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wireless communication networks. By way of example, data network 102 may be implemented through a Wide Area Network ("WAN"), Local Area Network ("LAN"), an intranet and/or the Internet. Further, the service center 106 may be connected to multiple data networks 102, such as, for example, to a wireless carrier network and to the Internet.

[026] Voice network 104 may provide telephony services to allow a calling party, such as calling party 120, to place a telephone call to user 110. In one embodiment, voice network 104 may be implemented using a network, such as the Public Switched Telephone Network ("PSTN"). Alternatively, voice network 104 may be implemented on a voice over broadband network, such as a network using voice-over Internet Protocol ("VoIP") technology. Additionally, in other embodiments, the voice network may be a video over broadband network, such as, for example, a network for providing 2-way video communications. In another example, the voice network may be a wireless broadband network, such as, for example, a network using WiFi (i.e., IEEE 802.11(b) and/or (g)). In yet another example, voice network 104 may be a wireless voice network(s), such as, for example, a cellular or third-generation cellular network). In addition, voice network 104 may be implemented using any single or combination of the above-described technologies consistent with the principles of the present invention. Further, service center 106 may be connected to multiple voice networks 104, such as for example, Verizon's™ Voice Network, voice networks operated by other carriers, and wireless carrier networks.

[027] Service center 106 provides a platform for managing communications over data network 102 and voice network 104. Service center 106 also provides gateway functions, such as code and protocol conversions, to transfer communications between data network 102 and voice network 104. Service center 106 may be implemented using a combination of hardware, software, and/or firmware. For example, service center 106 may be implemented using a plurality of general purpose computers or servers coupled by a network (not shown). Although service center 106 is shown

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with direct connections to data network 102 and voice network 104, any number and type of network elements may be interposed between service center 106, data network 102, and voice network 104.

[028] User terminal 112 provides user 110 an interface to data network 102. For example, user terminal 112 may be implemented using any device capable of accessing the Internet, such as a general purpose computer or personal computer equipped with a modem. User terminal 112 may also be implemented in other devices, such as the Blackberry™, and Ergo Audrey™. Furthermore, user terminal 112 may be implemented in wireless devices, such as pagers, mobile phones (with data access functions), and Personal Digital Assistants ("PDAs") with network connections.

[029] User terminal 112 also allows user 110 to communicate with service center 106. For example, user 110 may use Instant Messaging ("IM") to communicate with service center 106. In addition, user terminal 112 may use other aspects of TCP/IP including the Hypertext Transfer Protocol ("HTTP"); the User Datagram Protocol ("UDP"); the File Transfer Protocol ("FTP"); the Hypertext Markup Language ("HTML"); and the eXtensible Markup Language ("XML").

[030] Furthermore, user terminal 112 may communicate directly with service center 106. For example, a client application may be installed on user terminal 112, which directly communicates with service center 106. Also, user terminal 112 may communicate with service center 106 via a proxy.

[031] Phones 114, 116, 118, and 122 interface with voice network 104. Phones 114, 116, 118, and 122 may be implemented using known devices, including wireline phones and mobile phones. Although phones 114, 116, 118, and 122 are shown directly connected to voice network 104, any number of intervening elements, such as a Private Branch Exchange ("PBX"), may be interposed between phones 114, 116, 118, and 122 and voice network 104.

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[032] Fig. 2 is a block diagram of a user terminal consistent with the present invention. User terminal 112 may include a Central Processing Unit ("CPU") 200, a memory 202, a storage module 204, a network interface 206, an input interface 208, an output interface 210, an input device 216, and an output device 218.

[033] CPU 200 provides control and processing functions for user terminal 112. Although Fig. 2 illustrates a single CPU, user terminal 112 may include multiple CPUs. CPU 200 may also include, for example, one or more of the following: a co-processor, memory, registers, and other processing devices and systems as appropriate. CPU 200 may be implemented, for example, using a Pentium™ processor provided from Intel Corporation™.

[034] Memory 202 provides a primary memory for CPU 200, such as for program code. Memory 202 may be embodied with a variety of components of subsystems, including a Random Access Memory ("RAM") and a Read-Only Memory ("ROM"). When user terminal 112 executes an application installed in storage module 204, CPU 200 may download at least a portion of the program code from storage module 204 into memory 202. As CPU 200 executes the program code, CPU 200 may also retrieve additional portions of program code from storage module 204.

[035] Storage module 204 may provide mass storage for user terminal 112. Storage module 204 may be implemented with a variety of components or subsystems including, for example, a hard drive, an optical drive, CD ROM drive, DVD drive, a general-purpose storage device, a removable storage device, and/or other devices capable of storing information. Further, although storage module 204 is shown within user terminal 112, storage module 204 may be implemented external to user terminal 112.

[036] Storage module 204 includes program code and information for user terminal 112 to communicate with service center 106. Storage module 204 may include, for example, data structures or program code for a calendar application, such as GroupWise™ provided by Novell Corporation™ or Outlook provided by Microsoft Corporation™; a client application, such as a

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Microsoft Network Messenger Service ("MSNMS") client or America Online Instant Messenger (AIM) client; and an Operating System ("OS"), such as the Windows Operation System provided by Microsoft Corporation. In addition, storage module 204 may include other program code and information, such as program code for TCP/IP communications; kernel and device drivers; configuration information, such as a Dynamic Host Configuration Protocol ("DHCP") configuration; a web browser, such as Internet Explorer™ provided by Microsoft Corporation, or Netscape Communicator™ provided by Netscape Corporation™; and any other software that may be installed on user terminal 112.

[037] Network interface 206 provides a communications interface between user terminal 112 and data network 102. Network interface 206 may receive and transmit communications for user terminal 112. For example, network interface 206 may be a modem, or a LAN port.

[038] Input interface 208 receives input from user 110 via input device 212 and provides the input to CPU 200. Input device 212 may include, for example, a keyboard, a microphone, and a mouse. Other types of input devices may also be implemented consistent with the principles of the present invention.

[039] Output interface 210 provides information to user 110 via output device 214. Output device 214 may include, for example, a display, a printer, and a speaker. Other types of output devices may also be implemented consistent with the principles of the present invention.

[040] Fig. 3 is a diagram of a voice network, consistent with the principles of the present invention. As shown, voice network 104 includes an Intelligent Service Control Point ("ISCP") 302, Service Transfer Points ("STP") 304 and 306, service switching points ("SSP") 308 and 310, a Line Information Database ("LIDB") 312, an ISCP Service Provisioning And Creation Environment ("SPACE") 314, a Recent Change Environment ("RCE") 316, an Intelligent Peripheral ("IP") 320, and a switch access 322. Although this embodiment of a voice network 104 is described as a PSTN, as

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discussed above in other embodiments, voice network 104 may be, for example, a voice or video over broadband network a wireless broadband, a wireless voice network, etc.

[041] Voice network 104 may be implemented using the PSTN and SS7 as a signaling protocol. The SS7 protocol allows voice network 104 to provide features, such as call forwarding, caller-ID, three-way calling, wireless services such as roaming and mobile subscriber authentication, local number portability, and toll-free/toll services. The SS7 protocol provides various types of messages to support the features of voice network 104. For example, these SS7 messages may include Transaction Capabilities Applications Part ("TCAP") messages to support event "triggers," and queries and responses between ISCP 302 and SSPs 308 and 310.

[042] ISCP 302 may also be, for example, a standard service control point ("SCP") or an Advanced Intelligent Network ("AIN") SCP. ISCP 302 provides translation and routing services of SS7 messages to support the features of voice network 104, such as call forwarding. In addition, ISCP 302 may exchange information with the service center 106 using TCP/IP or SS7. ISCP 302 may include service logic used to provide a switch, such as SSP 308 or 310, with specific call processing instructions. ISCP 302 may also store data related to various features that a user may activate. Such features may include, for example, call intercept and voice mail. ISCP 302 may be implemented using a combination of known hardware and software. ISCP 302 is shown with a direct connection to service center 106 and a connection to ISCP SPACE 314, however, any number of network elements including routers, switches, hubs, etc., may be used to connect ISCP 302, ISCP SPACE 314, and service center 106. Further, information exchanged between the ISCP 302 and service center 106 may use, for example, the SR-3389 General Data Interface ("GDI") for TCP/IP.

[043] STPs 304 and 306 relay SS7 messages within voice network 104. For example, STP 304 may route SS7 messages between SSPs 308

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and 310. STP 302 may be implemented using known hardware and software from manufacturers such as NORTEL™ and LUCENT Technologies™.

[044] SSPs 308 and 310 provide an interface between voice network 104 and phones 114 and 120, respectively, to setup, manage, and release telephone calls within voice network 104. SSPs 308 and 310 may be implemented as a voice switch, an SS7 switch, or a computer connected to a switch. SSPs 308 and 310 exchange SS7 signal units to support a telephone call between calling party 120 and user 110. For example, SSPs 308 and 310 may exchange SS7 messages, such as TCAP messages, within Message Signal Units ("MSU") to control calls, perform database queries to configuration database 312, and provide maintenance information.

[045] LIDB 312 comprises one or more known databases to support the features of voice network 104. For example, LIDB 312 may include subscriber (i.e., a user that is a customer of a business entity providing services through one or more components of environment 100) information, such as a service profile, name and address, and credit card validation information. Although, in this figure, LIDB 312 is illustrated as directly connected to ISCP 302, LIDB 312 may be connected to ISCP 302 through an STP (e.g., 304 and 306). Additionally, this communication link may use, for example, the GR-2838 General Dynamic Interface ("GDI") for SS7.

[046] ISCP SPACE 314 may be included as part of the ISCP 302 or be separate from the ISCP 302. For example, the Telcordia™ ISCP may include an environment similar to SPACE 314 as part of the product. Further, ISCP SPACE 314 may include one or more servers. ISCP SPACE 314 is the point in the ISCP platform where customer record updates may be made.

[047] In one embodiment, customer records may be stored in ISCP SPACE 314 such that the records may be updated and sent to ISCP 302. These records may include information regarding how to handle calls directed to the customer. For example, these customer records may include information regarding whether or not calls for the customer are to be forwarded to a different number, and/or whether or not the call should be

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directed to an IP, such as a voice mail system, after a certain number of rings. Additionally, one ISCP SPACE 314 may provide updates to one or more ISCPs 302 via an ISCP network (not shown).

