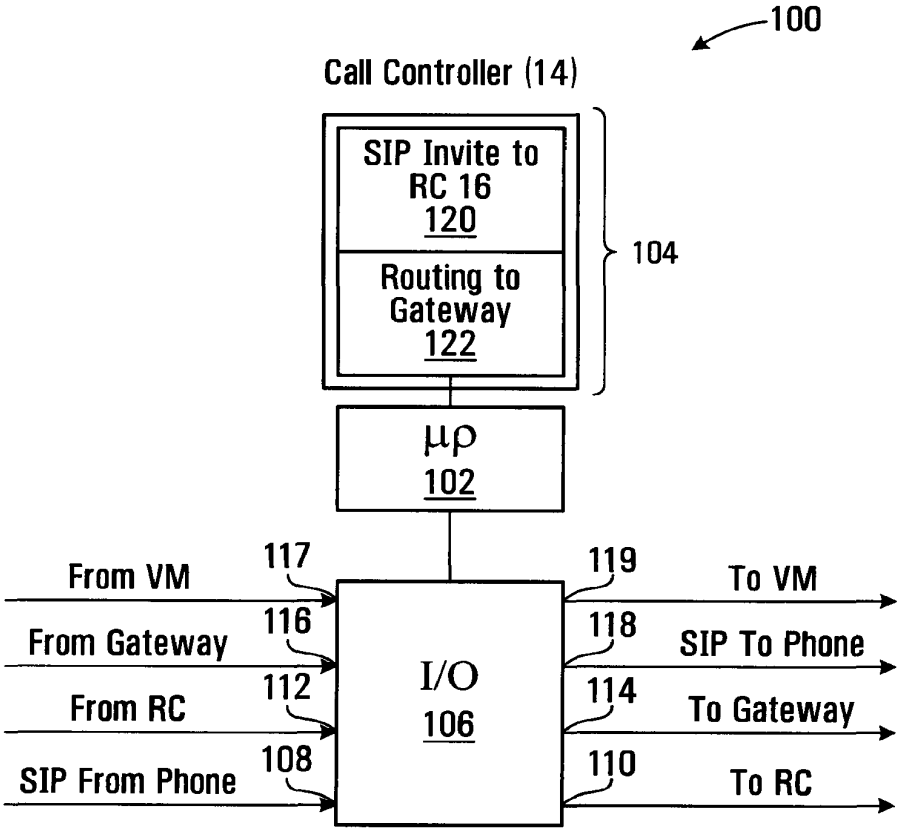


FIG. 4

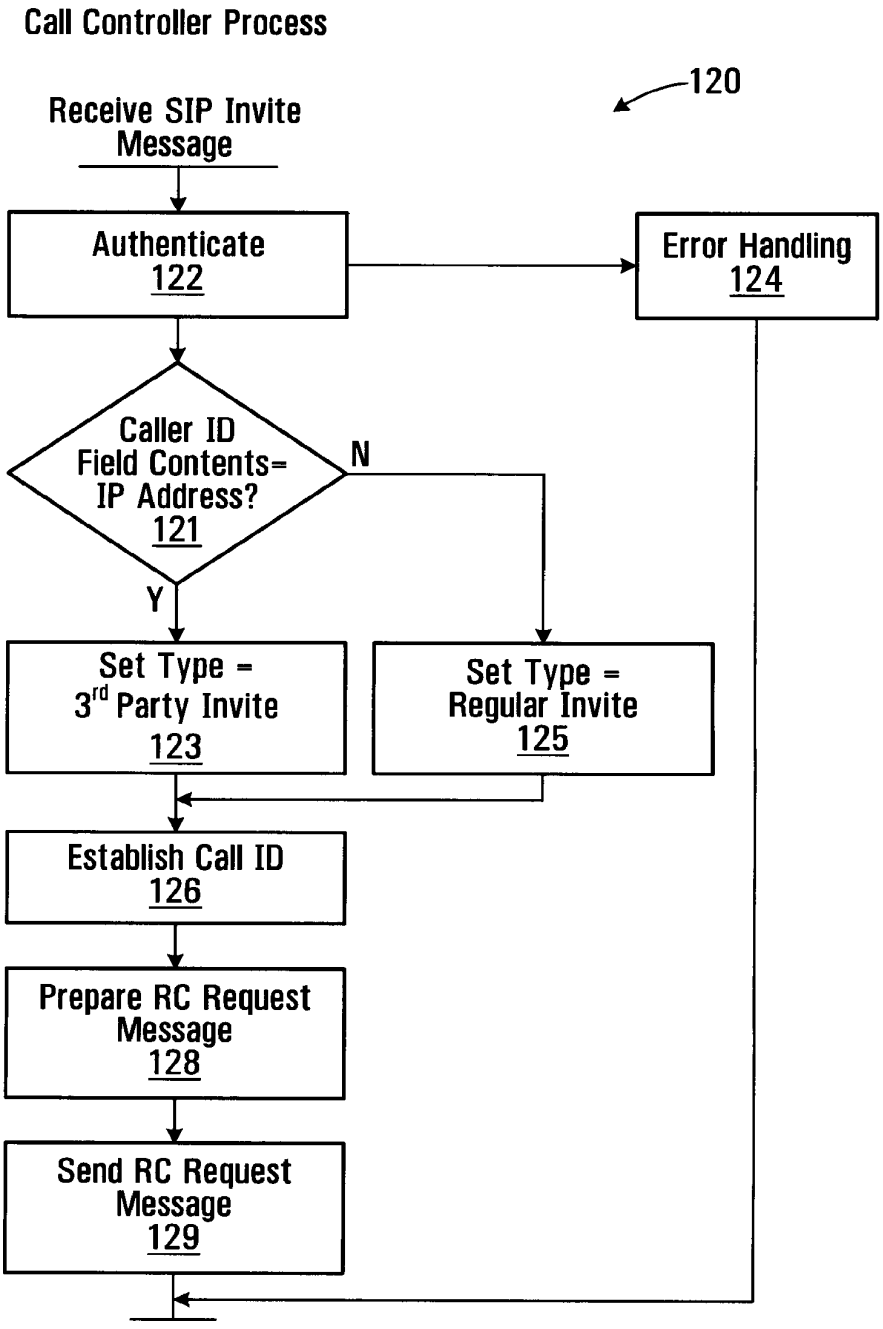

Call Controller Process
120
Receive SIP Invite Message
Authenticate 122
Error Handling 124
Caller ID Field Contents= IP Address? 121
N
Y
Set Type = 3rd Party Invite 123
Set Type = Regular Invite 125
Establish Call ID 126
Prepare RC Request Message 128
Send RC Request Message 129

FIG. 5

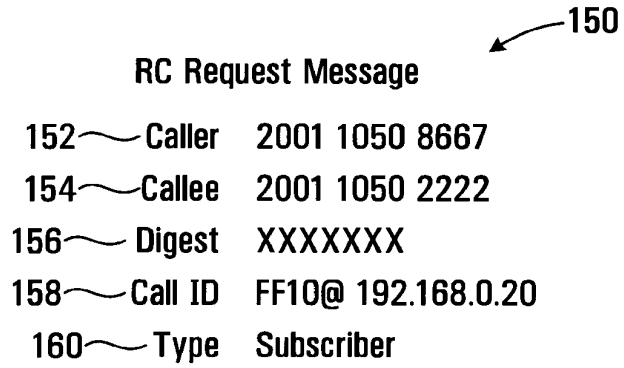

150
RC Request Message
152 Caller 2001 1050 8667
154 Callee 2001 1050 2222
156 Digest XXXXXXX
158 Call ID FF10@ 192.168.0.20
160 Type Subscriber

FIG. 6

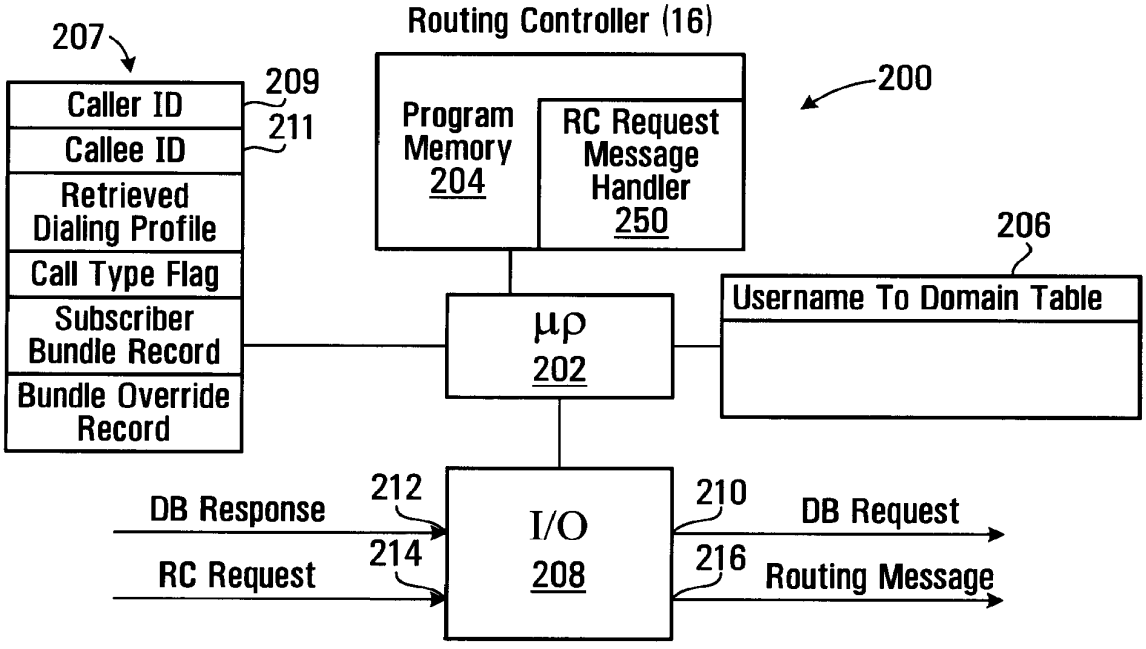

Routing Controller (16)
207
Caller ID
Callee ID
Retrieved Dialing Profile
Call Type Flag
Subscriber Bundle Record
Bundle Override Record
209
211
Program Memory 204
RC Request Message Handler 250
200
206
Username To Domain Table
μρ 202
DB Response
RC Request
212
214
I/O 208
210
216
DB Request
Routing Message

FIG. 7

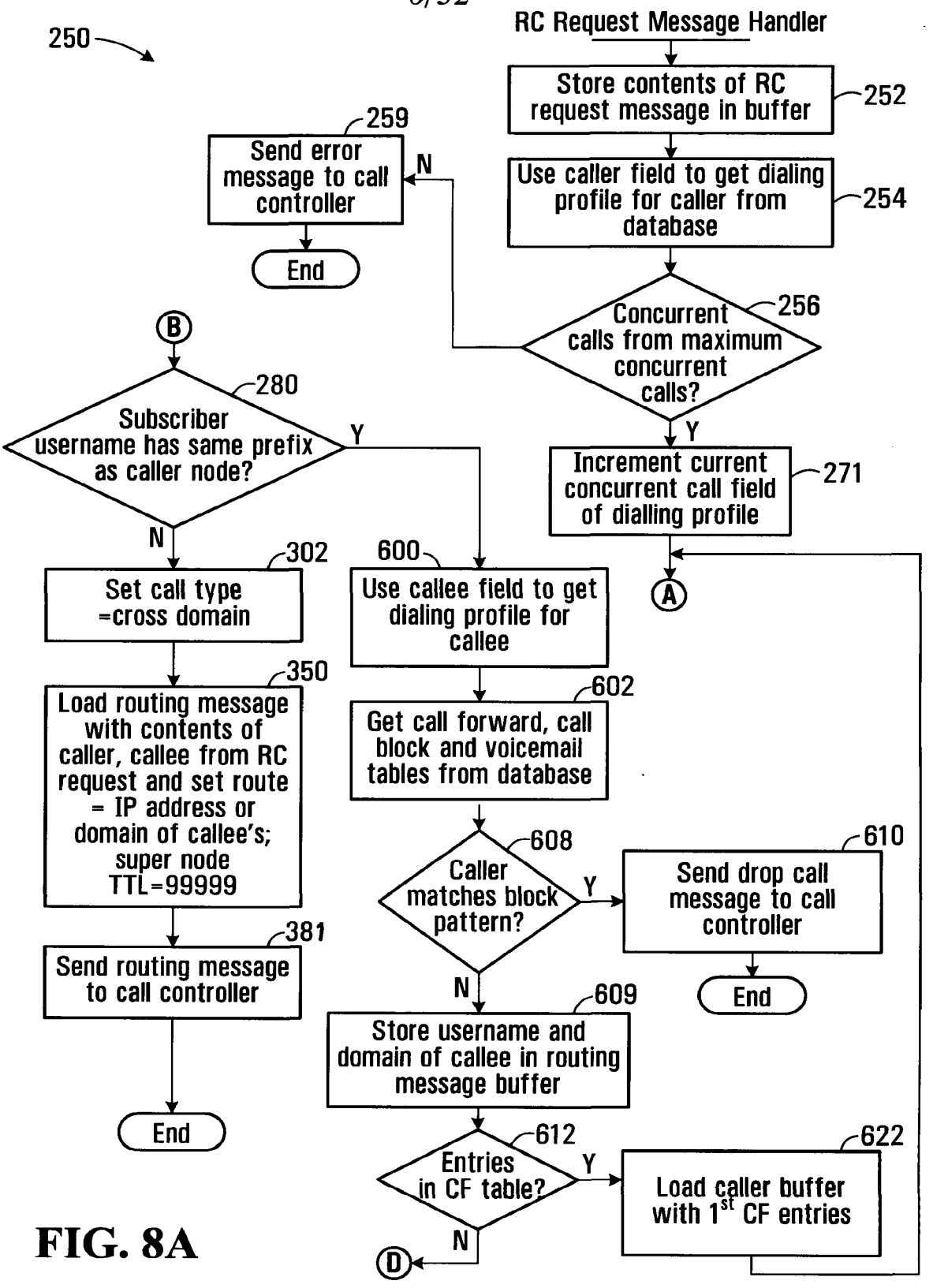

250
RC Request Message Handler
Store contents of RC request message in buffer
252
Use caller field to get dialing profile for caller from database
254
Concurrent calls from maximum concurrent calls?
256
N
Y
Send error message to call controller
259
End
Increment current concurrent call field of dialling profile
271
A
B
Subscriber username has same prefix as caller node?
280
Y
N
Set call type =cross domain
302
Load routing message with contents of caller, callee from RC request and set route = IP address or domain of callee's; super node TTL=99999
350
Send routing message to call controller
381
End
600
Use callee field to get dialing profile for callee
602
Get call forward, call block and voicemail tables from database
608
Caller matches block pattern?
Y
N
Send drop call message to call controller
610
End
609
Store username and domain of callee in routing message buffer
612
Entries in CF table?
Y
N
D
622
Load caller buffer with 1st CF entries