[048] Additionally, voice network 104 may include one or more recent change engines 316 such as, for example, an Enterprise Recent Change engine ("eRC"); an Assignment, Activation, and Inventory System ("AAIS"); or a Multi-Services Platform ("MSP"). As an example, the eRC and AAIS may be used in voice networks 104 located in the western part of the United States, while an MSP may be used in networks in the eastern part. The recent change engines may be used to update switch and ISCP databases. For example, a recent change engine may deliver database updates to SSPs and to ISCPs, such that when updating databases, these recent change engines emulate human operators. Additionally, if the instructions are to be sent to an ISCP 302, the recent change engine may first send the instructions to ISCP SPACE 314, which then propagates the instructions to ISCP 302 as discussed above. Further, an MSP or eRC may be used, for example, for providing updates to both SSPs 308 or 310 and ISCPs 302. Or, for example, an eRC may be used for providing updates to SSPs 308 or 310, while an AAIS is used for providing updates to ISCPs 302.

[049] Updates sent to SSPs 308 or 310 may be sent from the recent change engine 316 via a switch access 322 that may, for example, convert the updates into the appropriate protocol for SSP 308 or 310. For example, recent change engine 316 may send updates to SSPs 308 or 310 via TCP/IP. Switch access 322 may then convert the updates from TCP/IP to X.25. This switch access 322 may be implemented using hardware and/or software. These connections may include any number of elements, such as, for example, switches, routers, hubs, etc. and may be, for example, an internal data network for voice network 104.

[050] Voice network 104 may also include one or more IPs. For example, in Figure 3, an IP 320 is illustrated as being connected to SSP 308. These IPs may be used for providing functions for interaction between users

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and the voice network, such as voice mail services, digit collection, customized announcements, voice recognition, etc. Moreover, the communications between SSP 308 and IP 320 may use the Primary Rate interface ("PRI") (e.g., the 1129 protocol) protocol. Additionally, the IP 320 may be capable of sending and receiving information to/from the Service Center 106. These communications may use, for example, the SR-3511 protocol. Further, although Figure 3 illustrates this connection as a direct connection, this connection may include any number of elements including routers, switches, hubs, etc., and may be via, for example, an internal data network for voice network 104.

[051] Fig. 4 is a block diagram of a service center, consistent with the principles of the present invention. As shown, service center 106 may include firewalls 402 and 404, one or more digital companion servers 406, one or more communication portal servers 408, one or more network access servers 410, and a voice portal 412. The voice portal 412 may include a voice portal application server 414 and a voice recognition server 416. A network 418 may be used to interconnect the firewalls and servers. Additionally, back end server(s) 420 may be provided between the service center 106 and the voice network 104.

[052] Firewalls 402 and 404 provide security services for communications between service center 106, data network 102, and voice network 104, respectively. For example, firewalls 402 and 404 may restrict communications between user terminal 112 and one or more servers within service center 106. Any appropriate security policy may be implemented in firewalls 402 and 404 consistent with the principles of the present invention. Firewalls 402 and 404 may be implemented using a combination of known hardware and software, such as the Raptor Firewall provided by the Axent Corporation. Further, firewalls 402 and 404 may be implemented as separate machines within service center 106, or implemented on one or more machines external to service center 106.

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[053] Network 418 may be any appropriate type of network, such as an Ethernet or FDDI network. Additionally, network 418 may also include switches and routers as appropriate without departing from the scope of the invention. Further, additional firewalls may be present in network 418, for example, to place one or more of servers 406, 408, 410, or voice portal 412 behind additional firewalls.

[054] Each server (406, 408, 410, 414, 416, 420) may be any appropriate type of server or computer, such as a Unix or DOS-based server or computer. The servers may implement various logical functions, such as those described below. In Figure 4, a different server is illustrated as being used for each logical function. In other embodiments, the logical functions may be split across multiple servers, multiple servers may be used to implement a single function, all functions may be performed by a single server, etc.

[055] In general, a digital companion server 406 may provide the software and hardware for providing specific services of the service center. Exemplary services include, for example, permitting a customer to add contacts to their address book from a history of calls made or received by the customer, permitting a customer to make calls directly from their address book, scheduling a call to be placed at a specific time, or permitting the customer to look at the name and/or address associated with a phone number. Additionally, these services may include permitting the customer to listen to their voice mail on-line, forwarding their calls based on a scheduler and/or the calling parties number, setting up conference calls on-line, real-time call management, etc. In one embodiment, real-time call management enables a user to perform several functions as a call is being received, such as sending a call to voice mail, sending a call received on one device to another device, manually initiating protection from telemarketers, playing an announcement for the caller, scheduling a call back, bridging a caller onto a current call, etc.

[056] A communication portal server 408 may provide the hardware and software for managing a customer's account and interfacing with customer account information stored by the provider of customer's voice network 104. The network access servers 410 may provide the hardware and software for sending and receiving information to the voice network 104 in processing the applications provided by the service center. For example, the network access servers 410 may be used for transmitting and/or receiving information from/to an ISCP 302 or an SSP 308 or 310 of the voice network 104.

[057] Voice portal 412 includes software and hardware for receiving and processing instructions from a customer via voice. For example, a customer may dial a specific number for voice portal 412. Then the customer using speech may instruct the service center 105 to modify the services to which the customer subscribes. Voice portal 412 may include, for example, a voice recognition function 416 and an application function 414. Voice recognition function 416 may receive and interpret dictation, or recognize spoken commands. Application function 414 may take, for example, the output from voice recognition function 416, convert it to a format suitable for service center 106 and forward the information to one or more servers (406, 408, 410) in service center 106.

[058] Fig. 5 illustrates a logical architecture of an exemplary system, consistent with the present invention. As illustrated, the logical architecture may be split into four planes: client side plane 502, application service plane 504, network access plane 506, and voice network plane 508.

[059] Client side plane 502 includes user terminals 112_A and 112_B that a user may use to send and/or receive information to/from the service center 106. Additionally, client side plane 502 includes the user's phone(s) 114. As discussed above, user terminals 112 may be any type of device a user may use for communicating with service center 106. For example, user terminal 112_A may be a PDA running a program for communicating with service center 106, while user terminal 112_B may be a desktop type

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computer running a web browser for communicating with the service center 106 via the Internet. Additionally, the user may have one or more phones 114, such as, for example, one or more standard landline telephones and/or wireless phones.

[060] Application service plane 504 includes digital companion server(s) 406, communication portal server(s) 408, and voice portal 412. These entities may communicate between one another using, for example, web services or any other suitable protocols. Web services are a standardized way of integrating Web-based applications using XML, Simple Object Access Protocol ("SOAP"), Web Services Description Language ("WSDL") and Universal Description, Discovery and Integration ("UDDI") open standards over an Internet protocol backbone.

[061] As illustrated, a digital companion server 406 may provide the following functions: a client proxy function 512, a web server function 514, an application server function 516, a calendar server function 518, a notification server function 520, and a database function 522. Each of these functions may be performed in hardware, software, and/or firmware. Further, these functions may each be executed by a separate server, split across multiple servers, included on the same server functions, or any other manner.

[062] Client proxy function 512 provides a proxy function for the digital companion that may be used for security purposes. Client proxy function 512 may be included in a separate server such that all communications sent from the other digital companion functions/servers to a user terminal 112 via data network 102 go through client proxy 512. Also, if client proxy 512 is included on a separate server, for example, an additional firewall may be provided between client proxy 512 and the other digital companion servers to provide additional security.

[063] Web server function 514 provides functionality for receiving traffic over data network 102 from a customer. For example, web server function 514 may be a standard web server that a customer may access

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using a web browser program, such as Internet Explorer or Netscape Communicator.

[064] Application server function 516 encompasses the general functions performed by digital companion server(s) 406. For example, these functions may include interfacing with the various other digital companion functions to perform specific services provided by the service center. These services may include, for example, interfacing with other function(s), software, and/or hardware to provide a customer with the capability of managing their calls online. Such services may include, for example, permitting a customer to add contacts to their address book from a history of calls made or received by the customer, permitting a customer to make calls directly from their address book, scheduling a call to be placed at a specific time, or permitting the customer to look at the name and/or address associated with a phone number. Additionally, these services may include permitting the customer to listen to their voice mail on-line, forwarding their calls based on a scheduler and/or the calling parties number, setting up conference calls on-line, enabling call management with user intervention in real-time, etc.

[065] Additionally, application server function 516 may interface with one or more external devices, such as an external web server, for retrieving or sending information. For example, application server function 516 may interface with a voice network's data center 556 (e.g., verizon.com) to determine the services to which the customer subscribes (e.g., call waiting, call forwarding, voice mail, etc.).

[066] Calendar server function 518 may provide the capability of scheduling events, logging when certain events occurred, triggering the application-functions to perform a function at a particular time, etc. In one aspect of the invention, calendar server function 518 generates and maintains a scheduling data structure, such as a user calendar that includes scheduling events (e.g., meetings, tasks, etc.), that are created by the user through user terminal 112_A. For example, a user may schedule a conference call event in a calendar application reflecting a time and date when the user is to

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participate in a conference call that is be established in accordance with certain aspects related to the present invention. Calendar server function 518 may operate with, or leverage, application server function 516 to initiate conference call configuration processes consistent with aspects of the invention.

[067] Notification server function 520 provides the capability to send information from the service center 106 to a user terminal 112. For example, the notification server function 520 at the direction of the application server function 516 may send a notification to the user terminal 112 that the user is presently receiving a phone call at the user's phone 114. This notification may be, for example, an instant message pop-up window that provides an identification of the caller as well as the number being called. The notification may also have a number of user-selectable buttons or items associated with it that enable the user to manage a call in real-time.

[068] Database function 522 provides the storage of information useable by the various applications executed by the digital companion servers. These databases may be included in, for example, one or more external storage devices connected to the digital companion servers. Alternatively, the databases may be included in storage devices within the digital companion servers themselves. The storage devices providing database function 522 may be any type of storage device, such as for example, CD-ROMs, DVD's, disk drives, magnetic tape, etc.

[069] As discussed above, communication portal server(s) 408 provide the hardware and software for managing a customer's account and interfacing with customer account information stored by the provider of customer's voice network 104. As illustrated in Figure 5, a communication portal server 408 may provide the following functions: a web server function 526, an application server function 528, a contacts database function 530, and/or a customer profile function 532. Each of these functions may be performed by a separate server, split across multiple servers, included on the same server functions, or any other manner.

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[070] Web server function 526, as with web server function 514 of the digital companion servers, provides functionality for receiving traffic over data network 102 from a customer. For example, the web server may be a standard web server that a customer may access using a web browser, such as Internet Explorer or Netscape Communicator.

[071] Application server function 528 encompasses the general functions performed by communication portal servers 408. For example, these functions may include interfacing with the voice network to retrieve and/or modify customer profile information, and creating and editing an address book for the user. Additionally, application server function 528 may include the functionality of sending and/or receiving information to/from external servers and/or devices. For example, communication portal servers 408 may be connected to a network, such as, the Internet. Application server function 528 may then provide connectivity over the Internet to external servers 552 that provide web services, such as the Superpages web page. Application server function 528 could then contact these external services 552 to retrieve information, such as an address for a person in the user's address book.

[072] In another example, application server function 528 of communication portal 408 may interface a Single Sign On ("SSO") server 554. SSO 554 may be used to allow users to access all services to which the user subscribes, on the basis of a single authentication that is performed when they initially access the network.

[073] Moreover, application server function 528, similar to application server 516, may provide functionality to facilitate services performed by the service center. These services may include, for example, interfacing with other function(s), software, and/or hardware to provide a customer with the capability of managing their calls online. For example, permitting a customer to add contacts to their address book from a history of calls made or received by the customer, permitting a customer to make calls directly from their address book, scheduling a call to be placed at a specific time, or permitting

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the customer to look at the name and/or address associated with a phone number. Additionally, these services may include permitting the customer to listen to their voice mail on-line, forwarding their calls based on a scheduler and/or the calling parties number, setting up conference calls on-line, enabling call management with user intervention in real-time, etc.

[074] Contacts database function 530 includes storage devices for storing an address book for the user. This address book may be any appropriate type of address book. For example, the user's address book may include the names, phone numbers, and addresses of people and/or organizations. These storage devices may be internal or external to communication portal servers 406 or some combination in between. In addition, these storage devices may be any type of storage device, such as magnetic storage, memory storage, etc.

[075] Customer profile database function 532 includes storage devices for storing customer profile information for the user. These storage devices may be the same or separate storage devices used for the contacts database. The customer profile may include information regarding the user's account for their voice network. For example, this information may include the user's name, billing address, and other account information. Additionally, the customer profile may include information regarding voice services to which the user subscribes, such as, for example, call waiting, voice mail, etc.

[076] Application services plane 504 of the architecture may also include a voice portal 412. As discussed above, the voice portal 412 may include, for example, a voice recognition function 416 and an application server function 414, and be used for receiving and processing instructions from a customer via voice. The voice recognition function may be implemented using hardware and/or software capable of providing voice recognition capabilities. This hardware and/or software may be a commercially available product, such as the Voice Application platform available from Tellme Networks, Incorporated. Application server function 414 of voice portal 412 may include hardware and/or software for exchanging

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information between digital companion servers 406 and voice recognition function 416. Additionally, application server function 414 may be included on a separate server, included in the hardware and software providing voice recognition function 416, included in digital companion servers 406, etc.

[077] Network access plane 506 of the architecture includes the functions for providing connectivity between application service plane 502 and voice network 104. For example, this plane may include recent change engines 316, network access servers 410, and/or back end servers 420.

[078] As discussed above, recent change engines 316 may be used to update switches and ISCP databases included in the voice network 104. In one embodiment, recent change engines 316 may include an AAIS 544, an eRC 546, and/or an MSP 548. Additionally, a proxy 542 may be used between the digital companion servers 406 and recent change engines 542 for security purposes.

[079] Network access servers 410 may be included in the service center 106 and may provide the hardware and software for sending and receiving information to voice network 410 in processing the applications provided by the service center. For example, network access servers 410 may include a Caller ID ("CID") functionality for retrieving caller ID information from voice network 104, a Click To Dial ("CTD") functionality for instructing an intelligent peripheral in the voice network to place a call via an SSP, and/or a Real Time Call Management ("RTCM") functionality for interfacing with an ISCP of the voice network.

[080] Network access plane 506 may also include one or more back end server(s) 420. These back end server(s) 420 may include hardware and/or software for interfacing service center 106 and voice network 104. Back end server(s) 420 may be connected to service center 106 by a network, by a direct connection, or in any other suitable manner. Further, back end server(s) 420 may connect to one or more devices in voice network 104 by a network, a direct connection, or in any other suitable manner.

[081] Back end server(s) 420 may include, for example, a server providing a voice mail retrieval and notification function. This voice mail retrieval and notification function may include the capability to receive notifications when a user receives a voice mail, physically call a user's voice mail system, enter the appropriate codes to retrieve the voice mail, retrieve the voice mail, convert the voice mail to a digital file, and send it to digital companion servers 406.

[082] Additionally, these back end server(s) 420 may also include, for example, a directory assistance server. This directory assistance server may interface service center 106 with a Reverse Directory Assistance Gateway ("RDA Gateway") of the voice network 104. An RDA Gateway is a device for issuing requests to a Data Operations Center ("DOC") of voice network 104 for name and/or address information associated with a phone number and receiving the name and/or phone number in response to this request.

[083] In another example, back end server(s) 420 may include a wireless internet gateway that is used for interfacing with a Mobile Switching center ("MSC") of a wireless voice network. As with the above-described back end server(s) 420, this wireless internet gateway may be used for converting requests and information between the formats used by service center 106 and those used by the wireless voice network.

[084] In yet another example, back end server(s) 420 may include a Single Number Short Message Service ("SN SMS") server for interfacing service center 106 with a Short Message Service ("SMS") gateway in voice network 104. This may be used to permit the customer to have SMS messages addressed to their home phone number directed to an SMS capable device of the users choosing.

[085] Also, back end server(s) 420 may include a conference blasting server for instructing a conference bridge in voice network 106 to dial out via an SSP to the participants of a voice conference. Alternatively, for example, back end server(s) 420 may include a server for instructing an IP of the voice network to place a call between two parties by dialing out to each of the

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parties. The back end server(s) may also include the capability to instruct the bridge or IP device to call an audio digitizing device that can listen to the conference, convert the audio signals to digital format, and forward the digitized signals to a user device via, for example, an audio streaming server. The audio streaming server may, for example, allow a user to connect to it via, for example, the Internet. Additionally, the audio streaming device may buffer or record the signals to permit the user to pause, rewind, and/or fast-forward thru the conference.

[086] Voice network plane 508 includes the hardware and software included in the voice network 104, as discussed above with reference to Fig. 3. For example, voice network plane 508 may include ISCP SPACE 314, ISCP 302, intelligent peripherals 320, and SSP 308. Additionally, voice network plane 508 may also include the hardware and software included in a wireless carrier's network, such as, for example, the mobile switching center, etc.

Conference Call System Overview

[087] As explained, methods and systems consistent with certain aspects related to the present invention provide an environment from which a user may manage several different features associated with communicating over a telecommunications and/or information exchange network. Among those features are one or more conference call processes that automatically configure and maintain conference calls for a subscriber user through the subscriber user's scheduling calendar and/or address book. In one aspect of the invention, the conference call processes allow the subscriber user to schedule a conference call event that includes one or more other participant users. The processes automatically establish a conference call between the users at the scheduled time and/or date scheduled by the subscriber user.

[088] Fig. 6 shows an exemplary network environment 600 that performs the conference call processes consistent with certain aspects related to the present invention. As shown, environment 600 includes data

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network 603, voice network 604, service center 606, user terminal 612, back-end servers 620, conference bridge 630, user systems 637 and 640-1 to 640-N, users 645-1 to 645-N, and speech recognition server 650.

[089] User terminal 612 is a terminal operated by a subscriber user 605 that is configured and operates in a manner consistent with user terminal 112 shown in Fig. 1. Data network 602 is a network that provides communications between various entities in environment 600, such as user terminal 612 and server center 606, in a manner consistent with data network 102 described above in connection with Fig. 1. Voice network 602 is a telephony-based network that provides communications between selected entities in environment 600, such as users 645-1 to 645-N and conference bridge 630 in a manner consistent with voice network 104 described above in connection with Fig. 1.

[090] Service center 606 provides a platform for managing communications over data network 602 and voice network 604 in a manner consistent with service center 106 described above in connection with Fig. 1. In one aspect of the invention, server center 606 includes Digital Companion Server ("DCS") 607 that provides customer related services in a manner consistent with digital companion server 406 described above in connection with Figs. 4 and 5.

[091] Back end servers 620 may be one or more computing systems that interface service center 606 and voice network 604. Back end servers 620 may include hardware and/or software and operates in a manner consistent with back-end servers 420 described above in connection with Fig. 5. In one aspect of the invention, back end servers 620 includes a Conference Blasting Application ("CBA") server 610 that executes software to perform one or more conference call processes, including a process that generates messages used by environment 600 to set up a conference call associated with subscriber user 605.

[092] Conference bridge 630 is a network bridge that establishes communications between selected ones of users 645-1 to 645-N and

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subscriber user 605. In one aspect of the invention, conference bridge 630 receives instructions from DCS 607 to set up conference calls in accordance with a conference call event configured by DCS 607. Further, conference bridge 630 provides telecommunication services associated with one or more calls during a conference call process.

[093] User systems 640-1 to 640-N may be communication systems operated by respective users 645-1 to 645-N. Through user systems 640-1 to 640-N and voice network 604, users 645-1 to 645-N may communicate with each other using one or more types of communication devices. Accordingly, user systems 640-1 to 640-N may represent telephony based devices, such as wireline or wireless telephones. Alternatively, user system 640-1 to 640-N may represent user terminals that operate in a manner consistent with user terminal 612. User 605 may also communicate with users 640-1 to 640-N through user system 637 and voice network 604.

[094] Speech recognition server 650 is a computing system that converts audio information provided by conference bridge 630 into speech information. Server 650 transcribes the speech information into text data that is stored in one or more data storage devices (not shown). In one aspect of the invention, speech recognition server 650 is located external to any components of environment 600. Alternatively, server 650 may be included as a server within back end servers 620.

[095] In one aspect of the invention, user 605 may be a subscriber to one or more services offered by service center 606. That is, user 605 may be a user that registers with a business entity associated with service center 606 to receive one or more communication services offered by the components of environment 600, such as service center 607. For example, user 605 may be a subscriber to service center 606 to receive calls through a preferred device. A preferred device is one of a plurality of communication and/or computing devices associated with the subscriber (e.g., user 605). Such devices include, for example, PDAs, wireless telephones, wireline telephones, etc. In one aspect of the invention, service center 607 may receive

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information pertaining to a call directed to user 605 from a calling party, and retrieve data corresponding to user 605 using the received information. Service center 607 may determine a preferred device of user 605 based on the retrieved information. Thereafter, service center 607 may ascertain whether the preferred device of the user requires a new mode of delivery, and if so, routes the call to the preferred device of the user to be compatible with the new mode of delivery. The methods and systems associated with the preferred device aspects related to the present invention are further described in U.S. Patent Application No. (_____), entitled "METHODS AND SYSTEMS FOR AUTOMATIC FORWARDING OF CALLS TO A PREFERRED DEVICE," Attorney Docket No. 03-1021.