FIG. 8A

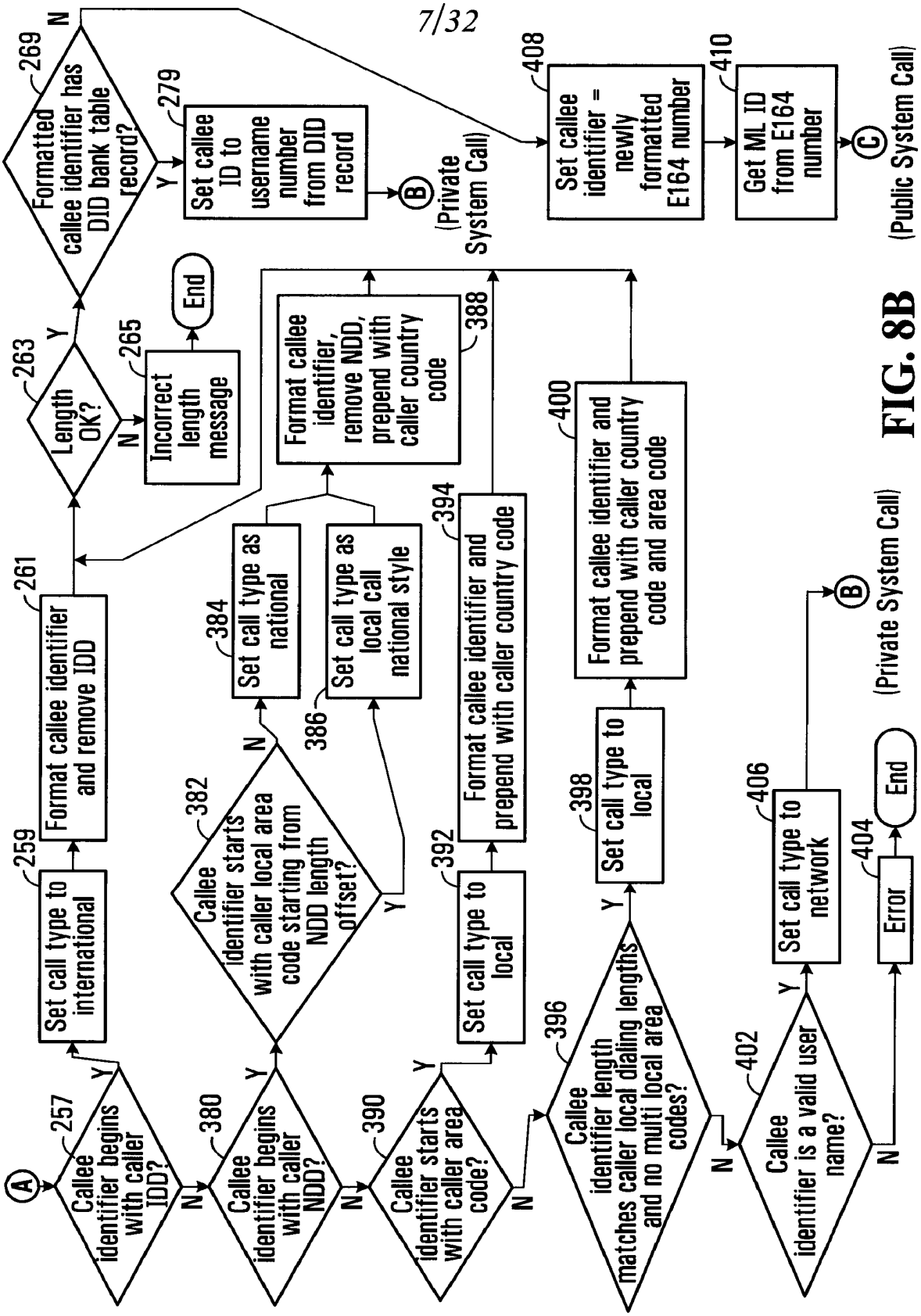

A
257
Callee identifier begins with caller IDD?
Y
259
Set call type to international
261
Format callee identifier and remove IDD
N
380
Callee identifier begins with caller NDD?
Y
382
Callee identifier starts with caller local area code starting from NDD length offset?
N
384
Set call type as national
Y
386
Set call type as local call national style
388
Format callee identifier, remove NDD, prepend with caller country code
N
390
Callee identifier starts with caller area code?
Y
392
Set call type to local
394
Format callee identifier and prepend with caller country code
N
396
Callee identifier length matches caller local dialing lengths and no multi local area codes?
Y
398
Set call type to local
400
Format callee identifier and prepend with caller country code and area code
N
402
Callee identifier is a valid user name?
Y
406
Set call type to network
B
(Private System Call)
N
404
Error
End
263
Length OK?
Y
265
Incorrect length message
End
N
269
Formatted callee identifier has DID bank table record?
Y
279
Set callee ID to username number from DID record
B
(Private System Call)
N
408
Set callee identifier = newly formatted E164 number
410
Get ML ID from E164 number
C
(Public System Call)

FIG. 8B

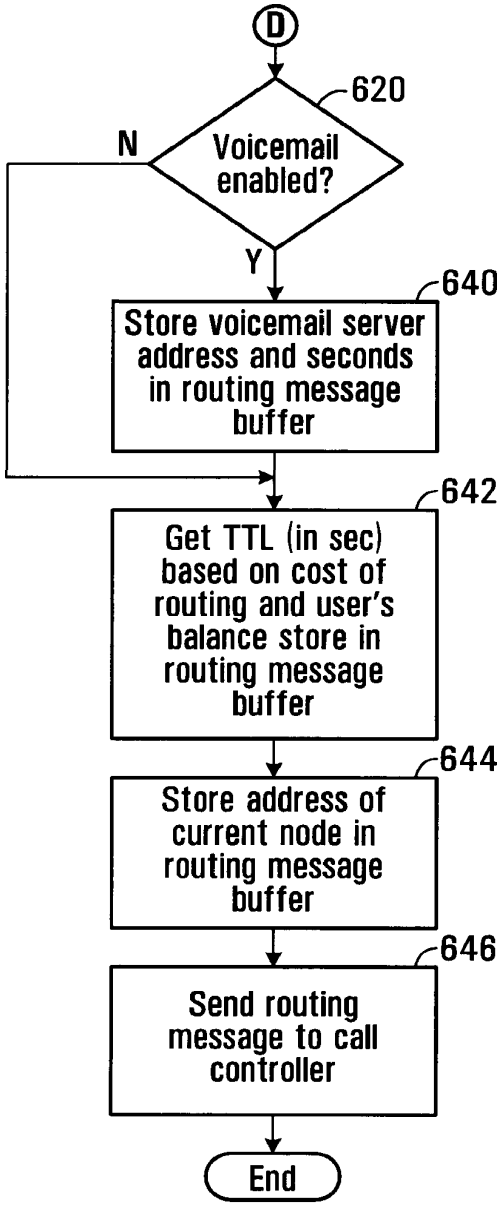

D
620
Voicemail enabled?
N
Y
640
Store voicemail server address and seconds in routing message buffer
642
Get TTL (in sec) based on cost of routing and user's balance store in routing message buffer
644
Store address of current node in routing message buffer
646
Send routing message to call controller
End

FIG. 8C

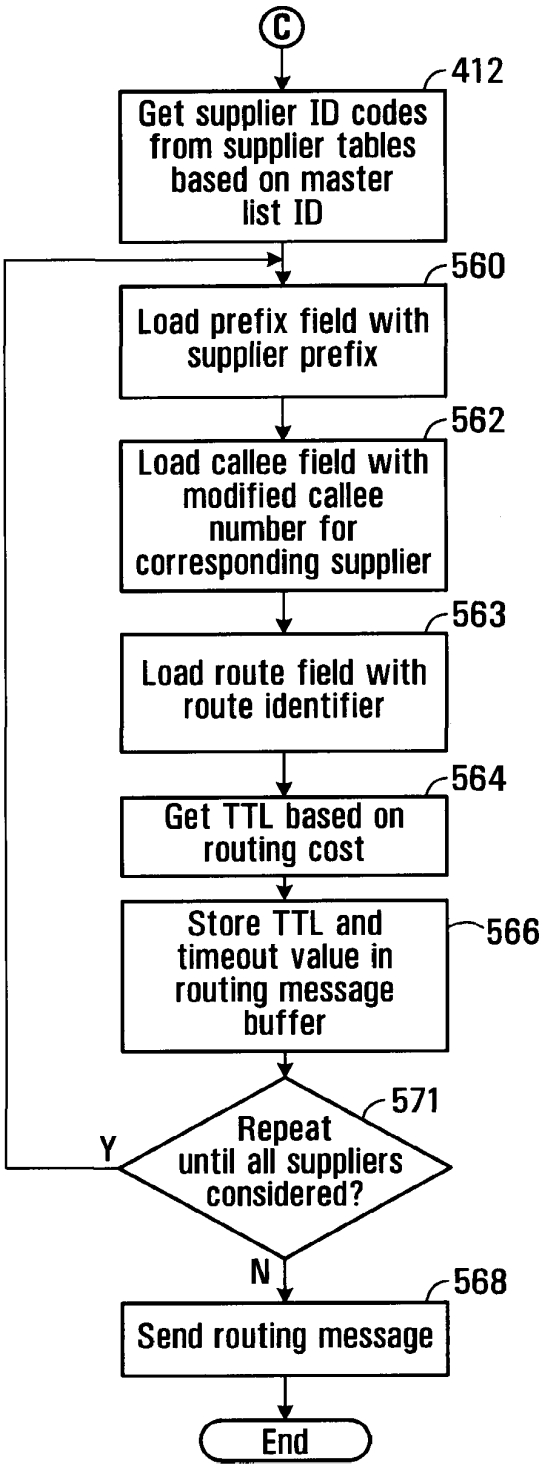

C
412
Get supplier ID codes from supplier tables based on master list ID
560
Load prefix field with supplier prefix
562
Load callee field with modified callee number for corresponding supplier
563
Load route field with route identifier
564
Get TTL based on routing cost
566
Store TTL and timeout value in routing message buffer
571
Repeat until all suppliers considered?
Y
N
568
Send routing message
End

FIG. 8D

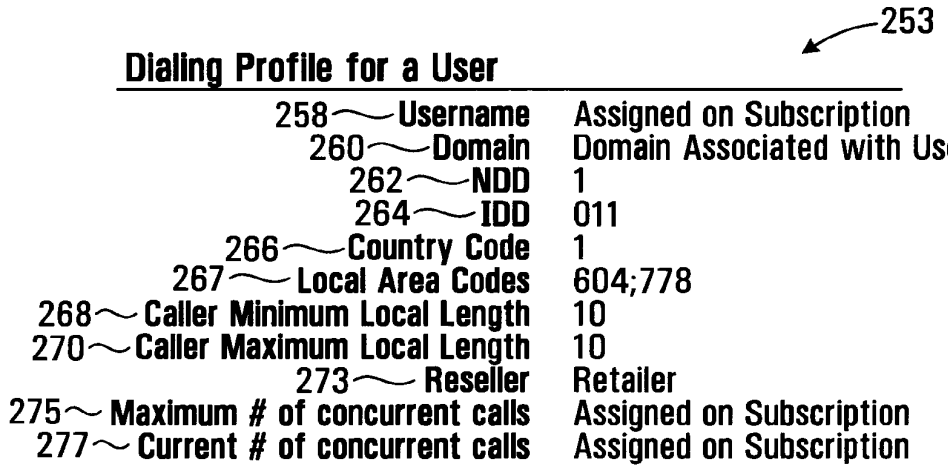


FIG. 9

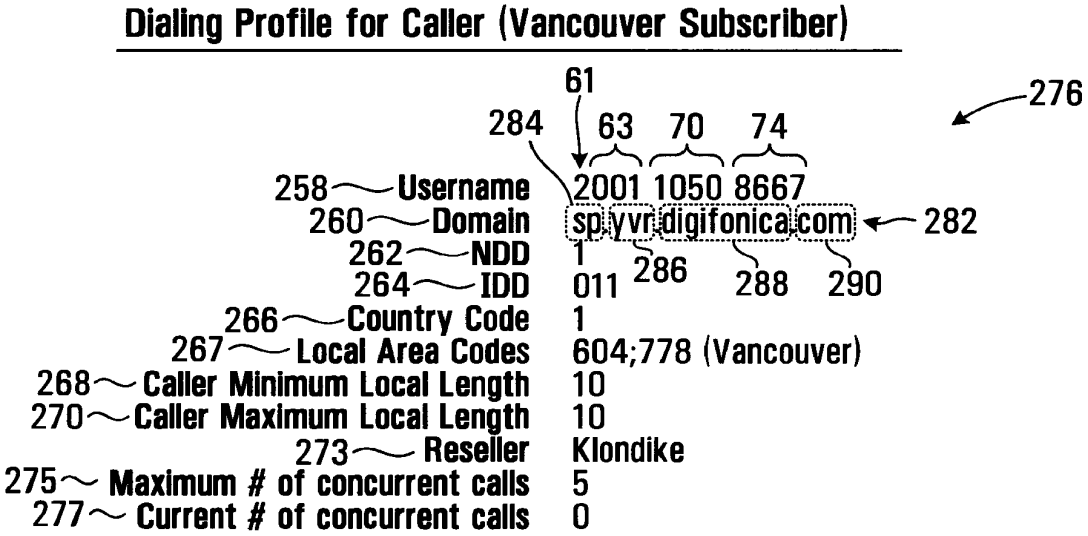


FIG. 10

META 1002
META V. VOIP-PAL

Callee Profile for Calgary Subscriber	
Username	2001 1050 2222
Domain	sp.yvr.digifonica.com
NDD	1
IDD	011
Country Code	1
Local Area Codes	403 (Calgary)
Caller Minimum Local Length	7
Caller Maximum Local Length	10
Reseller	Deerfoot
Maximum # of concurrent calls	5
Current # of concurrent calls	0

FIG. 11

Callee Profile for London Subscriber	
Username	4401 1062 4444
Domain	sp.lhr.digifonica.com
NDD	0
IDD	00
Country Code	44
Local Area Codes	20 (London)
Caller Minimum Local Length	10
Caller Maximum Local Length	11
Reseller	Marble Arch
Maximum # of concurrent calls	5
Current # of concurrent calls	0

FIG. 12

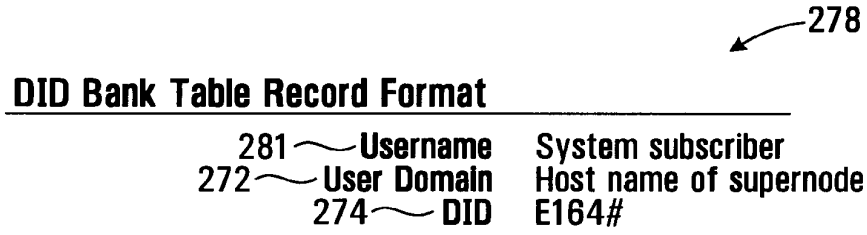

278
DID Bank Table Record Format
281 Username System subscriber
272 User Domain Host name of supernode
274 DID E164#

FIG. 13

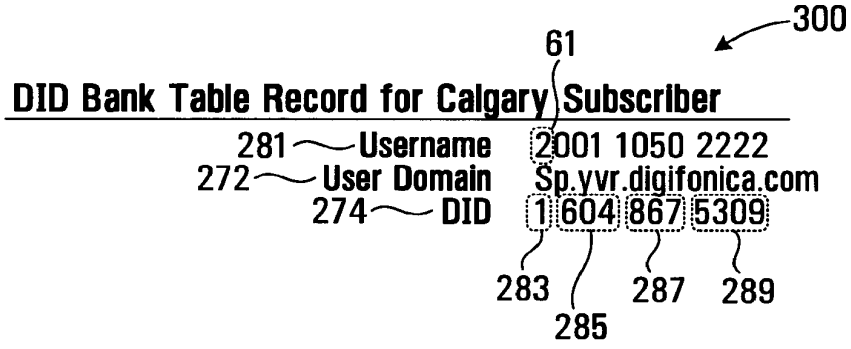

300
61
DID Bank Table Record for Calgary Subscriber
281 Username 2001 1050 2222
272 User Domain Sp.yvr.digifonica.com
274 DID 1 604 867 5309
283
285
287
289

FIG. 14

352

Routing Message Format

354	Supplier Prefix (optional)	Code identifying supplier traffic
356	Delimiter (optional)	Symbol separating fields
358	Callee	PSTN compatible number or Digifonica number
360	Route	Domain name or IP address
362	Time to Live(TTL)	In seconds
364	Other	TBD

FIG. 15

366

Example of Routing Message - Different Node

440110624444@sp.lhr.digifonica.com;ttl=9999

359 361 363

FIG. 16

370

Prefix to Supernode Table Record Format

372	Prefix	First n digits of callee identifier
374	Supernode Address	IP address or fully qualified domain name

FIG. 17

Prefix to Supernode Table Record for Calgary Subscriber

Prefix	20
Supernode Address	sp.yvr.digifonica.com

FIG. 18

META 1002
META V. VOIP-PAL

Master List Record Format

500	ml_id	Alphanumeric
502	Dialing code	Number Sequence
504	Country code	The country code is the national prefix to be used when dialing TO a particular country FROM another country.
506	Nat Sign #(Area Code)	Number Sequence
508	Min Length	Numeric
510	Max Length	Numeric
512	NDD	The NDD prefix is the access code used to make a call **WITHIN** that country from one city to another (when calling another city in the same vicinity, this may not be necessary).
514	IDD	The **IDD prefix** is the international prefix needed to dial a call **FROM** the country listed **TO** another country.
516	Buffer rate	Safe change rate above the highest rate charged by suppliers

FIG. 19

Example: Master List Record with Populated Fields

ml_id	1019
Dialing code	1604
Country code	1
Nat Sign #(Area Code)	604
Min Length	7
Max Length	7
NDD	1
IDD	011
Buffer rate	$0.009/min

FIG. 20

Suppliers List Record Format

540	Sup_id	Name code
542	MI_id	Numeric code
544	Prefix (optional)	String identifying supplier's traffic #
546	Specific Route	IP address
548	NDD/IDD rewrite	
550	Rate	Cost per second to Digifonica to use this route
551	Timeout	Maximum time to wait for a response when requesting this gateway

FIG. 21

Telus Supplier Record

Sup_id	2010 (Telus)
MI_id	1019
Prefix (optional)	4973#
Specific Route	72.64.39.58
NDD/IDD rewrite	011
Rate	$0.02/min
Timeout	20

FIG. 22

Shaw Supplier Record

Sup_id	2011 (Shaw)
MI_id	1019
Prefix (optional)	4974#
Specific Route	73.65.40.59
NDD/IDD rewrite	011
Rate	$0.025/min
Timeout	30

FIG. 23

Sprint Supplier Record

Sup_id	2012 (Sprint)
MI_id	1019
Prefix (optional)	4975#
Specific Route	74.66.41.60
NDD/IDD rewrite	011
Rate	$0.03/min
Timeout	40

FIG. 24

Routing Message Buffer for Gateway Call

4973#0116048675309@72.64.39.58;ttl=3600;to=20 ~570
4974#0116048675309@73.65.40.59;ttl=3600;to=30 ~572
4975#0116048675309@74.66.41.60;ttl=3600;to=40 ~574

FIG. 25

Call Block Table Record Format

604 ~	Username	Digifonica #
606 ~	Block Pattern	PSTN compatible or Digifonica #

FIG. 26

Call Block Table Record for Calgary Callee

604 ~	Username of Callee	2001 1050 2222
606 ~	Block Pattern	2001 1050 8664

FIG. 27

Call Forwarding Table Record Format for Callee

614 ~	Username of Callee	Digifonica #
616 ~	Destination Number	Digifonica #
618 ~	Sequence Number	Integer indicating order to try this

FIG. 28

Call Forwarding Table Record for Calgary Callee

614 ~	Username of Callee	2001 1050 2222
616 ~	Destination Number	2001 1055 2223
618 ~	Sequence Number	1

FIG. 29

Voicemail Table Record Format

624	Username of Callee	Digifonica #
626	Vm Server	domain name
628	Seconds to Voicemail	time to wait before engaging voicemail
630	Enabled	yes/no

FIG. 30

Voicemail Table Record for Calgary Callee

Username of Callee	2001 1050 2222
Vm Server	vm.yvr.digifonica.com
Seconds to Voicemail	20
Enabled	1

FIG. 31

Routing Message Buffer - Same Node

650 200110502222@sp.yvr.digifonica.com;ttl=3600
652 200110552223@sp.yvr.digifonica.com;ttl=3600
654 vm.yvr.digifonica.com;20;ttl=60
656 sp.yvr.digifonica.com

FIG. 32

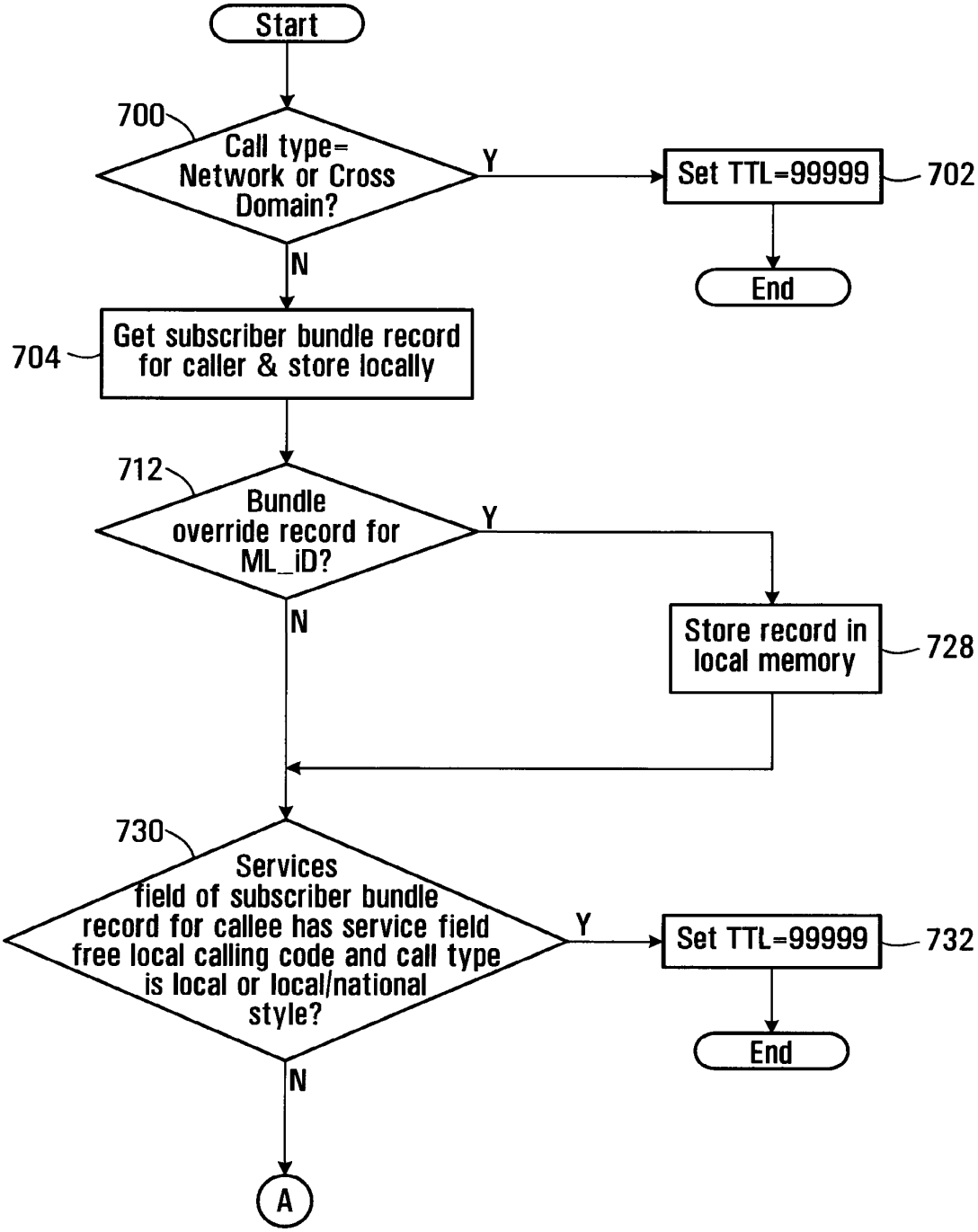

Start
700
Call type= Network or Cross Domain?
Y
Set TTL=99999
702
End
N
704
Get subscriber bundle record for caller & store locally
712
Bundle override record for ML_iD?
Y
Store record in local memory
728
N
730
Services field of subscriber bundle record for callee has service field free local calling code and call type is local or local/national style?
Y
Set TTL=99999
732
End
N
A