Conference Call Processes

[096] Methods and systems consistent with certain aspects of the present invention automatically configure and maintain a conference call between user 605 and one or more users 645-1 to 645-N. Fig. 7 is a flowchart of an exemplary conference call configuration process that may be performed by environment 600. In one aspect of the invention, user 605 may set up one or more conference call scheduling events through user terminal 612 and service center 606 (Step 705). For example, user 605 may access a calendar application and an address book provided by service center 606 to add participant users to a proposed conference call meeting. To do so, in one example, user 605 may access participant user information displayed in a graphical representation of an address book corresponding to user 605. Using an input device (e.g., a mouse, stylus, etc.), user 605 may select profile information for a participant user (e.g., user 645-1) from the address book and drop it in a conference call icon included in the calendar application for a proposed time and date (e.g., Monday, December 7, 2003, 9:00 AM EST). Alternatively, user 605 may select a date and time for a proposed conference call and add participant users to a participant list provided in graphical form on the display device in user terminal 612. In one aspect of the invention,

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service center 606 employs an application server function (e.g., application server function 516) within DCS 607 to allow user 605 to schedule a conference call to take place at a specific time and date. One skilled in the art will appreciate, however, that different mechanisms and methods may be employed by environment 600 to enable user 605 to schedule a conference call with one or more participant users 645-1 to 645-N without departing from the scope of the invention.

[097] Periodically, DCS 607 may scan the calendar application associated with user 605 to determine whether a conference call event is detected (Step 710). The conference call event may act as a trigger that directs DCS 607 to automatically set up a conference call for user 605. For example, DCS 607 may determine whether a current date and time matches, or is within some predetermined range of, a scheduled conference call for user 605. For instance, suppose user 605 previously scheduled a conference call with three participant users listed in user 605's address book for a particular date (e.g., Monday, November 17, 2003, at 10:00 AM EST). On that date and/or time (i.e., November 17, 2003), or sometime earlier, a DCS 607 function, such as calendar service function 518, may trigger a conference call event that instructs DCS 607 to begin configuring a conference call in accordance with the particulars set up by user 605 and represented in the user's calendar.

[098] Once a conference call event is detected, DCS 607 generates a Conference Blasting Application ("CBA") message (Step 720). The CBA message includes profile information for each participant user included in a participant list associated with the detected scheduled conference call. For example, DCS 607 may collect the telephone numbers and identifiers for each participant user and add them to the CBA message. Once the CBA message is generated, DCS 607 formats the message for an appropriate transmission protocol and sends the message to CBA server 610 (Step 730). In one aspect of the invention, DCS 607 may format the CBA message into an XML message and transmit the message to CBA server 610 using TCP/IP

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protocol. Other types of transmission protocols and information formats, however, may be implemented by service center 606 and DCS 607 to transport the CBA message to CBA server 610.

[0099] CBA server 610 may extract the information included in the CBA message to generate a CBA bridge message (Step 740). A CBA bridge message is a set of commands that are directed to a specific bridge (e.g., conference bridge 630) that instructs the bridge to set up a conference call by dialing out to the participant user's identified in the CBA message. In one aspect of the invention, CBA server 610 may determine which one a plurality of bridges (not shown) in environment 600 is configured to handle conference call communications with any of users 645-1 to 645-N. CBA server 610 may identify conference bridge 630 as being dedicated to handle conference call operations and thus generate a CBA message based on the identification of bridge 630. That is, CBA server 610 may format a message that includes one or more commands directed to conference bridge 630.

[0100] Conference bridge 630 receives the CBA bridge message and, based on the commands included in the message, sets up a conference call that includes user 605 and any participant users identified in the CBA bridge message (Step 750). In one aspect of the invention, conference bridge 750 collects the telephone number for each participant user 645-1 to 645-N from the CBA bridge message. Once the conference call is configured, conference bridge 630 uses the collected telephone numbers to establish a connection with participant users 645-1 to 645-N (Step 750). In one aspect of the invention, conference bridge 630 may dial out to each user system 640-1 to 640-N operated by a participant user 645-1 to 645-N using the signaling protocols and/or components implemented by voice network 604, such as SCPs, SSPs, SCPs, ISCPs, SS7 protocol infrastructures, etc.

[0101] In one aspect of the invention, one or more participant users 645-1 to 645-N may be subscribers to service center 606. Accordingly, when generating the CBA message, DCS 607 may determine whether a participant user is a subscriber to service center 606. If so, DCS 607 may determine

whether the subscriber participant user 645-1 to 645-N has subscribed for a preferred device service, (i.e., a Digital Companion (DC) user). In such a case, DCS 607, through components of service center 606 and/or DCS 607 (e.g., network access servers 410, application server 516, etc.) formats the CBA message to reflect the type of preferred device by which the subscriber participant user prefers to be contacted. For example, if the subscriber participant user indicates a desire to be contacted via a PDA, service center 606 formats the CBA message to include the contact information required to establish a connection with the subscriber participant user's PDA. Further, if the type of delivery mode requires adjustment because of the type of preferred device indicated by the subscriber participant user, DCS 607 may format the CBA message to reflect the type of delivery mode that is to be used to establish contact with the subscriber participant user. The manner by which DCS 607 determines and establishes communications with a subscriber participant user having a preferred device service may be consistent with the features and processes described in U.S. Patent Application No. (_____), entitled "METHODS AND SYSTEMS FOR AUTOMATIC FORWARDING OF CALLS TO A PREFERRED DEVICE," Attorney Docket No. 03-1021, previously referenced.

[0102] Once communication is established with a participant user (e.g., user 645-1), methods and system consistent with certain aspects related to the present invention may perform a conference call process. Fig. 8 shows a flowchart of an exemplary conference call process that may be performed by one or more components of environment 600 in a manner consistent with certain aspects related to the present invention. As shown, the conference call process may begin when conference bridge 630 attempts to establish communication contact with a target participant user (e.g., user 645-1) by calling the participant user's corresponding user system (e.g., user system 640-1) (Step 810). If the target participant user does not answer the conference call from bridge 630 (Step 812; NO), conference bridge 630 may

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generate and provide a NO CONTACT message to DCS 607 through CBA server 610 (Step 815).

[0103] In one aspect of the invention, the NO CONTACT message may indicate to DCS 607 that the target participant user did not answer the conference call. Additionally, the NO CONTACT message may also identify the target participant user, the telephone number called, the time called, and any other type of information associated with the attempted conference call contact operation. DCS 607 may generate a DCS NO CONTACT message including some or all of the NO CONTACT message information provided by bridge 630, and send the DCS NO CONTACT message to user terminal 612 through data network 602. Alternatively, or additionally, DCS 607 may provide the DCS NO CONTACT message information to conference bridge 630 through CBA server 610. Bridge 630 may provide the NO CONTACT information to user 605 through user system 637 in, for example, voice format. In another aspect of the invention, conference bridge 630 may provide the NO CONTACT message information to user system 637 upon determining that the target participant user did not answer the conference call. One skilled in the art will appreciate that the above described examples are not intended to be limiting and that methods and system consistent with aspects of the present invention may use other types of messages, information, and components to notify user 605 of the unanswered conference call.

[0104] Following the generation and provision of the no-contact message, conference bridge 630 may determine whether there are any more participant users included in the scheduled conference call that bridge 630 has not yet contacted (Step 816). If so (Step 816; YES), bridge 630 attempts to establish contact with the new target participant user (Step 810). On the other hand, if conference bridge 630 has attempted to contact every participant user identified in the CBA bridge message for the conference call, and none of the users have answered the call (or communication could not be established for another reason) (Step 816; NO), conference bridge 630 may

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generate a no conference call message to DCS 607 indicating that the scheduled conference call can not be processed because of the lack of participant users (Step 818). In one aspect of the invention, DCS 607 may provide the no conference call message to user terminal 612 in a manner similar to that described above in connection with the NO CONTACT message. Further, bridge 630 may provide the no conference call message to user system 637 in a manner similar to that described above in connection with the NO CONTACT message.

[0105] Referring back to Fig. 8, if a target participant user did answer the conference call (Step 812; YES), conference bridge 630 may generate and play one or more announcements to the target participant user (via the user's corresponding user system) (Step 820). The announcement may include a voice message indicating to the participant user that the target participant user is scheduled to participate in a scheduled conference call. The announcement may also identify other participant users (e.g., 645-N) and the user initiating the conference call (e.g., user 605). Other types of announcements may be provided by conference bridge 630 that include additional information associated with the conference call, such as subject of the call, a proposed agenda, etc. Also, additional messages may be provided to user 605, such as a message indicating that a conference call has been configured in accordance with a scheduled conference call previously set up by user 605 through service center 606. Further, other components of environment 600 (shown or not shown in Fig. 6) may provide the one or more announcements to the participant user. That is, conference bridge 630 may leverage another processing component to provide conference call participant announcements to participant users 645-1 to 645-N.

[0106] Additionally, the announcements may be pre-configured and/or selected by user 605 using service center 606. In one example, when scheduling a conference call, user 605 may provide service center 606 with a pre-configured announcement that is used by conference bridge 630 when a participant user answers a conference call. In this instance, DCS 607 may

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include the pre-configured announcement in the CBA message provided to CBA server 610. And, in turn, CBA server 610 may include the pre-configured announcement message in the CBA bridge message provided to conference bridge 630. In another aspect of the invention, conference bridge 630 may provide a generic conference call announcement that is common to other conference calls established by bridge 630 prior to, during, or subsequent to, the conference call established for user 605 in the conference call process described in connection with Figures 7 and 8.

[0107] Upon answering the conference call, the target participant user may have the option of accepting or not accepting the call. If the target participant user accepts the call (Step 830; YES), the conference call process proceeds at Step 850, described below. If, on the other hand, the target participant user does not accept the conference call (Step 830; NO), conference bridge 630 may perform a participant feedback process consistent with certain aspects of the present invention (Step 840).

[0108] The participant feedback process provides the target participant user one or more options for refusing the conference call. For example, bridge 630 may provide a feedback message to the target participant user including a list of options the target participant user may select to refuse the call. These options may include refusing the conference call outright (i.e., do not accept the call and hang up), leave a message for the initiating user (e.g., user 605), set an alternate telephone number or preferred device to contact the target participant user, and instruct the conference bridge 630 to dial back in a user-specified amount of time (e.g., a number of minutes). The target participant user may make the appropriate selection using voice commands and/or Dual Tone Multi-Frequency ("DTMF") responses that are processed by conference bridge 630 using speech recognition and/or DTMF processes, respectively.