FIG. 33A

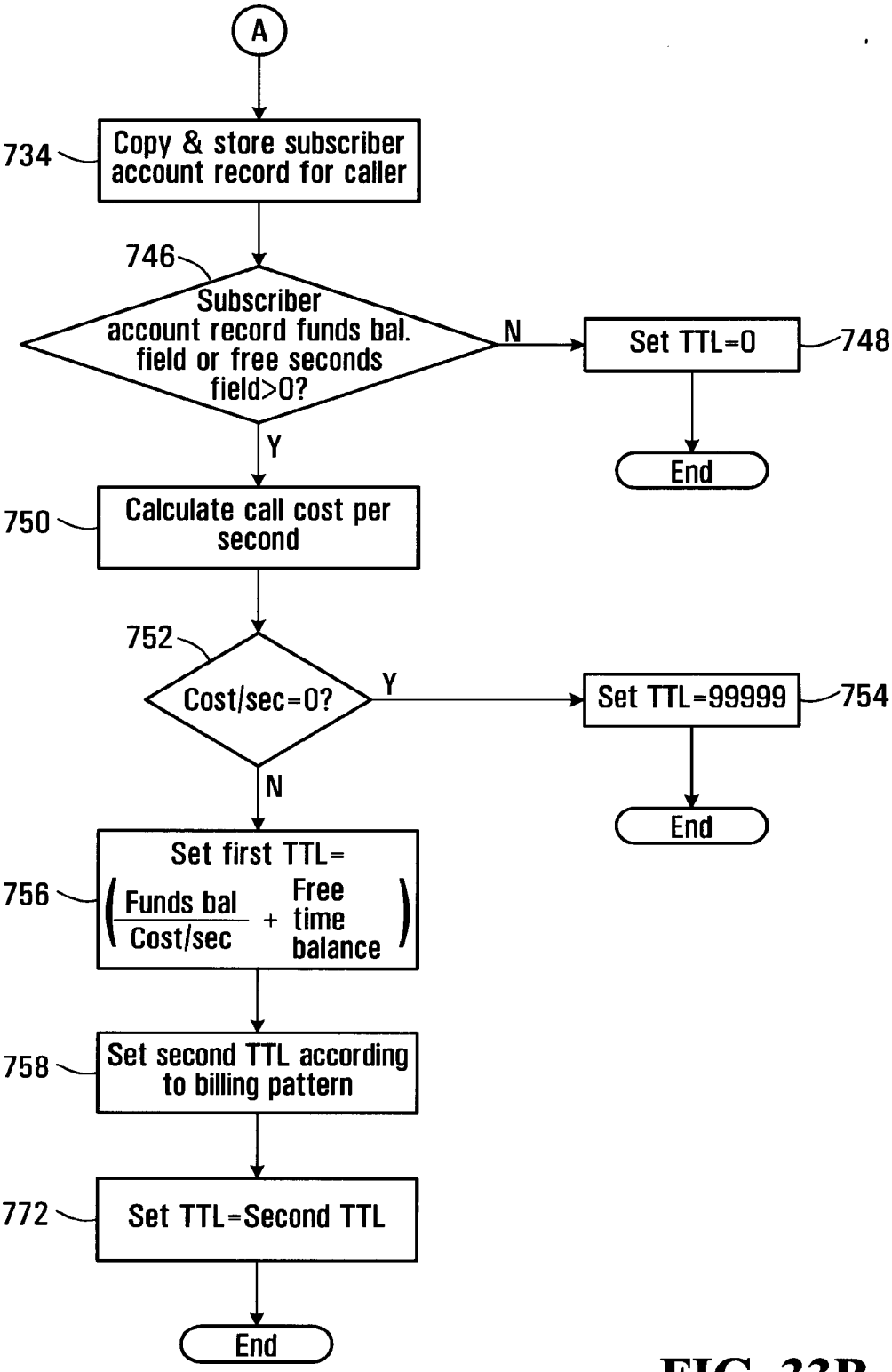

A
734
Copy & store subscriber account record for caller
746
Subscriber account record funds bal. field or free seconds field>0?
N
Set TTL=0
748
End
Y
750
Calculate call cost per second
752
Cost/sec=0?
Y
Set TTL=99999
754
End
N
Set first TTL=
Funds bal / Cost/sec + Free time balance
756
758
Set second TTL according to billing pattern
772
Set TTL=Second TTL
End

FIG. 33B

706

Subscriber Bundle Table Record

708	**Username**	Subscriber username
710	**Services**	Codes identifying service features (e.g. Free local calling; call blocking, voicemail)

FIG. 34

Subscriber Bundle Record for Vancouver Caller

708	**Username**	2001 1050 8667
710	**Services**	10; 14; 16

FIG. 35

714

Bundle Override Table Record

716	**ML_Id**	Master list ID code
718	**Override type**	Fixed; percent; cents
720	**Override value**	real number representing value of override type
722	**Inc1**	first level of charging (minimum # of seconds) charge
724	**Inc2**	second level of charging

FIG. 36

726

Bundle Override Record for Located ML_iD

716	**ML_Id**	1019
718	**Override type**	percent
720	**Override value**	10.0
722	**Inc1**	30 seconds
724	**Inc2**	6 seconds

FIG. 37

736

Subscriber Account Table Record

738	**Username**	Subscriber username
740	**Funds balance**	real number representing $ value of credit
742	**Free time balance**	integer representing # of free seconds

FIG. 38

744

Subscriber Account Record for Vancouver Caller

738	**Username**	2001 1050 8667
740	**Funds balance**	$10.00
742	**Free time balance**	100

FIG. 39

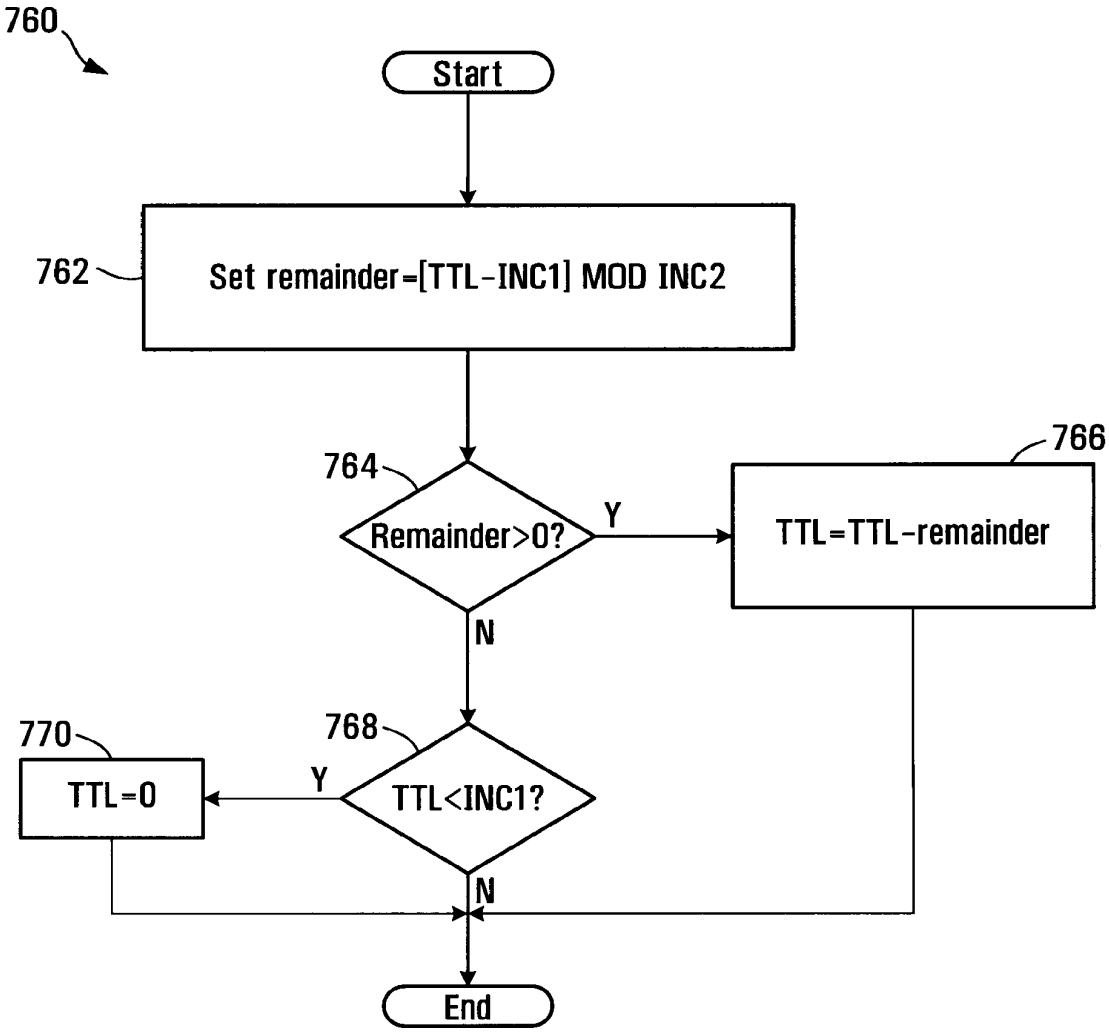

760
Start
762
Set remainder=[TTL-INC1] MOD INC2
764
Remainder>0?
Y
766
TTL=TTL-remainder
N
768
TTL<INC1?
Y
770
TTL=0
N
End

FIG. 40

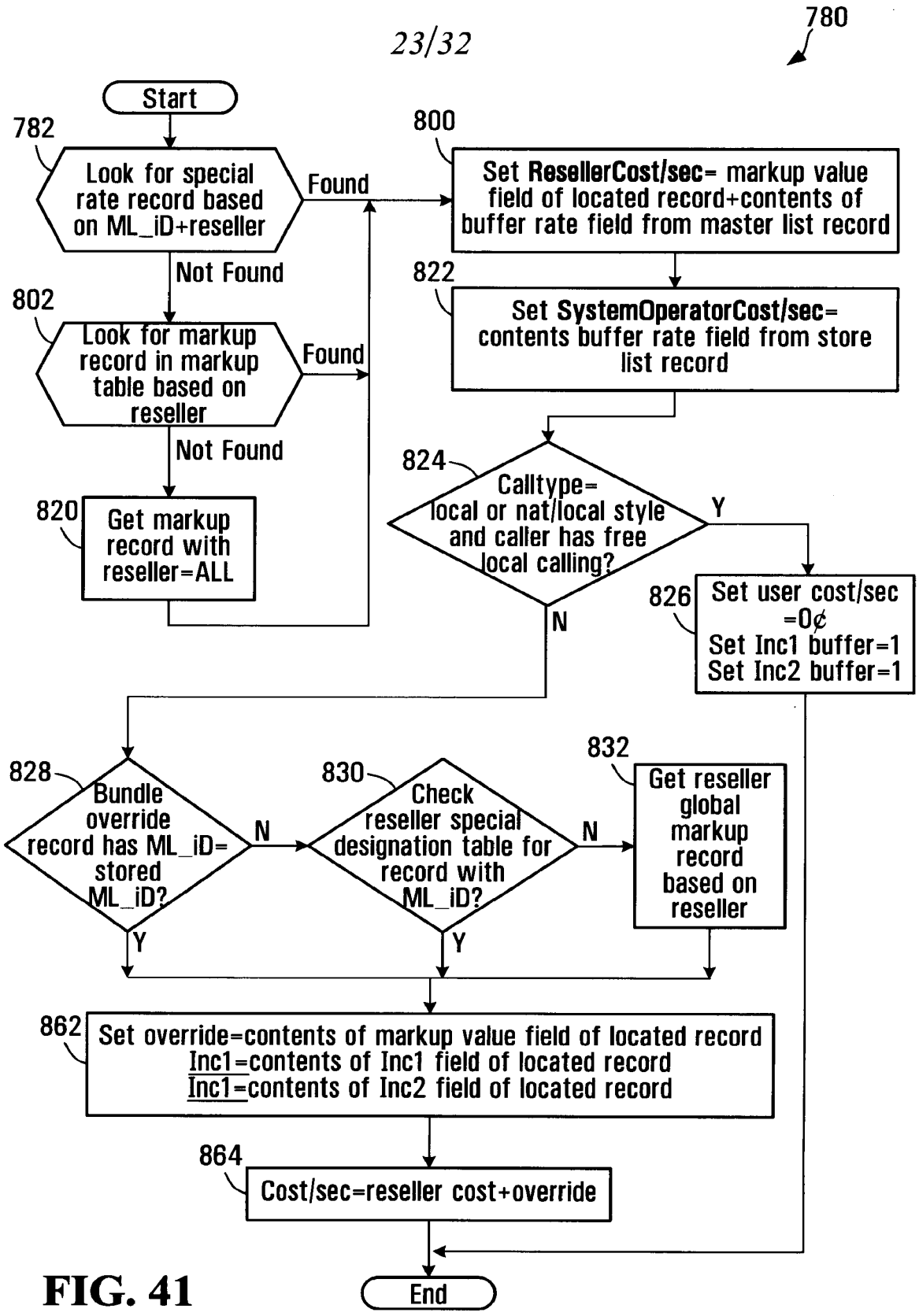

780
Start
782
Look for special rate record based on ML_iD+reseller
Found
Not Found
802
Look for markup record in markup table based on reseller
Found
Not Found
820
Get markup record with reseller=ALL
800
Set ResellerCost/sec= markup value field of located record+contents of buffer rate field from master list record
822
Set SystemOperatorCost/sec= contents buffer rate field from store list record
824
Calltype= local or nat/local style and caller has free local calling?
Y
N
826
Set user cost/sec =0¢
Set Inc1 buffer=1
Set Inc2 buffer=1
828
Bundle override record has ML_iD= stored ML_iD?
N
Y
830
Check reseller special designation table for record with ML_iD?
N
Y
832
Get reseller global markup record based on reseller
862
Set override=contents of markup value field of located record
Inc1=contents of Inc1 field of located record
Inc1=contents of Inc2 field of located record
864
Cost/sec=reseller cost+override
End

FIG. 41

784

System Operator Special Rates Table Record

786	Reseller	retailer id
788	ML_Id	master list id
790	Markup Table	fixed; percent; cents
792	Markup Value	real number representing value of markup type
794	Inc1	first level of charging (minimum # of seconds) charge
796	Inc2	second level of charging

FIG. 42

798

System Operator Special Rates Table Record for Klondike

786	Reseller	Klondike
788	ML_Id	1019
790	Markup Table	cents
792	Markup Value	$0.001
794	Inc1	30
796	Inc2	6

FIG. 43

804

System Operator Markup Table Record

806	Reseller	reseller id code
808	Markup Table	fixed; percent; cents
810	Markup Value	real number representing value of markup type
812	Inc1	first level of charging (minimum # of seconds) charge
814	Inc2	second level of charging

FIG. 44

System Operator Markup Table Record for the Reseller Klondike

806	Reseller	Klondike
808	Markup Table	cents
810	Markup Value	$0.01
812	Inc1	30
814	Inc2	6

FIG. 45

System Operator Markup Table Record

806	Reseller	all
808	Markup Table	percent
810	Markup Value	1.0
812	Inc1	30
814	Inc2	6

FIG. 46

832

Reseller Special Destinations Table Record

834	Reseller	reseller id code
836	ML_id	Master List ID code
838	Markup Table	fixed; percent; cents
840	Markup Value	real number representing value of markup type
842	Inc1	first level of charging (minimum # of seconds) charge
844	Inc2	second level of charging

FIG. 47

846

Reseller Special Destinations Table Record for the Reseller Klondike

834	Reseller	Klondike
836	ML_id	1019
838	Markup Table	percent
840	Markup Value	5%
842	Inc1	30
844	Inc2	6

FIG. 48

848

Reseller Global Markup Table Record

850	Reseller	reseller id code
852	Markup Table	fixed; percent; cents
854	Markup Value	real number representing value of markup type
856	Inc1	first level of charging (minimum # of seconds) charge
858	Inc2	second level of charging

FIG. 49

860

Reseller Global Markup Table Record for the Reseller Klondike

850	Reseller	Klondike
852	Markup Table	percent
854	Markup Value	10%
856	Inc1	30
858	Inc2	6

FIG. 50

900

SIP Bye Message

902	**Caller**	Username
904	**Callee**	PSTN compatible # or Username
906	**Call ID**	unique call identifier (hexadecimal string@IP))

FIG. 51

908

SIP Bye Message

902	**Caller**	2001 1050 8667
904	**Callee**	2001 1050 2222
906	**Call ID**	FA10@192.168.0.20

FIG. 52

META 1002
META V. VOIP-PAL

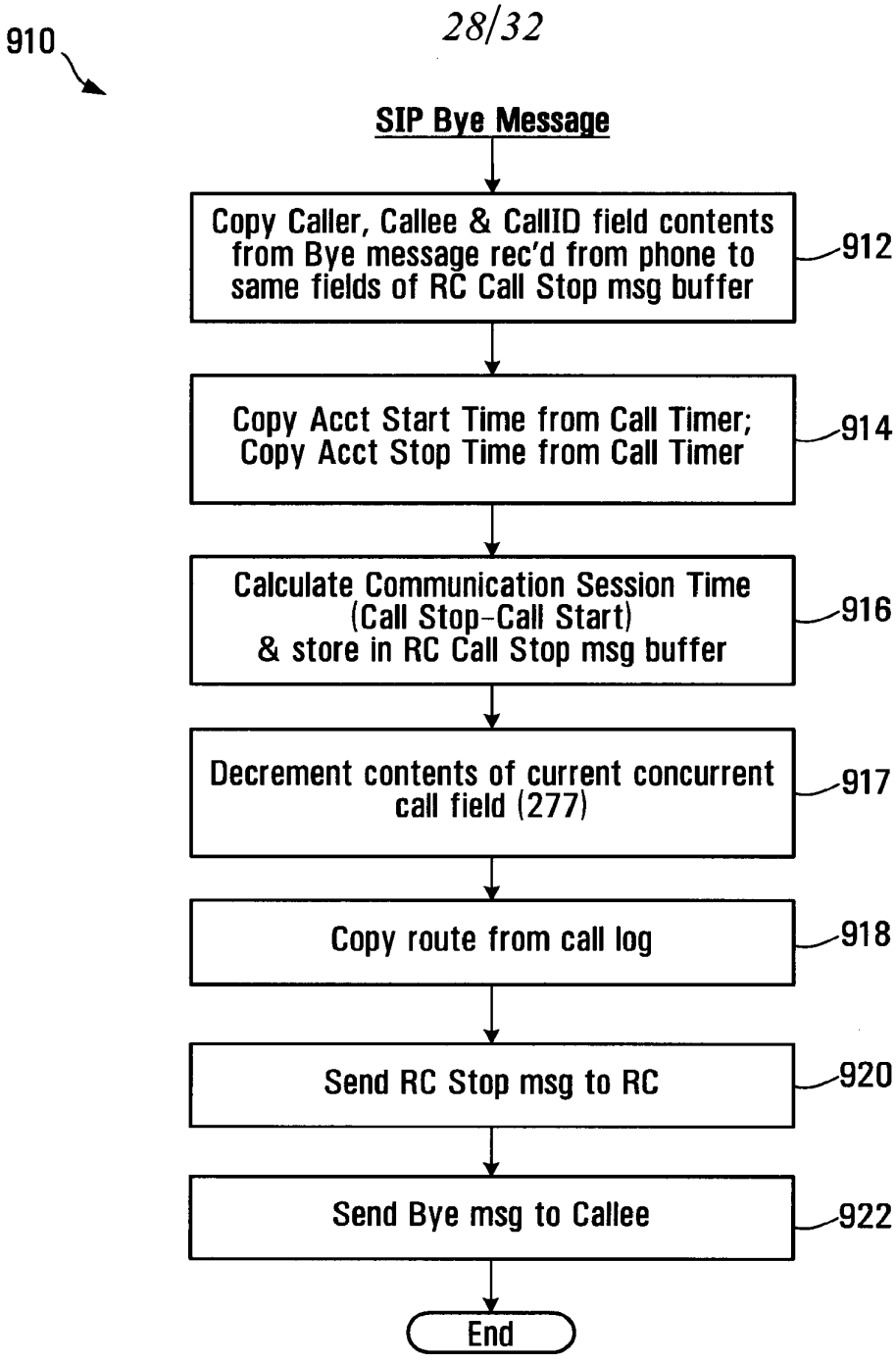


FIG. 53

META 1002
META V. VOIP-PAL

1000

RC Call Stop Message

1002	**Caller**	Username
1004	**Callee**	PSTN compatible # or Username
1006	**Call ID**	unique call identifier (hexadecimal string@IP)
1008	**Acct Start Time**	start time of call
1010	**Acct Stop Time**	time the call ended
1012	**Acct Session Time**	start time-stop time (in seconds)
1014	**Route**	IP address for the communications link that was established

FIG. 54

1020

RC Call Stop Message for Calgary Callee

1002	**Caller**	2001 1050 8667
1004	**Callee**	2001 1050 2222
1006	**Call ID**	FA10@192.168.0.20
1008	**Acct Start Time**	2006-12-30 12:12:12
1010	**Acct Stop Time**	2006-12-30 12:12:14
1012	**Acct Session Time**	2
1014	**Route**	72.64.39.58

FIG. 55

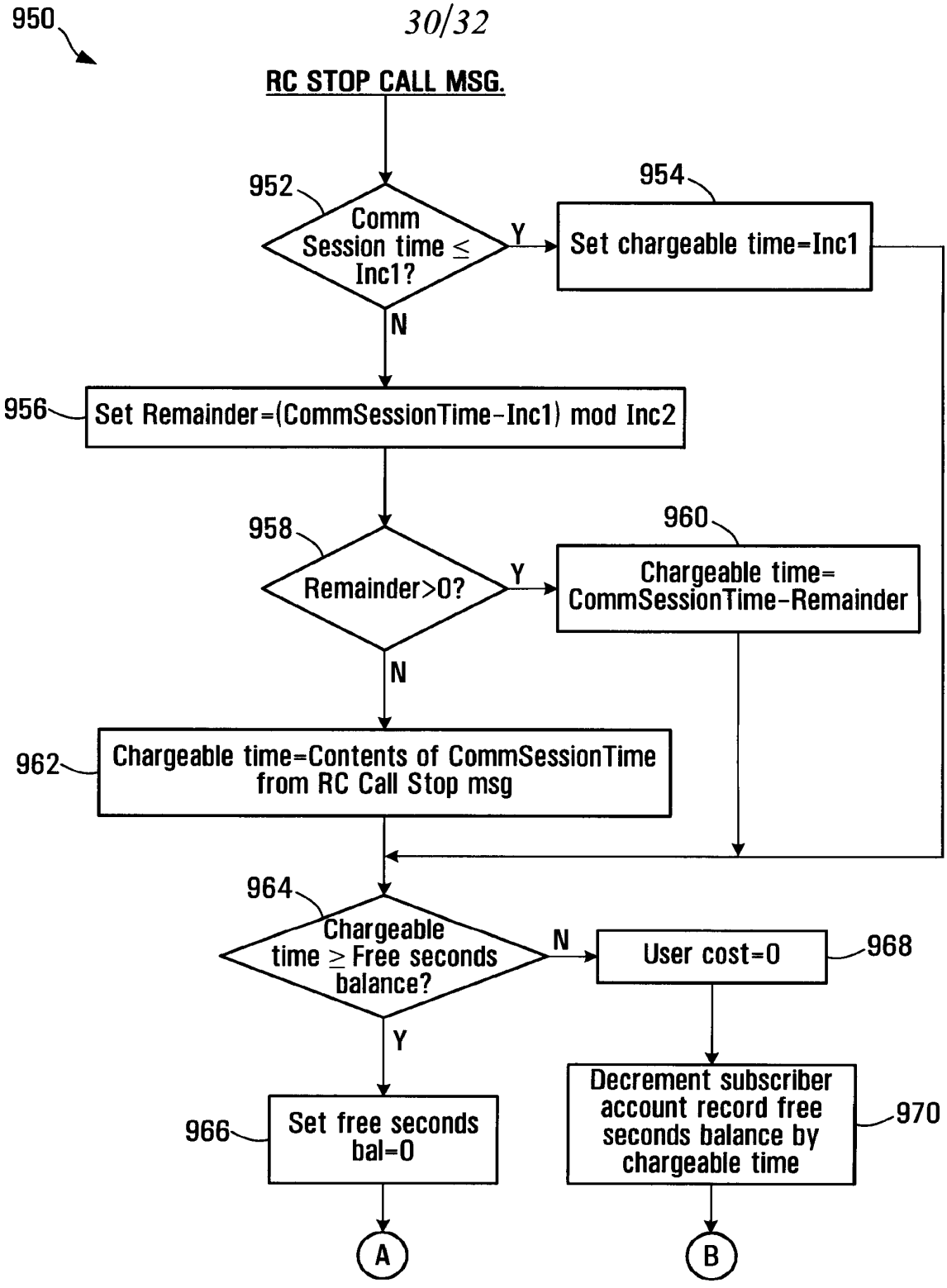

950
RC STOP CALL MSG.
952
Comm Session time ≤ Inc1?
Y
954
Set chargeable time=Inc1
N
956
Set Remainder=(CommSessionTime-Inc1) mod Inc2
958
Remainder>0?
Y
960
Chargeable time= CommSessionTime-Remainder
N
962
Chargeable time=Contents of CommSessionTime from RC Call Stop msg
964
Chargeable time ≥ Free seconds balance?
N
User cost=0
968
Y
966
Set free seconds bal=0
Decrement subscriber account record free seconds balance by chargeable time
970
A
B