[0109] If the target participant user refuses the conference call, conference bridge 630 may generate and provide a refusal message to user 605 through user system 637 by way of voice network 604 and/or user

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terminal 612 by way of CBA server 610 and DCS 607. Alternatively, if the target participant user wishes to leave a message for user 605, conference bridge 630 may record the target participant user's response to the conference call and provide the response to user 605 in a similar manner (e.g., via user system 637 and/or user terminal 612). Also, if the target participant user wishes to instruct bridge 630 to call them back using an alternate telephone number, conference bridge 630 may receive the alternate number from the target participant user and attempt to re-establish the conference call connection with the target participant user based on the alternate number (Step 810). Moreover, if the target participant user instructs bridge 630 to call back, bridge 630 may request and record a specified period of time that bridge 630 is to attempt to contact the target participant user. When the specified period of time expires, bridge 630 may perform Step 810 to re-execute the conference call process for that target participant user.

[0110] It should be noted that conference bridge may offer and perform any of the feedback process options individually or in combination with each other. That is, bridge 630 may allow the target participant user to set an alternate telephone number and instruct bridge 630 to call back in a specified period of time. Further, the participant feedback process options described above are exemplary and not intended to be limiting, thus conference bridge 630 may offer and process other options associated with a target participant user's refusal to accept a conference call. Once the participant feedback process is completed, the conference call process may end.

[0111] Referring back to Fig. 8, if the target participant user accepts the conference call (Step 830; YES), conference bridge 630 may generate and provide a participant user acceptance message to DCS 607 (Step 850). The acceptance message includes information reflecting the target participant user's acceptance of the call. This information may also include information identifying the target participant user, the status of the target participant user's acceptance of the call, and any other type of information associated with the conference call and/or the target participant user. For example, the

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acceptance message may include voice or text information indicating that the target participant user has accepted the conference call and is joining in the conference initiated by DCS 607. The target participant user may be identified by a conference call identifier, the participant user's name, and/or the participant user's telephone number. DCS 605 forwards this information to user 605 in the appropriate format through user terminal 612. Alternatively, or additionally, conference bridge 630 may provide the acceptance message to user 605 through user system 637 and voice network 604.

[0112] Once the target participant user accepts the conference call, the user may participate in the conference with user 605 and any other participant users (645-1 to 645-N) that have also joined the call (e.g., 860). During the conference call, conference bridge 630 may detect a disconnection event reflecting a condition when the target participant user's connection with conference bridge 630 has been terminated.

[0113] Termination of the participant user's call may be the result of many actions. For instance, the participant user may have ended the call voluntarily (e.g., hung up) or the connection between the participant user and conference bridge 630 may have been terminated for unwanted reasons (i.e., dropped off, line disconnections, user system communication problems, voice network 604 failure, etc.). Accordingly, in one embodiment, the conference call process may determine whether the target participant user ends their participation in the conference call (Step 870). Alternatively, or additionally, conference bridge 630 may monitor the connections of each participant user 630 to detect an involuntary disconnection (e.g., Step 871, described further below).

[0114] In one embodiment of the invention, the conference call process may implement one or more rules to determine whether the participant user voluntarily or involuntarily dropped from the conference call. For example, one or more rules may be stored as software code in a memory device that enables the conference call process to determine the type of disconnection that may have occurred. To do so, in one aspect of the invention, the

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conference call process may determine the type of device the participant user (i.e., user system 640) used to participate in the call. Based on the type of device, the process may determine whether the participant user ended participation voluntarily (e.g., hung up) or was disconnected. For example, a rule may be established that determines that an involuntary disconnection event occurs when the participant user was using a wireless or IP phone when the user's participation ended before the conference call was completed. On the other hand, another rule may establish that the user voluntarily ended participation in the call when they were using a wireline device. The above rules are intended to be exemplary and different rules and/or automated logic may be implemented with methods and systems consistent with this invention to determine the type of drop off that occurs during a conference call.

[0115] If the target participant user voluntarily ends participation in the call (Step 870; YES), conference bridge 630 may generate and provide a participant user termination message to DCS 880 (Step 880). The termination message may include information identifying the target participant user, the participant user's telephone number, a time that the participant user ended participation, and any other type of information bridge 630 may be programmed to provide upon the target participant user terminating their participation in the conference call. DCS 607 may provide the termination message to user 605 through user terminal 612. Alternatively, or additionally, conference bridge 630 may provide the termination message to user 605 through user system 637 and voice network 604.

[0116] Following the delivery of the termination message, the conference call process may end for the target participant user. Environment 600 may, however, continue to execute the conference call process for any other remaining participant users and user 605 until the conference call no longer includes any participant users. In one aspect of the invention, the conference call process may continue for participant users 645-1 to 645-N following termination of the call by user 605. Additionally, while the

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conference call is still established and includes at least one participant user and/or user 605, any participant users (or user 605) may rejoin the call by contacting conference bridge 630 using a predetermined telephone number and/or conference identifier. Alternatively, user 605 may instruct conference bridge 630, through a CBA message generated by DCS 607, to contact a participant user who previously terminated their participation in the call.

These processes are described with reference to Steps 871-877.

[0117] Referring back to Step 870, if the participant user did not end the call voluntarily (Step 870; NO), the conference call process may determine whether the participant user was dropped from the call involuntarily due to a disconnection event (e.g., diminishing cellular phone communication power, communication line disconnections, power loss, network infrastructure failures, etc.) (Step 871). If the user was not dropped from the call (Step 871; NO), the user continues with participation in the conference call (Step 860). On the other hand, if the participant user was dropped from the call (Step 871; YES), the conference call process determines whether the participant user is a DC user (i.e., a user registered with service center 606 and has access to the services offered by DCS 607) (Step 872). If the user is not a DC user (Step 872; NO), a best effort re-establishment process is performed (Step 873).

[0118] In one embodiment of the invention, the best effort re-establishment process may determine the phone number of the device the participant user operated when participating in the conference call. Using that number, the best effort re-establishment process may leverage conference bridge 630 to call the number in an attempt to re-establish communications with the participant user. If communications are not re-established with the participant user (Step 874; NO), the process repeats communication attempts for a predetermined number of times before proceeding to Step 880. If, however, communications is re-established with the participant user (Step 874; YES), the conference call process may query the participant user whether they wish to be reconnected or dropped from the conference call

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(Step 875). If the user wishes to end participation (Step 875; YES), the process proceeds to Step 880. If the user wishes to continue participation in the call (Step 875; NO), the process returns to Step 860.

[0119] Referring back to Step 872, in the event the participant user is a DC user (Step 872; YES), the conference call process may perform a DC user re-establishment process (Step 876). In this process, the profile for the participant user is accessed from DCS 607 to determine, among other things, the user's preferred device list, the telephone or contact numbers for each preferred device. Further, the DC user re-establishment process may collect information from DCS 607 indicating whether the user desires to be automatically contacted during a conference call drop event (i.e., if the user was disconnected during a conference call). The conference call process uses the collected profile information to attempt to re-establish communications with the participant user through the user's preferred device. If one device does not work, the process may attempt communications with another preferred device. If communication is not established after exhausting the desired profile communication contact techniques (Step 877; NO), the conference call process proceeds to Step 880. If, however, communications was re-established with the participant user (Step 877; YES), the process continues at Step 875.

[0120] It should be noted that the conference call process determines the participant user's preferred device at the time of participation in the conference call. For example, consider an exemplary scenario where a participant user is participating in a conference call using a wireless device while traveling in a vehicle to a place of employment (i.e., their office). If the participant user is dropped from the conference call (or the user hangs up when they arrive at their place of employment), the user may access DCS 607 to reassign their preferred device (or reprioritize) to their office wireline device. Accordingly, DCS 607 may attempt to re-connect with the user through the user's office wireline device. Alternatively, if DCS 607 unsuccessfully attempts to re-connect the participant user to the conference

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call through the wireless device used during the call prior to drop off, DCS 607 may be configured to call the user's office wireline device. Accordingly, DCS 607 determines a DC participant user's preferred device based on either the device used by the user during the conference call, or through the user's profile registered with DCS 607.

Conference Call Transcription Processes

[0121] In addition to establishing and maintaining conference calls, methods and systems consistent with certain aspects related to the present invention enable voice communications during a conference call to be recorded and transcribed. In one aspect of the invention, conference bridge 630 may independently collect audio information from each channel of a conference call (i.e., audio signals received from each user system 640-1 to 640-N that are participating in a conference call). Bridge 630 may route the collected audio information to speech recognition server 650, which converts the audio information to speech information. Conference bridge 630 may directly route the audio information to server 650 or indirectly route the signals through back end servers 620 or other components of environment 600. Speech recognition server 650 may create a transcript of text data from the speech information that identifies each speaker (i.e., user 645-1 to 645-N and user 605) based on the speaker's communication connection established between voice network 604 and the speaker's corresponding user system 640-1 to 640-N. Speech recognition server 650 may also synchronize the transcript using time data that is encoded in the recorded audio information provided by conference bridge 630. In one aspect of the invention, speech recognition server 650 stores any audio information that cannot be converted to text data as audio files. Server 650 may insert pointers to the audio files in an appropriate location of the transcript. Thus, speech recognition server 650 may create a transcript that includes a sequence of text data that corresponds to the conversations between participant users 645-1 to 645-1

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and user 605 during a conference call. The transcript may include, at some temporal based location in the transcript where text data could not be converted, a pointer to one or more audio files that includes non-converted audio information corresponding to a speaker that was talking during that moment in time reflected in the transcript.

Speech recognition server 650 may be configured to store the transcript in one or more storage devices located within server 650 or external to server 650, such as a storage device associated with database 555. Further, speech recognition server 650 may format the transcript into a file that is provided to user 605's email address included in DCS 607. Alternatively, speech recognition server 650 may allow subscriber user 605 to access the stored transcript through service center 606.

Ad Hoc Conferencing

[0122] Methods and systems consistent with certain embodiments of the present invention allow user 605 to set up conference call scheduling events that system 600 uses to configure subsequent conference calls. In another embodiment, system 600 may also enable user 605 to configure a conference call to take place immediately, or nearly immediately, following user 605's request. Figure 9 show a flowchart of an exemplary ad-hoc conference call process that may be performed by system 600 consistent with certain aspects of the invention. Using the ad-hoc conference call process, user 605 may leverage DCS 607 to create a conference call "on the fly" such that DCS 607 does not monitor a scheduling application (e.g., calendar application) to detect a scheduling event. In other words, user 605 may set up a scheduling event to take place at the time user 605 requests a conference call (Step 910).