FIG. 56A

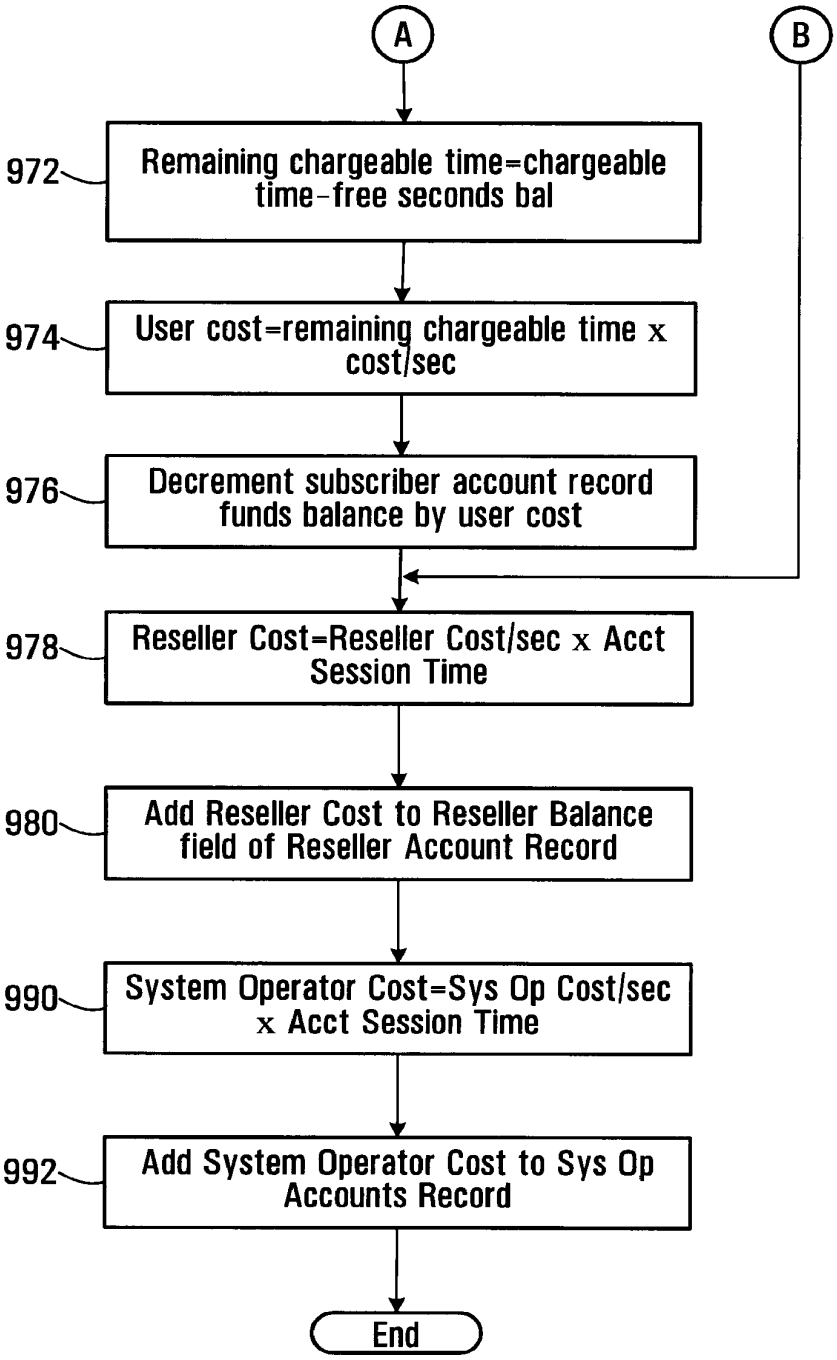

A
B
972
Remaining chargeable time=chargeable time-free seconds bal
974
User cost=remaining chargeable time x cost/sec
976
Decrement subscriber account record funds balance by user cost
978
Reseller Cost=Reseller Cost/sec x Acct Session Time
980
Add Reseller Cost to Reseller Balance field of Reseller Account Record
990
System Operator Cost=Sys Op Cost/sec x Acct Session Time
992
Add System Operator Cost to Sys Op Accounts Record
End

FIG. 56B

982

Reseller Accounts Table Record

984	**Reseller ID**	reseller id code
986	**Reseller balance**	accumulated balance of charges

FIG. 57

988

Reseller Accounts Table Record for Klondike

984	**Reseller ID**	Klondike
986	**Reseller balance**	$100.02

FIG. 58

994

System Operator Accounts Table Record

996	**System Operator balance**	accumulated balance of charges

FIG. 59

System Operator Accounts Record for this System Operator

996	**System Operator balance**	$1000.02

FIG. 60

INTERNATIONAL SEARCH REPORT

International application No.
PCT/CA2007/001956

A. CLASSIFICATION OF SUBJECT MATTER
IPC: ***H04L 12/66*** (2006.01) , ***H04L 12/14*** (2006.01) , ***H04M 11/06*** (2006.01) , ***H04M 15/00*** (2006.01) , ***H04Q 3/64*** (2006.01)
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC: H04L (2006.01), H04M (2006.01), H04Q (2006.01); US classes: 370, 379 in combination with keywords

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic database(s) consulted during the international search (name of database(s) and, where practicable, search terms used)
Canadian Patent Database, USPTO West, Delphion. Keywords: public network, private network, routing message, instant messaging, ip phone, voip, routing controller, sip, gateway, ttl, metric, skype, data structure, routing message, billing, communication session, prepaid

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	CA2249668 C (Bruno et al.) 7 April 1999 (07-04-1999) * Page 9, line 4 to page 14, line 18; Figs 1, 2 *	1-59
A	US7120682 B1 (Salama) 10 October 2006 (10-10-2006) * Col. 1, line 47 to col. 4, line 67 *	1-59
A	US2006/0160565 A1 (Singh et al.) 20 July 2006 (20-07-2006) * Paragraphs 14, 15, 18; Figs 1, 2 *	1-59
A	US2006/0177035 A1 (Cope et al.) 10 August 2006 (10-08-2006) * Paragraphs 5, 6, 12 *	1-59
A, P	US7212522 B1 (Shankar et al.) 1 May 2007 (01-05-2007) * Col. 4, line 47 to col. 5, line 11; Fig. 1 *	1-59

[X] Further documents are listed in the continuation of Box C. [X] See patent family annex.

* Special categories of cited documents :
"A" document defining the general state of the art which is not considered to be of particular relevance
"E" earlier application or patent but published on or after the international filing date
"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed
"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
6 February 2008 (06-02-2008)	20 February 2008 (20-02-2008)
Name and mailing address of the ISA/CA Canadian Intellectual Property Office Place du Portage I, C114 - 1st Floor, Box PCT 50 Victoria Street Gatineau, Quebec K1A 0C9 Facsimile No.: 001-819-953-2476	Authorized officer Arthur Smith 819-953-1360

Form PCT/ISA/210 (second sheet) (April 2007) Page 3 of 5

INTERNATIONAL SEARCH REPORT

International application No.
PCT/CA2007/001956

Box No. II Observations where certain claims were found unsearchable (Continuation of item 2 of the first sheet)

This international search report has not been established in respect of certain claims under Article 17(2)(a) for the following reasons :

1. [ ] Claim Nos. :
because they relate to subject matter not required to be searched by this Authority, namely :

2. [ ] Claim Nos. :
because they relate to parts of the international application that do not comply with the prescribed requirements to such an extent that no meaningful international search can be carried out, specifically :

3. [ ] Claim Nos. :
because they are dependant claims and are not drafted in accordance with the second and third sentences of Rule 6.4(a).

Box No. III Observations where unity of invention is lacking (Continuation of item 3 of first sheet)

This International Searching Authority found multiple inventions in this international application, as follows :

Group I Claims 1-59
Group II Claims 60, 61
Group III Claims 62-84
Group IV Claims 85-107

1. [X] As all required additional search fees were timely paid by the applicant, this international search report covers all searchable claims.

2. [ ] As all searchable claims could be searched without effort justifying additional fees, this Authority did not invite payment of additional fees.

3. [ ] As only some of the required additional search fees were timely paid by the applicant, this international search report covers only those claims for which fees were paid, specifically claim Nos. :

4. [ ] No required additional search fees were timely paid by the applicant. Consequently, this international search report is restricted to the invention first mentioned in the claims; it is covered by claim Nos. :

Remark on Protest
[ ] The additional search fees were accompanied by the applicant's protest and, where applicable, the payment of a protest fee.
[ ] The additional search fees were accompanied by the applicant's protest but the applicable protest fee was not paid within the time limit specified in the invitation.
[X] No protest accompanied the payment of additional search fees.

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US7068772 (Widger et al.) 27 June 2006 (27-06-2006) * Col. 12, line 49 to col. 14, line 44; col. 15, line 26 to col. 16, line 30 Figs. 3, 5 *	60, 61
A	US2006/0209768 A1 (Yan et al.) 21 September 2006 (21-09-2006) * Paras. 71-99, 111-118, 128-141, 179-188; Figs. 3, 4, 7-9 *	60, 61
X	US6058300 (Hanson) 2 May 2000 (02-05-2000) * Col. 2, lines 9-13; col. 5, line 55 to col. 6, line 23; col. 6, line 55 to	62, 63, 73-75
A	col. 7, line 18 *	64-72, 76-84
X	US2005/0177843 A1 (Williams) 11 August 2005 (11-08-2005) * Paragraphs 64 - 69 *	62, 63,73-75
A		64-72, 76-84
A	US6188752 B1 (Lesley) 13 February 2001 (13-02-2001) * Col. 4, line 24 to col. 9, line 6; Figs 1, 3 *	85-107
A	US6507644 B1 (Henderson et al.) 14 January 2003 (14-01-2003) * Col. 1, line 51 to col. 6, line 28 *	85-107
A	US5359642 (Castro) 25 October 1994 (25-10-1994) * Abstract; Col. 5, lines 7-12, col. 6, line 5 to col. 8, line 38; col. 10, line 10 to col. 12, line 68 *	85-107

INTERNATIONAL SEARCH REPORT
Information on patent family members

International application No.
PCT/CA2007/001956

Patent Document Cited in Search Report	Publication Date	Patent Family Member(s)	Publication Date
CA2249668	07-04-1999	EP0915594 A2	12-05-1999
		US6614765 B1	02-09-2003
US7120682	10-10-2006	NONE	
US2006160565	20-07-2006	NONE	
US2006177035	10-08-2006	CA2595429 A1	03-08-2006
		WO2006081115 A1	03-08-2006
US7212522	01-05-2007	US6570869 B1	27-05-2003
		US6658022 B1	02-12-2003
		US6768733 B1	27-07-2004
US7068772	27-06-2006	NONE	
US2006209768	21-09-2006	CA2512959 A1	10-09-2004
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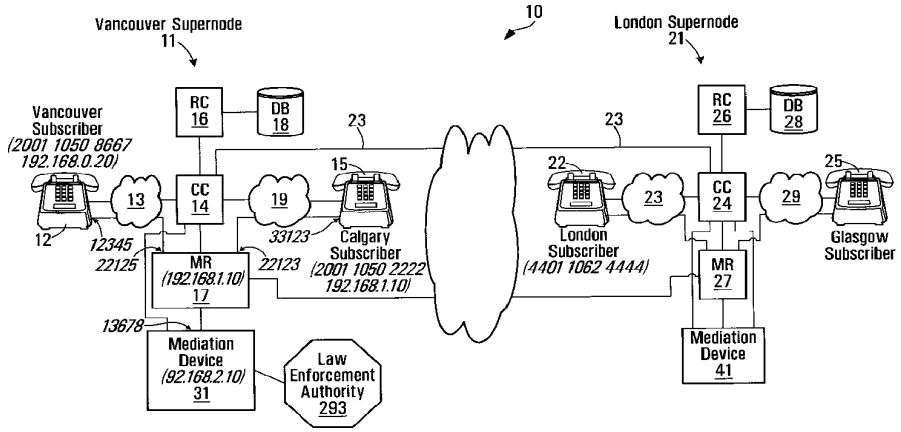
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(57) Abstract: Methods and apparatus for intercepting communications in an Internet Protocol (IP) network involve maintaining dialing profiles for respective subscribers to the IP network, each dialing profile including a username associated with the corresponding subscriber, and associating intercept information with the dialing profile of a subscriber whose communications are to be monitored. Intercept information will include determination information for determining whether to intercept a communication involving the subscriber, and destination information identifying a device to which intercepted communications involving the subscriber are to be sent. When the determination information meets intercept criteria communications are established with a media relay through which communications involving the subscriber will be conducted or are being conducted to cause the media relay to send a copy of the communications involving the subscriber to a mediation device specified by the destination information.

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INTERCEPTING VOICE OVER IP COMMUNICATIONS AND OTHER DATA COMMUNICATIONS

CROSS REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of U.S. Provisional Application No. **60/861,431** filed November 29, 2006.

BACKGROUND OF THE INVENTION

1. Field of Invention

This invention relates to data communications and methods and apparatus for intercepting data communications, particularly voice over IP data communications, in an IP network.

2. Description of Related Art

The term "lawful intercept" is used to describe a procedure which allows law enforcement agencies to perform electronic surveillance of telecommunications. Lawful intercept of telecommunications, particularly phone calls, is premised on a notion that a law enforcement agency has identified a person of interest, obtained a legal authorization for the surveillance (for example, a judicial or administrative warrant), and then contacted the person's telecommunications service provider that will be required to provide the law enforcement agency with a real-time copy of the person's communications. This real-time copy can then be used by the law enforcement agency to monitor or record the person's communications. Within the framework of traditional telecommunications networks, such as, for example, the Public Switched Telephone Network (PSTN) or cellular networks, lawful intercept generally presents a purely economic problem for the service providers that have to ensure that sufficient interception equipment and dedicated links to the law enforcement agencies have been deployed to satisfy lawful intercept requirements mandated by law. However, in the context of Voice over Internet Protocol (VoIP) communications, in addition to the economic problems mentioned above, lawful intercept presents

significant technological challenges which often makes compliance with legally mandated lawful intercept requirements exceedingly difficult.

The problem lies in the very nature of the VoIP technology and the Internet Protocol (IP) networks (for example, the Internet) that underlie it.

Traditional telecommunications networks are "connection-oriented" or "circuit-switched". Communications over such networks occur via dedicated "circuits". Although the networks typically comprise a plurality of available parallel paths, when a circuit is established, only a single one of the available paths is picked. In situations where a circuit has failure protection, a redundant path, also determined at the time of the circuit establishment, can also be reserved. Once the circuit is established, all communications traverse from end to end. Interception of such communications is easy as the service provider can "tap" the circuit at any point in the network that is under its lawful control.

In contrast to circuit-switched networks, IP-based networks are "connectionless" by design. A connectionless IP network essentially comprises a plurality of interconnected network devices (routers) which establish a plurality of paths from any point on the network to any other point. Information that needs to traverse an IP network is divided into small "packets", each one comprising an IP header containing source and destination addressing information, and service flags; and user payload. The specific path that each packet in a communication between parties takes across an IP network is not determined in advance such as in a circuit-switched network. The path is defined on a hop-by-hop basis (router-by-router), each router at which the packet arrives examines the source and destination addresses contained in the IP header and applies a number of service variables such as hop-count (number of routers between the current router and the destination), latency and bandwidth of available links, and administrative considerations such as inter-provider agreements, to determine

the next hop to which the packet will be forwarded. Because the service variables change dynamically, for example in response to a failure of a link in the network, the available paths may change significantly and it is impossible to reliably predict the path or paths that the packets that comprise a specific a specific communication will traverse. Furthermore, it is not even possible to predict the order in which the packets will arrive at their destination as the different paths taken may have different latency. While the plurality of available paths and out-of-order arrivals present no problems to IP-based applications that usually keep track of the packet sequence to reassemble the communication, the same factors present formidable problems for the lawful intercept of communication over IP networks, particularly lawful intercept of VoIP calls.

The problem of lawful intercept in VoIP systems is further exacerbated by the distributed technologies often utilized in such systems. While a VoIP caller typically communicates with a VoIP call controller to facilitate the connection to the VoIP callee, the actual communication between the parties typically occurs by establishing a direct IP connection between them using the User Datagram Protocol (UDP) to encapsulate audio information into IP packets. These packets may take any available path across the IP network as described above. Even if a service provider could place an interception device at every point in the network through which a subscriber's packet could traverse, in order to provide a useful copy of the communication to a law enforcement agency, the service provider would have to reassemble all of the intercepted packets at a single device and only then pass the result to the law enforcement agency. In essence, the service provider would have to mirror the functions of the callee VoIP telephone, except the packets that comprise the communication would have to be collected from multiple points in the network. The technological challenges and economic costs associated with this proposition have thus far resulted in lack of meaningful lawful intercept capabilities in VoIP systems.

SUMMARY OF THE INVENTION

In accordance with one aspect of the invention, there is provided a method for intercepting communications in an Internet Protocol (IP) network. The method involves maintaining dialing profiles for respective subscribers to the IP network, each dialing profile including a username associated with the corresponding subscriber. The method also involves associating intercept information with the dialing profile of a subscriber whose communications are to be monitored, the intercept information including determination information for determining whether to intercept a communication involving the subscriber, and destination information identifying a device to which intercepted communications involving the subscriber are to be sent. The method further involves, when the determination information meets intercept criteria, communicating with a media relay through which the communications involving the subscriber will be conducted or are being conducted to cause the media relay to send a copy of the communications to a mediation device specified by the destination information.

Associating intercept information may involve associating the intercept information with the dialing profile when communications involving the subscriber are not in progress.

Associating intercept information may involve associating the intercept information when communications involving the subscriber are in progress.

Associating the intercept information may involve populating intercept information fields in the dialing profile of the subscriber whose communications are to be monitored.

The method may involve producing a routing message for routing communications involving the subscriber through components of the IP network and determining whether the determination information meets the

intercept criteria prior to producing the routing message and including at least some of the intercept information in the routing message when the determination information meets the intercept criteria.

Determining whether the determination information meets the intercept criteria may involve determining whether a current date and time is within a range specified by the determination information.

The method may involve identifying a media relay through which communications involving the subscriber will be conducted in response to the routing message.

The method may involve pre-associating at least one media relay with the dialing profile of the subscriber whose communications are to be monitored and identifying the media relay may involve identifying the media relay pre-associated with the subscriber whose communications are to be monitored.

Pre-associating may involve populating media relay fields in the dialing profile with an identification of at least one media relay.

The intercept information may be associated with the dialing profile of the subscriber whose communications are to be monitored, in response to receipt of an intercept request message, and the intercept request message may include the intercept information.

The method may involve invoking an intercept request message handler to find a dialing profile associated with the subscriber whose communications are to be monitored, and to perform the step of associating the intercept information with the dialing profile, and to determine whether the intercept criteria are met, and identify a media relay through which the communications are being conducted.

The method may involve maintaining active call records for communications in progress, and the active call records may include a username identifier and a media relay identifier identifying the media relay through which the communications are being conducted and identifying a media relay through which the communications are being conducted may involve locating an active call record associated with communications of the subscriber whose communication are to be monitored to find the media relay associated with the communications.

The method may involve maintaining direct-inward-dialing (DID) records associating PST telephone numbers with usernames of users subscribing to the IP network, and finding a dialing profile associated with the subscriber whose communications are to be monitored may involve finding a username in a DID record bearing a PSTN number associated with the subscriber whose communications are to be monitored. The username may be used to locate a dialing profile associated with the username.

In accordance with another aspect of the invention, there is provided an apparatus for intercepting communications in an Internet Protocol (IP) network. The apparatus includes provisions for maintaining dialing profiles for respective subscribers to the IP network, each dialing profile including a username associated with the corresponding subscriber. The apparatus also includes provisions for associating intercept information with the dialing profile of a subscriber whose communications are to be monitored, the intercept information including determination information for determining whether to intercept a communication involving the subscriber, and destination information identifying a device to which intercepted communications involving the subscriber are to be sent. The apparatus further includes provisions for communicating with a media relay through which the communications involving the subscriber will be conducted or are being conducted to cause the media relay to send a copy of the communications to a mediation device

specified by the destination information, when the determination information meets intercept criteria.

The provisions for associating intercept information may be operably configured to associate the intercept information with the dialing profile when communications involving the subscriber are not in progress.

The provisions for associating intercept information may be operably configured to associate the intercept information when communications involving the subscriber are in progress.

The provisions for associating the intercept information may be operably configured to populate intercept information fields in the dialing profile of the subscriber whose communications are to be monitored.

The apparatus may further include provisions for producing a routing message for routing communications involving the subscriber through components of the IP network and provisions for determining whether the determination information meets the intercept criteria prior to producing the routing message and the provisions for producing the routing message may be operably configured to include at least some of the intercept information in the routing message when the determination information meets the intercept criteria.

The provisions for determining whether the determination information meets the intercept criteria may be operably configured to determine whether a current date and time is within a range specified by the determination information.

The apparatus may further include provisions for identifying a media relay through which communications involving the subscriber will be conducted in response to the routing message.

The apparatus may further include provisions for pre-associating at least one media relay with the dialing profile of the subscriber whose communications are to be monitored and the routing provisions may be operably configured to identify from the dialing profile the media relay pre-associated with the subscriber whose communications are to be monitored.

The provisions for pre-associating may be operably configured to populate media relay fields in the dialing profile with an identification of at least one media relay.

Provisions for associating the intercept information may be operably configured to associate the intercept information associated with the dialing profile of the subscriber whose communications are to be monitored, in response to receipt of an intercept request message, wherein the intercept request message comprises the intercept information.

The apparatus may further include provisions for handling an intercept request message. The provisions for handling an intercept request message may include provisions for finding a dialing profile associated with the subscriber whose communications are to be monitored. The provisions for finding a dialing profile may cooperate with the provisions for associating the intercept information with the dialing profile to cause the intercept information to be associated with the dialing profile. The provisions for handling an intercept request message may include provisions for determining whether the intercept criteria are met and provisions for identifying a media relay through which the communications are being conducted.

The apparatus may further include provisions for maintaining active call records for communications in progress, the active call records including a username identifier and a media relay identifier identifying the media relay through which the communications are being conducted and the provisions for

identifying a media relay through which the communications are being conducted may be operably configured to locate an active call record associated with communications of the subscriber whose communication are to be monitored to find the media relay associated with the communications.

The apparatus may further include provisions for maintaining direct-inward-dialing (DID) records associating PST telephone numbers with usernames of users subscribing to the IP network, and the provisions for finding a dialing profile associated with the subscriber whose communications are to be monitored may be operably configured to find a username in a DID record bearing a PSTN number associated with the subscriber whose communications are to be monitored and use the username to locate a dialing profile associated with the username.

By employing a media replay, all VoIP communications traverse a point in the VoIP system that is under a provider's control and at which the communications can be copied in real-time to a mediation device that passes the intercepted communication to a law enforcement agency.

By maintaining dialing profiles for respective subscribers and associating intercept information of the type described, with the dialing profiles of subscribers whose communications are to be monitored, the dialing profile can serve as the source of determination information for determining whether or not communications involving the subscriber will be monitored and for providing destination information for specifying where the copy of the communications is to be sent. Use of the dialing profile in this manner easily facilitates the dialing profile to be considered a respository for intercept information for a given subscriber and this respository can be addressed whether a call is being initiated or in progress, thereby simplifying control algorithms because they can cooperate with a common source and format of data in the dialing profile.

Other aspects and features of the present invention will become apparent to those ordinarily skilled in the art upon review of the following description of specific embodiments of the invention in conjunction with the accompanying figures.

BRIEF DESCRIPTION OF THE DRAWINGS

In drawings which illustrate embodiments of the invention,

Figure **1** is a block diagram of a system according to a first embodiment of the invention;

Figure **2** is a block diagram of a caller VoIP telephone according to the first embodiment of the invention;

Figure **3** is a schematic representation of a SIP Invite message transmitted between the caller telephone and a call controller (CC) shown in Figure **1**;

Figure **4** is a block diagram of the call controller shown in Figure **1**;

Figure **5** is a flowchart of a process executed by the call controller shown in Figure **1**;

Figure **6** is a schematic representation of a routing controller (RC) request message produced by the call controller shown in Figure **1**;

Figure **7** is a block diagram of a routing controller (RC) processor circuit of the system shown in Figure **1**;

Figures **8A-8D** are flowcharts of a RC Request message handler executed by the RC processor circuit shown in Figure **7**;

Figure **9** is a tabular representation of a dialing profile stored in a database accessible by the RC shown in Figure **1**;

Figure **10** is a tabular representation of a dialing profile for a Vancouver subscriber ;

Figure **11** is a tabular representation of a dialing profile for a Calgary subscriber;

Figure **12** is a tabular representation of a dialing profile for a London subscriber;

Figure **13** is a tabular representation of a direct-inward-dialing (DID) bank table record stored in the database shown in Figure **1**;

Figure **14** is a tabular representation of an exemplary DID bank table record for the London subscriber referenced in Figure **12**;

Figure **15** is a tabular representation of a routing message transmitted from the routing controller to the call controller shown in Figure **1**;

Figure **16** is a tabular representation of a routing message buffer holding a routing message for routing a call to the London callee referenced in Figure **12**;

Figure **16A** is a tabular representation of a routing message buffer holding a message for routing a call to the London callee and to a law enforcement agency for the purpose of lawful intercept;

Figure **17** is a tabular representation of a prefix to supernode table record stored in the database shown in Figure **1**;

Figure **18** is a tabular representation of a prefix to supernode table record that would be used for the Calgary callee referenced in Figure **11**;

Figure **19** is a tabular representation of a master list record stored in a master list table in the database shown in Figure **1**;

Figure **20** is a tabular representation of an exemplary populated master list record;

Figure **21** is a tabular representation of a suppliers list record stored in the database shown in Figure **1**;

Figure **22** is a tabular representation of a specific supplier list record for a first supplier;

Figure **23** is a tabular representation of a specific supplier list record for a second supplier;

Figure **24** is a tabular representation of a specific supplier list record for a third supplier;

Figure **25** is a tabular representation of a routing message, held in a routing message buffer, identifying to the routing controller a plurality of possible suppliers that may carry the call;

Figure **25A** is a tabular representation of a routing message held in a routing message buffer, with lawful intercept fields appended;

Figure **26** is a tabular representation of a call block table record;

Figure **27** is a tabular representation of a call block table record for the Calgary callee;

Figure **28** is a tabular representation of a call forwarding table record;

Figure **29** is a tabular representation of am exemplary call forwarding table record specific for the Calgary callee;

Figure **30** is a tabular representation of a voicemail table record specifying voicemail parameters to enable the caller to leave a voicemail message for the callee;

Figure **31** is a tabular representation of an exemplary voicemail table record for the Calgary callee;

Figure **32** is a tabular representation of an exemplary routing message, held in a routing message buffer, indicating call forwarding numbers and a voicemail server identifier;

Figure **32A** is a tabular representation of an exemplary routing message, held in a routing message buffer, indicating call forwarding numbers and a voicemail server identifier with caller lawful intercept fields appended;

Figure **32B** is a tabular representation of an exemplary routing message, held in a routing message buffer, indicating call forwarding numbers and a voicemail server identifier with caller and callee lawful intercept fields appended;

Figure **33** is a flowchart of a routing message handler process executed by the call controller.

Figure **34** is a schematic representation of messages exchanged during execution of process for establishing audio paths between telephones and a media relay;

Figure **35** is a tabular representation of an active call record maintained by the call controller of Figure **1**;

Figure **36** is a tabular representation of an active call record maintained by the routing controller of Figure **1**;

Figure **37** is a tabular representation of a SIP Invite message transmitted from the call controller to the mediation device;

Figure **38** is a tabular representation of a SIP OK message transmitted from the mediation device to the call controller.