[0123] When setting up a conference call scheduling event "on the fly," user 605 may access service center 606 through user terminal 612 to establish a conference call with participant users 645-1 to 645-N. In one aspect, user 605 may access DCS 607 to identify one or more participant users 645-1 to 645-N. For example, user 605 may access an address book

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that includes a list of participant users from which user 605 may select to participate in a conference call. This may be performed, for example, by highlighting one or more participant users listed in a menu displayed to user 605 via user interface 612. Alternatively, or additionally, user 605 may designate one or more participant users 645-1 to 645-N that are not included in the list. In this instance, user 605 may provide DCS 607 with contact information (e.g., telephone numbers, IP addresses, etc.) corresponding to each designated participant user.

[0124] Once user 605 selects and/or designates target participant users to be included in a conference call, DCS 607 automatically sets up and initiates the conference call. In one embodiment, DCS 607 may also identify a conference bridge that may be used to establish and facilitate the conference call (Step 920). For example, user 605 may have a profile entry registered with DCS 607 that identifies a conference bridge for user 605. Accordingly, DCS 607 may determine whether that conference bridge is available for use when user 605 initiates a request for a conference call. If the bridge is available, DCS collects the bridge identification number for user 605's designated bridge and adds the number to a generated CBA message that further includes the contact information for the target participant users designated by user 605. On the other hand, if the designated bridge is not available, or user 605 does not have a designated bridge, DCS 607 may search for and temporarily assigns a conference bridge to user 605 for the duration of the conference call requested by user 605. In one embodiment, DCS 607 may search a pool of conference bridges maintained in a database for an available bridge that may be used to facilitate the requested conference call.

[0125] Once an available conference bridge (e.g., conference bridge 630) is identified, DCS 607 collects and adds a bridge identification number corresponding to the available bridge to the CBA message (Step 930). DCS 607 then sends the CBA message to CBA server 610 (Step 940). Once received, CBA server 610 extracts the contact information and bridge

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identification number and creates a CBA bridge message in a manner consistent with the process described in connection with Step 740 of Figure 7 (Step 950). CBA server 610 then sends the CBA bridge message to the available conference bridge 630, which uses commands included in the CBA message to set up the conference call requested by user 605 in a manner consistent with the processes previously described in connection with Steps 750 and 760 of Figure 7 (Step 960).

[0126] Accordingly, methods and systems consistent with certain embodiments of the present invention enable user 605 to set up a conference call to take place immediately following the time when user 650 initiates the call request. These embodiments allow user 605 to select participant users 645-1 to 645-N and allow system 600 to automatically configure and establish the conference call at the time user 605 requests a conference call with the selected participant users.

Conclusion

[0127] While the present invention has been described in connection with various embodiments, many modifications will be readily apparent to those skilled in the art. One skilled in the art will also appreciate that all or part of the systems and methods consistent with the present invention may be stored on or read from computer-readable media, such as secondary storage devices, like hard disks, floppy disks, and CD-ROM; a carrier wave received from a network such as the Internet; or other forms of ROM or RAM. Accordingly, embodiments of the invention are not limited to the above described embodiments and examples, but instead is defined by the appended claims in light of their full scope of equivalents..

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WHAT IS CLAIMED IS:

1. A method for providing a conference call, comprising:
detecting a conference call event that was previously configured by a subscriber user;
identifying participant users associated with the conference call event;
contacting the participant users;
receiving a response from the participant users; and
establishing a conference call between the participant users and the subscriber user based on the received responses,
wherein at least one of the detecting and identifying steps is performed without user intervention.
2. The method of claim 1, wherein detecting a conference call event includes:
scanning a data structure for the conference call event.
3. The method of claim 2, wherein the data structure is a calendar application associated with the subscriber user.
4. The method of claim 1, wherein the conference call event is a trigger indicating a proposed conference call previously scheduled by the user.
5. The method of claim 4, wherein the proposed conference call identifies the participant users and identifying participant users includes:
collecting an identifier for the participant users from a first data structure corresponding to the conference call event; and
collecting contact information for the participant users from a second data structure based on the participant user identifiers.

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6. The method of claim 5, wherein the first data structure is a calendar application and the second data structure is an address book listing at least the participant users and their corresponding contact information.

7. The method of claim 1, wherein contacting the identified participant users includes:

- collecting contact information associated with the participant users; and
- establishing a communication connection with the participant users using the contact information.

8. The method of claim 7, wherein the contact information comprises telephone numbers associated with the participant users and establishing a communication connection includes:

- dialing out to participant users using a respective telephone number.

9. The method of claim 1, wherein contacting the participant users includes calling the participant users using a telephone number and receiving a response from the participant users includes:

- for a participant user:

- providing an indication to the subscriber user reflecting whether the participant user answers the call.

10. The method of claim 9, wherein providing an indication includes:

- providing a no contact message to the subscriber user when the participant user does not answer the call.

11. The method of claim 10, further including:

- providing a no conference call message to the subscriber user when none of the participant users answer their respective call.

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12. The method of claim 9, wherein providing an indication includes:
providing an acceptance message to the subscriber user when the participant user answers the call.

13. The method of claim 12, wherein providing the acceptance message includes:

determining whether the participant user accepts the call.

14. The method of claim 13, wherein determining whether the participant user accepts the call includes:

performing a feedback process when the participant user declines the call.

15. The method of claim 14, wherein performing a feedback process includes:
providing options for the participant user to decline the call, wherein the options include at least one of:

allowing the participant user to record a message for subsequent play back to the subscriber user,

declining the call without any further processing by the participant user;

allowing the participant user to set an alternate telephone number for contacting the participant user, and

allowing the participant user to set a period of time in which the participant user is to be contacted again.

16. The method of claim 1, wherein establishing a conference call includes:
establishing a communication connection between participant users and the subscriber user such that the subscriber user and the participant users may conduct a conference call.

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17. The method of claim 1, further comprising:
detecting when one of the participant users terminates its contact during the conference call;
providing a termination message to the subscriber user indicating that the one participant user has ended participation in the conference call.

18. The method of claim 1, further comprising:
recording audio information associated with the subscriber user and participant users' participation during the conference call;
converting at least some of the audio information to text information; and
recording the text information in a transcript reflecting a textual temporal based representation of communications that have taken place between the users during the conference call.

19. The method of claim 18, further comprising:
storing a portion of the audio information that cannot be converted to text information as an audio file; and
inserting a pointer to the audio file in the transcript.

20. The method of claim 19, further comprising:
providing the transcript to the subscriber user.

21. The method of claim 19, wherein providing the transcript includes attaching the transcript to an e-mail addressed to the subscriber user.

22. The method of claim 1, wherein contacting the participant users includes:
determining whether a participant user has a preferred device registered with a service center, and

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contacting a participant user through the preferred device when the participant user has a registered preferred device.

23. The method of claim 1, wherein the detecting and identifying steps are performed by a digital companion server that provides services to the subscriber user, and wherein contacting the participant users includes:

generating, by the digital companion server, a first message that includes at least instructions for configuring the conference call and telephone numbers for the participant users; and

providing the first message to a conference blasting server.

24. The method of claim 23, wherein providing the first message includes:

generating a second message by the conference blasting server based on the first message, wherein the second message includes instructions for setting up the conference call and the telephone numbers; and

providing the second message to a conference bridge.

25. The method of claim 24, wherein providing the second message includes:

calling, by the conference bridge, the participant users using the telephone numbers included in the second message.

26. The method of claim 1, wherein the subscriber user configures the conference call event by scheduling the conference call in a calendar application for a predetermined date and time and adding the names of the participant users that the subscriber user intends to participate in the conference call.

27. The method of claim 26, comprising receiving input from the subscriber user to add the names of participant users from an address book to the calendar application, the input being received via a graphical user interface.

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28. The method of claim 1, further comprising:
detecting when a first participant users was dropped from the conference call;
and
determining whether the first participant user has a preferred device registered
with a service center.

29. The method of claim 28, wherein when the first participant user has a
registered preferred device,
attempting to contact the first participant user through the preferred device.

30. The method of claim 29, further comprising:
re-establishing the first user with the conference call based on a determination
that the first user wishes to continue participation in the conference call.

31. The method of claim 28, wherein when the first participant user does not
have a registered preferred device,
determining contact information associated with a device used by the first
participant user to participate in the conference call; and
re-establishing the first user with the conference call using the contact
information.

32. The method of claim 1, further comprising:
detecting when a first participant user has terminated participation in the
conference call; and
determining whether the first participant user voluntarily or involuntarily
terminated participation in the conference call based on the type of device the first
participant user was operating during participation in the conference call.

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33. A method for providing a conference call, the method performed by a server comprising:

- detecting a conference call event previously configured by a subscriber user that indicates when a conference call should be established between the subscriber user and participant users;

- collecting contact information for the participant users;

- providing a message including the contact information to a conference server that is configured to instruct a bridge to establish the conference call between the users by calling the participant users using the contact information included in the message; and

- receiving a response message from the conference server including information associated with the conference call and at least one participant user.

34. The method of claim 33, wherein detecting a conference call event includes:

- scanning a calendar application to determine whether the subscriber user has scheduled a conference call.

35. The method of claim 34, wherein collecting contact information includes determining the participant users from the calendar application;

- accessing an address book associated with the subscriber user for collecting the contact information for the participant users; and

- generating the message using the collected contact information.

36. The method of claim 33, wherein providing a message includes:

- generating the message using the collected contact information and instructions associated with a date for commencing the conference call, and

- using the message by the conference server to generate a second message for instructing the bridge to establish the conference call.

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37. The method of claim 33, comprising providing the response message by the bridge when attempting to establish communications with the participant users over a voice network.

38. The method of claim 33, wherein the response message comprises at least one of:

- information stating that at least one participant user is declining the call;
- information stating that at least one participant user is not answering the call;
- information stating that at least one participant user is accepting the call; and
- none of the participant users is answering the call.

39. A method for participating in a conference call automatically established by at least one or more communication entity, the method comprising:

- scheduling a conference call at a predetermined time using a calendar application, wherein scheduling includes identifying participant users that are to participate in the conference call; and

- at the predetermined time:

- receiving an indication that a conference call has been configured in accordance with the scheduled conference call,

- receiving a indication that at least one participant user has either joined or declined to join the conference call, and

- during the conference call, receiving an indication that another participant user has ended communications with the conference call when that participant user terminates a connection with a bridge that establishes the conference call in accordance with instructions provided by a server that executes the calendar application.