Figure **39** is a tabular representation of a SIP Bye message transmitted from either of the telephones shown in Figure **1** to the call controller;

Figure **40** is a tabular representation of a SIP Bye message sent to the call controller from the Calgary callee;

Figure **41** is a flowchart of a process executed by the call controller for producing a RC stop message in response to receipt of a SIP Bye message;

Figure **42** is a tabular representation of an exemplary RC Call Stop message;

Figure **43** is a tabular representation of an exemplary RC Call Stop message for the Calgary callee;

Figure **44** is a flowchart of a routing controller Law Enforcement Authority request message handler executed by the routing controller shown in Figure **1**;

Figure **45** is a flowchart of a call controller in-call intercept message handler executed by the call controller shown in Figure **1**;

Figure **46** is a flowchart of a routing controller in-call intercept shut down routine executed by the routing controller shown in Figure **1**;

Figure **47** is a flowchart of a call controller cease intercept message handler routing executed by the call controller shown in Figure **1**.

DETAILED DESCRIPTION

Referring to Figure **1**, a system for making voice over IP telephone calls is shown generally at **10**. The system includes a first supernode shown generally at **11** and a second supernode shown generally at **21**. The first supernode **11** is located in a geographical area, such as Vancouver B.C., for example and the second supernode **21** is located in London England, for example. Different supernodes may be located in different geographical regions throughout the world to provide telephone service to subscribers in respective regions. These supernodes may be in communication with each other through high speed / high data throughput links including optical fiber, satellite and/or cable links, for example, forming a system backbone. These supernodes may alternatively or in addition be in communication with each other through conventional Internet services. In the embodiment shown, data communication media for providing for data communications between the first and second supernodes **11** and **21** are shown generally at **23** and may include very high speed data links, for example.

In the embodiment shown, the Vancouver supernode **11** provides telephone service to a geographical region comprising Western Canadian customers

from Vancouver Island to Ontario and includes a Vancouver subscriber and a Calgary subscriber. Another supernode (not shown) may be located in Eastern Canada to provide services to subscribers in that area.

Other, smaller supernodes similar to the type shown may also be employed within the geographical area serviced by a supernode, to provide for call load sharing, for example within a region of the geographical area serviced by the supernode. However, in general, all supernodes are similar and have the properties described below in connection with the Vancouver supernode **11**.

In this embodiment, the Vancouver supernode includes a call controller (CC) **14**, a routing controller (RC) **16**, a database **18**, a media relay **17** and one or more mediation devices (MD), only one of which is shown at **31**. Subscribers such as the Vancouver subscriber and the Calgary subscriber communicate with the Vancouver supernode **11** using their own Internet Service Providers (ISPs) **13** and **19** which route Internet traffic from these subscribers over the Internet. To these subscribers the Vancouver supernode **11** is accessible at a pre-determined IP address or a fully qualified domain name (FQDN) so that it can be accessed in the usual way through a subscriber's ISP. The subscriber in the city of Vancouver uses a telephone **12** that is capable of communicating with the Vancouver supernode **11** using Session Initiation Protocol (SIP) messages and the Calgary subscriber uses a similar telephone **15**, to communicate with the Vancouver supernode from Calgary, AB.

It should be noted that throughout the description of the embodiments of this invention, the IP/UDP addresses of all elements such as the caller and callee telephones, call controller, media relay, and any others, will be assumed to be valid IP/UDP addresses directly accessible via the Internet or a private IP network, for example, depending on the specific implementation of the system. As such, it will be assumed, for example, that the caller and callee telephones will have IP/UDP addresses directly accessible by the call controllers and the media relays on their respective supernodes, and that will

not be obscured by Network Address Translation (NAT) or similar mechanisms. In other words, the IP/UDP information contained in SIP messages (for example the SIP Invite message or the RC Request message which will be described below) will match the IP/UDP addresses of the IP packets carrying these SIP messages.

It will be appreciated that in many situations, the IP addresses assigned to various elements of the system may be in a private IP address space, and thus not directly accessible from other elements. Furthermore, it will also be appreciated that NAT is commonly used to share a "public" IP address between multiple devices, for example between home PCs and IP telephones sharing a single Internet connection. For example, a home PC may be assigned an IP address such as **192.168.0.101** and a Voice over IP telephone may be assigned an IP address of **192.168.0.103**. These addresses are located in so called "non-routable" address space and cannot be accessed directly from the Internet. In order for these devices to communicate with other computers located on the Internet, these IP addresses have to be converted into a "public" IP address, for example **24.10.10.123** assigned to the subscriber by the Internet Service Provider, by a device performing NAT, typically a home router. In addition to translating the IP addresses, the NAT typically also translates UDP port numbers, for example an audio path originating at an IP telephone and using a UDP port **12378** at its private IP address may have been translated to a UDP port **23465** associated with the public IP address of the NAT device. In other words, when a packet originating from the above IP telephone arrives at an Internet-based supernode, the source IP/UDP address contained in the IP packet header will be **24.10.10.1:23465**, whereas the source IP/UDP address information contained in the SIP message inside this IP packet will be **192.168.0.103:12378**. The mismatch in the IP/UDP addresses may cause a problem for SIP-based systems because, for example, a supernode will attempt to send messages to a private address of a telephone – the messages will never get there.

It will be appreciated that a number of methods are available to overcome this problem. For example, the SIP NATHelper open source software module may run on the supernode to correlate public IP/UDP address contained in the headers of the IP packets arriving from SIP devices with private IP/UDP addresses in the SIP messages contained in these packets. Therefore, the embodiments of the invention described below will function whether or not any of the elements of the system are located behind NAT devices that obscure their real IP/UDP addresses.

Referring to Figure **1**, in an attempt to make a call by the Vancouver telephone **12** to the Calgary telephone **15**, for example, the Vancouver telephone sends a SIP Invite message to the Vancouver supernode **11** and in response, the call controller **14** sends an RC Request message to the routing controller **16** which makes various enquiries of the database **18** to produce a routing message which is sent to the call controller **14**. The call controller **14** then causes a communications link including audio paths to be established through the media relay **17** which may include the same Vancouver supernode **11**, a different supernode or a communications supplier gateway, for example, to carry voice traffic to and from the call recipient or callee. Subject to certain conditions being satisfied, as will be described below, when lawful intercept of data is to occur, data on the audio paths is copied to the mediation device **31** which may provide for real time listening of the audio data or recording of same.

Subscriber Telephone

Referring to Figure **2**, in this embodiment, the telephones **12**, **15**, **22** and **25** each includes a processor circuit shown generally at **30** comprising a microprocessor **32**, program memory **34**, an input/output (I/O) interface **36**, parameter memory **38** and temporary memory **40**. The program memory **34**, I/O interface **36**, parameter memory **38** and temporary memory **40** are all in communication with the microprocessor **32**. The I/O interface **36** has a dial

input **42** for receiving a dialed telephone number from a keypad, for example, or from a voice recognition unit or from pre-stored telephone numbers stored in the parameter memory **38**, for example. For simplicity, a box labelled dialing functions **44** represents any device capable of informing the microprocessor **32** of a callee identifier, e.g., a callee telephone number.

The microprocessor **32** stores the callee identifier in a dialed number buffer **41**. In the case of the Vancouver subscriber for example,the dialed number may be **2001 1050 2222**, identifying the Calgary subscriber or the dialed number may be a PSTN number, for example. The I/O interface **36** also has a handset interface **46** for receiving and producing signals from and to a handset **45** that the user may place to his ear. The handset interface **46** may include a BLUETOOTH™ wireless interface, a wired interface or speakerphone, for example. The handset **45** acts as a termination point for an audio path (not shown) which will be appreciated later.

The I/O interface **36** also has a network interface **48** to an IP network which may provide a high speed Internet connection, for example, and is operable to connect the telephone to an ISP. The network interface **48** also acts as a part of the audio path, as will be appreciated later.

The parameter memory **38** has a username field **50**, a password field **52** an IP address field **53** and a SIP proxy address field **54**. The username field **50** is operable to hold a username, which, for the Vancouver subscriber, is **2001 1050 8667**. The username is assigned upon subscription or registration into the system and, in this embodiment includes a twelve digit number having a continent code **61**, a country code **63**, a dealer code **70** and a unique number code **74**. The continent code **61** is comprised of the first or left-most digit of the username in this embodiment. The country code **63** is comprised of the next three digits. The dealer code **70** is comprised of the next four digits and the unique number code **74** is comprised of the last four digits. The password field **52** holds a password of up to **512** characters, in this example. The IP

address field **53** stores an IP address and UDP port number of the telephone **12**, which, for this explanation, is **192.168.0.20:12345**. The SIP proxy address field **54** stores an IP address of a SIP proxy which may be provided to the telephone **12** through the network interface **48** as part of a registration procedure.

The program memory **34** stores blocks of codes for directing the microprocessor **32** to carry out the functions of the telephone, one of which includes a firewall block **56** which provides firewall functions to the telephone, to prevent unauthorized access through the network connection to the microprocessor **32** and memories **34**, **38** and **40**. The program memory **34** also stores call ID codes **57** for establishing a call ID. The call ID codes **57** direct the microprocessor **32** to produce call identifiers having the format of a hexadecimal string and an IP address of the telephone stored in the IP address field **53**. Thus, an exemplary call identifier for a call might be FF**10@192.168.0.20**.

Generally, in response to activating the handset **45** and using the dialing function **44**, the microprocessor **32** produces and sends a SIP Invite message as shown in Figure **3**, to the call controller **14** shown in Figure **1**.

Referring to Figure **3**, the SIP Invite message includes a caller identifier field **60**, a callee identifier field **62**, a digest parameters field **64**, a call identifier field **65**, a caller IP address field **67** and a caller UDP port field **69**. In this embodiment, the caller identifier field **60** includes the username **2001 1050 8667**, which is the username stored in the username field **50** of the parameter memory **38** in the Vancouver telephone **12** shown in Figure **2**. In addition, as an example, referring back to Figure **3**, the callee identifier field **62** includes the username **2001 1050 2222** which is the dialed number of the Calgary subscriber stored in the dialed number buffer **41** shown in Figure **2**. The digest parameters field **64** includes digest parameters and the call identifier field **65** includes a code comprising a generated prefix code (FF**10**) and a

suffix which is the IP address of the telephone **12** stored in the IP address field **53**. The caller IP address field **67** holds the IP address assigned to the telephone, in this embodiment **192.168.0.20**, and the caller UDP port field **69** includes a UDP port identifier identifying a UDP port to which audio data is to be sent for reception by the caller's telephone.

Call Controller

Referring to Figure **4**, a call controller circuit of the call controller **14** (Figure **1**) is shown in greater detail at **100**. The call controller circuit **100** includes a microprocessor **102**, program memory **104** and an I/O interface **106**. The call controller circuit **100** may include a plurality of microprocessors, a plurality of program memories and a plurality of I/O interfaces to be able to handle a large volume of calls. However, for simplicity, the call controller circuit **100** will be described as having only one microprocessor, program memory and I/O interface, it being understood that there may be more.

Generally, the I/O interface **106** includes an input **108** for receiving messages, such as the SIP Invite message shown in Figure **3**, from the telephone shown in Figure **2**. The I/O interface **106** also has an RC Request message output **110** for transmitting an RC Request message to the routing controller **16** of Figure **1**, an RC message input **112** for receiving routing messages from the routing controller **16** (Figure **1**), a media relay (MR) output **114** for transmitting messages to the media relay (Figure **1**) to advise the media relay to establish an audio path, and a MR input **116** for receiving messages from the media relay to which a message has been sent to attempt to establish the audio path. The I/O interface **106** further includes a SIP output **118** for transmitting SIP messages to the telephone **12** (Figure **1**) to advise the telephone of the IP address of the media relay **17** (Figure **1**) which will establish the audio path. The I/O interface **106** further includes mediation device input **119** and output **121** for communicating with the mediation device **31** (Figure **1**).

While certain inputs and outputs have been shown as separate, it will be appreciated that some may be associated with a single IP address and TCP or UDP port. For example, the messages sent and received from the routing controller **16** may be transmitted and received at the same single IP address and TCP or UDP port.

The program memory **104** of the call controller circuit **100** includes blocks of code for directing the microprocessor **102** to carry out various functions of the call controller **14**. For example, these blocks of code include a first block **120** for causing the call controller circuit **100** to execute a SIP Invite-to-RC request process to produce an RC Request message in response to a received SIP Invite message. In addition, there is a Routing Message Handler block **122** which causes the call controller circuit **100** to engage the mediation device and/or execute a call handling routine to establish audio paths through a media relay to establish the call. The program memory **104** further includes an in-call intercept message handler **1450** for intercepting a call in progress and a cease intercept message handler **1520** for ceasing the interception of a call in progress.

Referring to Figure **5**, the SIP Invite-to-RC Request process is shown in more detail at **120**. On receipt of a SIP Invite message of the type shown in Figure **3**, block **132** of Figure **5** directs the call controller circuit **100** of Figure **4** to authenticate the user operating the telephone from which the SIP Invite message originated. This may be done, for example, by prompting the user for a password, by sending a message back to the telephone **12** which is interpreted at the telephone as a request for password entry or the password may automatically be sent to the call controller **14** from the telephone, in response to the message. The call controller **14** may then make enquiries of databases to which it has access, to determine whether or not the user's password matches a password stored in the database. Various functions may be used to pass encryption keys or hash codes back and forth to ensure the secure transmission of passwords.