40. A method for providing a conference call, comprising:

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receiving a message from a first server including instructions for establishing a conference call between participant users and a subscriber user that previously scheduled the conference call with a second server, wherein the second server automatically initiates configuration of the conference call based on the subscriber user's schedule by providing to the first server contact information for the participant users and the message includes the contact information;

calling the participant users using the contact information included in the message;

receiving a response from the participant users, wherein the response reflects whether the participant user has answered, accepted, not answered, or declined to accept the call; and

establishing a conference call between the users based on the response received from the participant users.

41. A system for providing a conference call, comprising:

means for detecting a conference call event that was previously configured by a subscriber user;

means for identifying participant users associated with the conference call event;

means for contacting the participant users;

means for receiving a response from the participant users; and

means for establishing a conference call between the participant users and the subscriber user based on the received responses,

wherein at least one of the means for detecting and means for identifying perform their functions without user intervention.

42. The system of claim 41, wherein the means for detecting a conference call event comprises:

means for scanning a data structure for the conference call event.

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43. The system of claim 42, wherein the data structure is a calendar application associated with the subscriber user.

44. The system of claim 41, wherein the conference call event is a trigger indicating a proposed conference call previously scheduled by the user.

45. The system of claim 44, wherein the proposed conference call identifies the participant users and the means for identifying participant users includes:

means for collecting identifiers for the participant users from a first data structure corresponding to the conference call event; and

means for collecting contact information for the participant users from a second data structure based on the participant user identifiers.

46. The system of claim 45, wherein the first data structure is a calendar application and the second data structure is an address book listing at least the participant users and their corresponding contact information.

47. The system of claim 41, wherein the means for contacting the identified participant users includes:

means for collecting contact information associated with the participant users;
and

means for establishing a communication connection with the participant users using the contact information.

48. The system of claim 47, wherein the contact information comprises telephone numbers associated with participant users, and the means for establishing a communication connection includes:

means for dialing out to participant users using corresponding telephone numbers.

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49. The system of claim 41, wherein the means for contacting the participant users includes means for calling the participant users using telephone numbers and the means for receiving a response from the participant users includes:

means for providing, for the participant users, an indication to the subscriber user reflecting whether a specific participant user answers the call.

50. The system of claim 49, wherein the means for providing an indication includes:

means for providing a no contact message to the subscriber user when the participant user does not answer the call.

51. The system of claim 50, further including:

means for providing a no conference call message to the subscriber user when none of the participant users answer their respective call.

52. The system of claim 49, wherein the means for providing an indication includes:

means for providing an acceptance message to the subscriber user when a participant user answers the call.

53. The system of claim 52, wherein the means for providing the acceptance message includes:

means for determining whether a participant user accepts the call.

54. The system of claim 53, wherein the means for determining whether a participant user accepts the call includes:

means for performing a feedback process when a participant user declines the call.

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55. The system of claim 54, wherein the means for performing a feedback process includes:

means for providing options for the participant user to decline the call, wherein the options include at least one of:

- allowing the participant user to record a message for subsequent play back to the subscriber user,
- declining the call without any further processing by the participant user;
- allowing the participant user to set an alternate telephone number for contacting the participant user, and
- allowing the participant user to set a period of time in which the participant user is to be contacted again.

56. The system of claim 41, wherein the means for establishing a conference call includes:

means for establishing communication connections between participant users and the subscriber user, such that the users may conduct a conference call.

57. The system of claim 41, further comprising:

means for detecting when one of the participant users terminates its contact during the conference call;

means for providing a termination message to the subscriber user indicating that the one participant user has ended participation in the conference call.

58. The system of claim 41, further comprising:

means for recording audio information associated with participant user's participation during the conference call;

means for converting at least some of the audio information to text information;
and

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means for recording the text information in a transcript reflecting a textual temporal based representation of communications that have taken place between the users during the conference call.

59. The system of claim 58, further comprising:

means for storing a portion of the audio information that cannot be converted to text information as an audio file; and

means for inserting a pointer to the audio file in the transcript.

60. The system of claim 59, further comprising:

means for providing the transcript to the subscriber user.

61. The system of claim 59, wherein the means for providing the transcript comprises means for attaching the transcript to an e-mail addressed to the subscriber user.

62. The system of claim 41, wherein means for contacting the participant users comprises:

means for determining, for participant users, whether a specific participant user has a preferred device registered with a service center; and

means for contacting a participant user through a preferred device when the participant user has a registered preferred device.

63. The system of claim 41, wherein the means for detecting and means for identifying steps are performed by a digital companion server that provides services to the subscriber user, and wherein the means for contacting the participant users comprises:

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means for generating, by the digital companion server, a first message that includes at least instructions for configuring the conference call and telephone numbers for the participant users; and

means for providing the first message to a conference blasting server.

64. The system of claim 63, wherein the means for providing the first message comprises:

means for generating a second message by the conference blasting server based on the first message, wherein the second message includes instructions for setting up the conference call and the telephone numbers; and

means for providing the second message to a conference bridge.

65. The system of claim 64, wherein the means for providing the second message includes:

means for calling, by the conference bridge, the participant users using the telephone numbers included in the second message.

66. The system of claim 41, wherein the subscriber user configures the conference call event by scheduling the conference call in a calendar application for a predetermined time and adding the names of the participant users that the subscriber user intends to participate in the conference call.

67. The system of claim 66, comprising receiving input from the subscriber user to add the names of participant users from an address book to the calendar application, the input being received via a graphical user interface.

68. The system of claim 41, further comprising:

means for detecting when a first participant users was dropped from the conference call; and

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means for determining whether the first participant user has a preferred device registered with a service center.

69. The system of claim 68, wherein when the first participant user has a registered preferred device,

means for attempting to contact the first participant user through the preferred device.

70. The system of claim 69, further comprising:

means for re-establishing the first user with the conference call based on a determination that the first user wishes to continue participation in the conference call.

71. The system of claim 68, further including

means for determining contact information associated with a device used by the first participant user to participate in the conference call when the first participant user does not have a registered preferred device; and

means for re-establishing the first user with the conference call using the contact information.

72. The system of claim 41, further comprising:

means for detecting when a first participant user has terminated participation in the conference call; and

means for determining whether the first participant user voluntarily or involuntarily terminated participation in the conference call based on the type of device the first participant user was operating during participation in the conference call.

73. A server for providing a conference call, comprising:

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means for detecting a conference call event previously configured by the subscriber user that indicates that a conference call should be established between a subscriber user and participant users;

means for collecting contact information for the participant users;

means for providing a message including the contact information to a conference server that is configured to instruct a bridge to establish a conference call between the users by calling the participant users using the contact information included in the message; and

means for receiving a response message from the conference server including information associated with the conference call and at least one participant user.

74. The system of claim 73, wherein the means for detecting a conference call event comprises:

means for scanning a calendar application to determine whether the subscriber user has scheduled a conference call.

75. The system of claim 73, wherein the means for collecting contact information comprises:

means for determining the participant users from the calendar application;

means for accessing an address book associated with the subscriber user for collecting the contact information for the participant users; and

means for generating the message using the collected contact information.

76. The system of claim 73, wherein the means for providing a message comprises:

means for generating the message using the collected contact information and instructions associated with a time for commencing a conference call previously scheduled by the subscribed user,

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wherein the message is used by the conference server to generate a second message for instructing the bridge to establish the conference call.

77. The system of claim 73, wherein the response message is provided by the bridge when attempting to establish communications with the participant users over a voice network.

78. The system of claim 73, wherein the response message reflects at least one of:

- the least one participant user declining the call;
- the at least one participant user not answering the call;
- the at least one participant user accepting the call; and
- none of the participant users answering the call.

79. A system for participating in a conference call automatically established by at least one communication entity, the system comprising:

means for scheduling a conference call for a predetermined date using a calendar application, wherein the means for scheduling identifies participant users that are to participate in the conference call; and

means for receiving, on the predetermined date, an indication that a conference call has been configured in accordance with the scheduled conference call,

means for receiving a indication that at least one participant user has either joined or declined to join the conference call, and

means for receiving, during the conference call, an indication that another participant user has ended communications with the conference call when that participant user terminates a connection with a bridge that establishes the conference call in accordance with instructions provided by a server that executes the calendar application.

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80. A conference bridge for providing a conference call, comprising:

means for receiving a message from a first server including instructions for establishing a conference call between participant users and a subscriber user that previously scheduled the conference call with a second server, wherein the second server automatically initiates configuration of the conference call based on the subscriber user's schedule by providing to the first server contact information for the participant users and the message includes the contact information;

means for calling the participant users using the contact information included in the message;

means for receiving a response from the participant users, wherein the responses reflect whether the participant user has answered, accepted, not answered, or declined to accept the call; and

means for establishing a conference call between the users based on the received responses.

81. A system for providing conference calls, comprising:

a first server for detecting a conference call event previously configured by a subscriber user and generating a first message based on the event, wherein the first message includes telephone numbers for a plurality of participant users identified by the subscriber user;

a second server for receiving the first message from the first server and generating a second message based on the first message, wherein the second message includes instructions for configuring a conference call and the telephone numbers; and

a conference bridge for receiving the second message, extracting the telephone numbers from the second message, and calling the participant users using the telephone numbers, and establishing a conference call between the participant users and the subscriber user,

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wherein the subscriber user accesses the first server to schedule conference calls for future dates and the first server, second server, and conference bridge automatically attempt to establish the conference calls when the future dates arrive.

82. A computer-readable medium including instructions for performing, when executed by a processor, a method for providing a conference call, comprising:

- detecting a conference call event that was previously configured by a subscriber user;
- identifying participant users associated with the conference call event;
- contacting the participant users;
- receiving responses from the participant users; and
- establishing a conference call between the participant users and the subscriber user based on the received responses,

wherein at least one of the detecting and identifying steps is performed without user intervention.

83. A computer-readable medium including instructions for performing, when executed by a processor, a method for providing a conference call comprising:

- detecting a conference call event previously configured by the subscriber user that indicates that a conference call should be established between a subscriber user and participant users;
- collecting contact information for the participant users;
- providing a message including the contact information to a conference server that is configured to instruct a bridge to establish a conference call between the users by calling the participant users using the contact information included in the message; and
- receiving a response message from the conference server including information associated with the conference call and at least one participant user.

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84. A computer-readable medium including instructions for performing, when executed by a processor, a method for participating in a conference call automatically established by at least one communication entity, the method comprising:

- scheduling a conference call for a predetermined time using a calendar application, wherein scheduling includes identifying participant users that are to participate in the conference call; and

- on the predetermined time:

- receiving an indication that a conference call has been configured in accordance with the scheduled conference call,

- receiving a indication that at least one participant user has either joined or declined to join the conference call, and

- during the conference call, receiving an indication that another participant user has ended communications with the conference call when that participant user terminates a connection with a bridge that establishes the conference call in accordance with instructions provided by a server that executes the calendar application.

85. A computer-readable medium including instructions for performing, when executed by a processor, a method for providing a conference call, including:

- receiving a message from a first server including instructions for establishing a conference call between participant users and a subscriber user that previously scheduled the conference call with a second server, wherein the second server automatically initiates configuration of the conference call based on the subscriber user's schedule by providing to the first server contact information for the participant users and the message includes the contact information;

- calling the participant users using the contact information included in the message;

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receiving responses from the participant users, wherein the responses reflect whether the participant user has answered, accepted, not answered, or declined to accept the call; and

establishing a conference call between the users based on the received responses.

86. A method for providing a conference call, comprising:

detecting a request to establish a conference call initiated by a subscriber user;

identifying participant users identified in the request;

contacting the participant users;

receiving a response from the participant users; and

establishing a conference call between the participant users and the subscriber user based on the received responses,

wherein at least one of the contacting, receiving, and establishing steps is performed without user intervention.

87. A method for providing a conference call, the method performed by a server comprising:

detecting a conference call event configured by a subscriber user that indicates a conference call to be established between the subscriber user and participant users;

collecting contact information for the participant users;

providing a message including the contact information to a conference server that is configured to instruct a bridge to establish the conference call between the users by calling the participant users using the contact information included in the message; and

receiving a response message from the conference server including information associated with the conference call and at least one participant user.

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88. A method for participating in a conference call automatically established by at least one or more communication entities, the method comprising:

receiving an indication that a conference call has been configured in response to a user's request to establish the conference call,

receiving a indication that at least one participant user has either joined or declined to join the conference call, and

during the conference call, receiving an indication that another participant user has ended communications with the conference call when that participant user terminates a connection with a bridge that establishes the conference call in accordance with instructions provided by a server.

89. A method for providing a conference call, comprising:

receiving a message from a first server including instructions for establishing a conference call between participant users and a subscriber user that schedules the conference call with a second server, wherein the second server automatically initiates configuration of the conference call by providing to the first server contact information for the participant users and the message includes the contact information;

calling the participant users using the contact information included in the message;

receiving a response from the participant users, wherein the response reflects whether the participant user has answered, accepted, not answered, or declined to accept the call; and

establishing a conference call between the users based on the response received from the participant users.

90. A system for providing a conference call, comprising:

means for detecting a conference call event that is configured by a subscriber user;

means for identifying participant users associated with the conference call event;

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means for contacting the participant users;
means for receiving a response from the participant users; and
means for establishing a conference call between the participant users and the subscriber user based on the received responses,
wherein at least one of the means for contacting, means for receiving, and means for identifying perform their functions without user intervention.

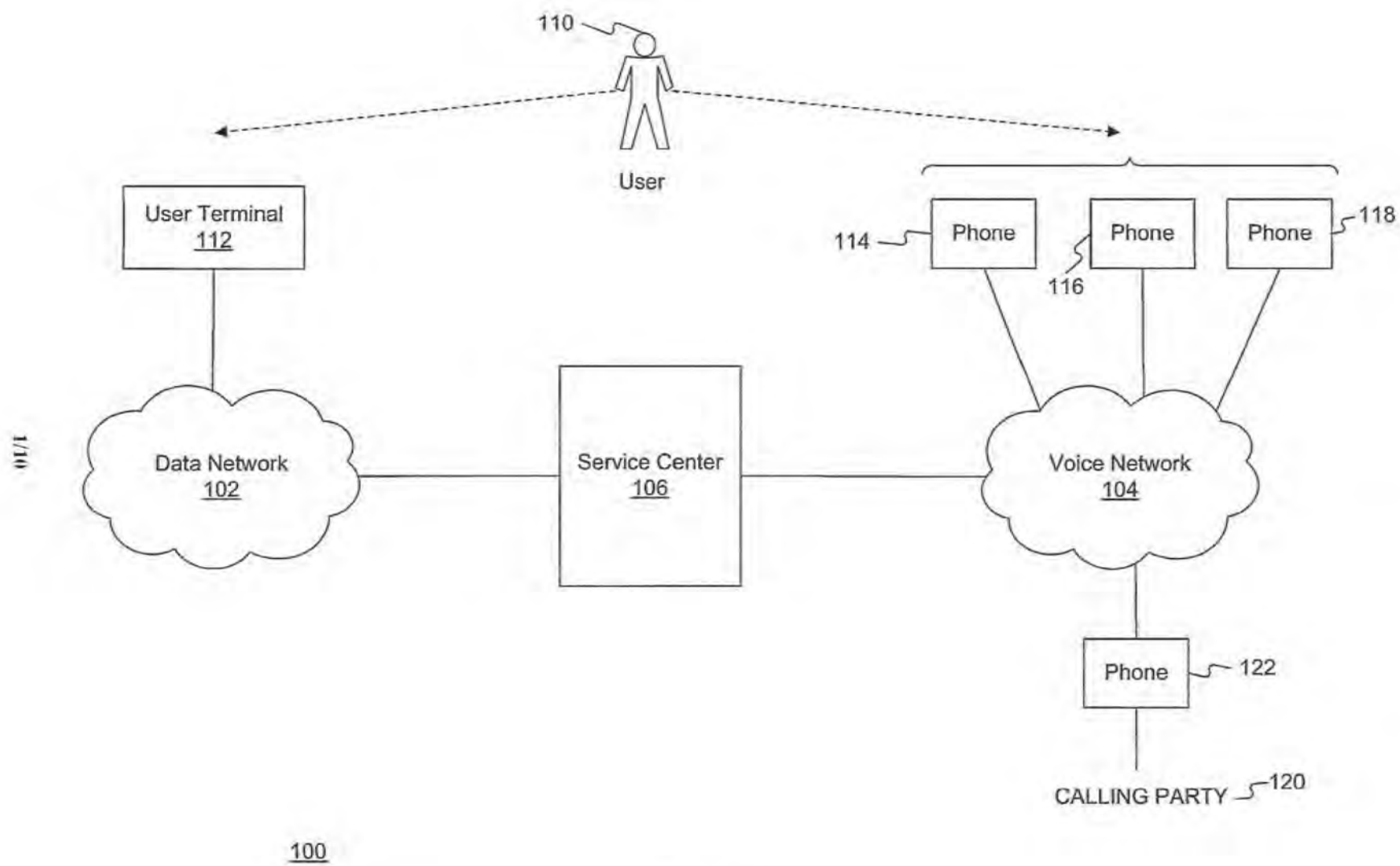


Figure 1

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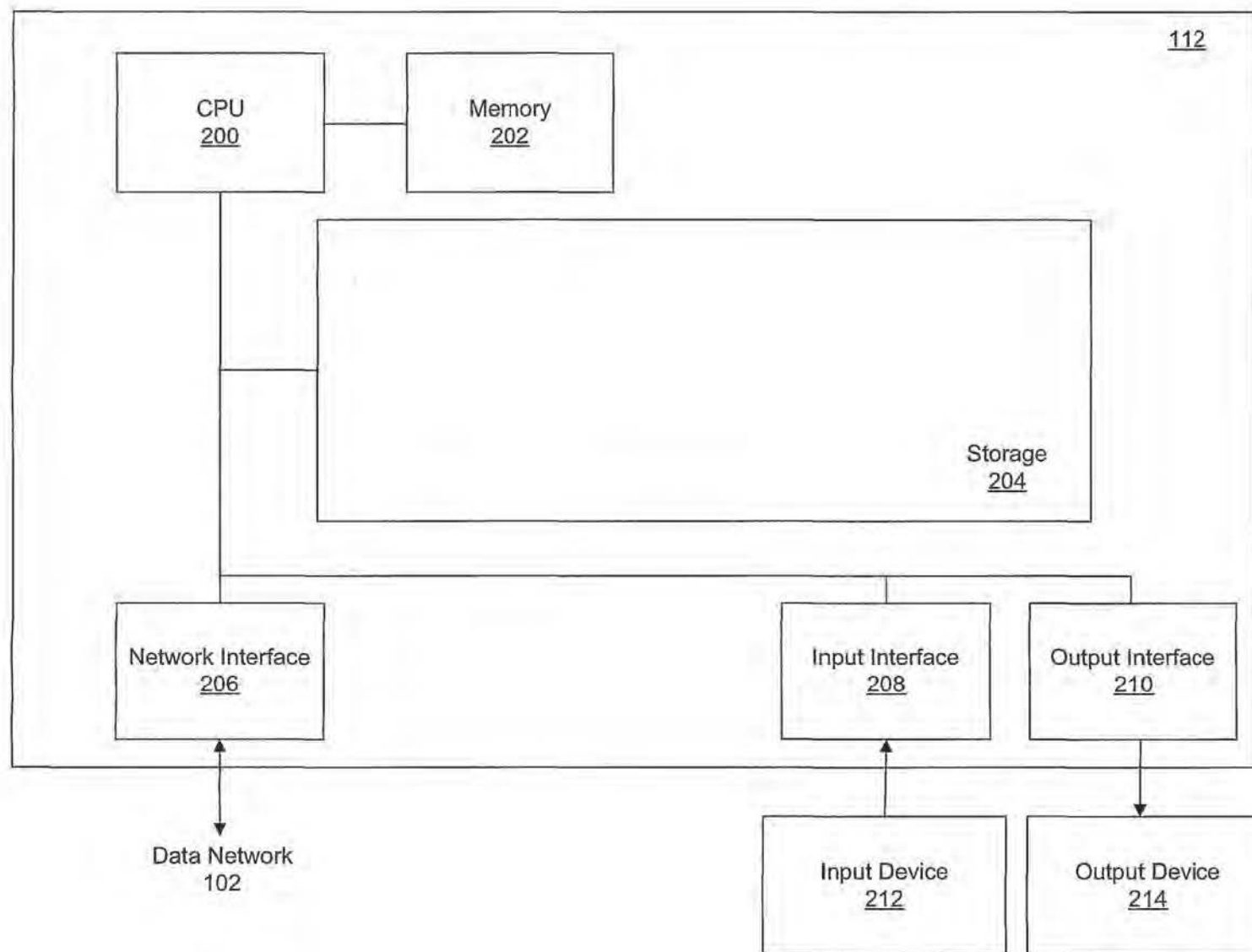


Figure 2

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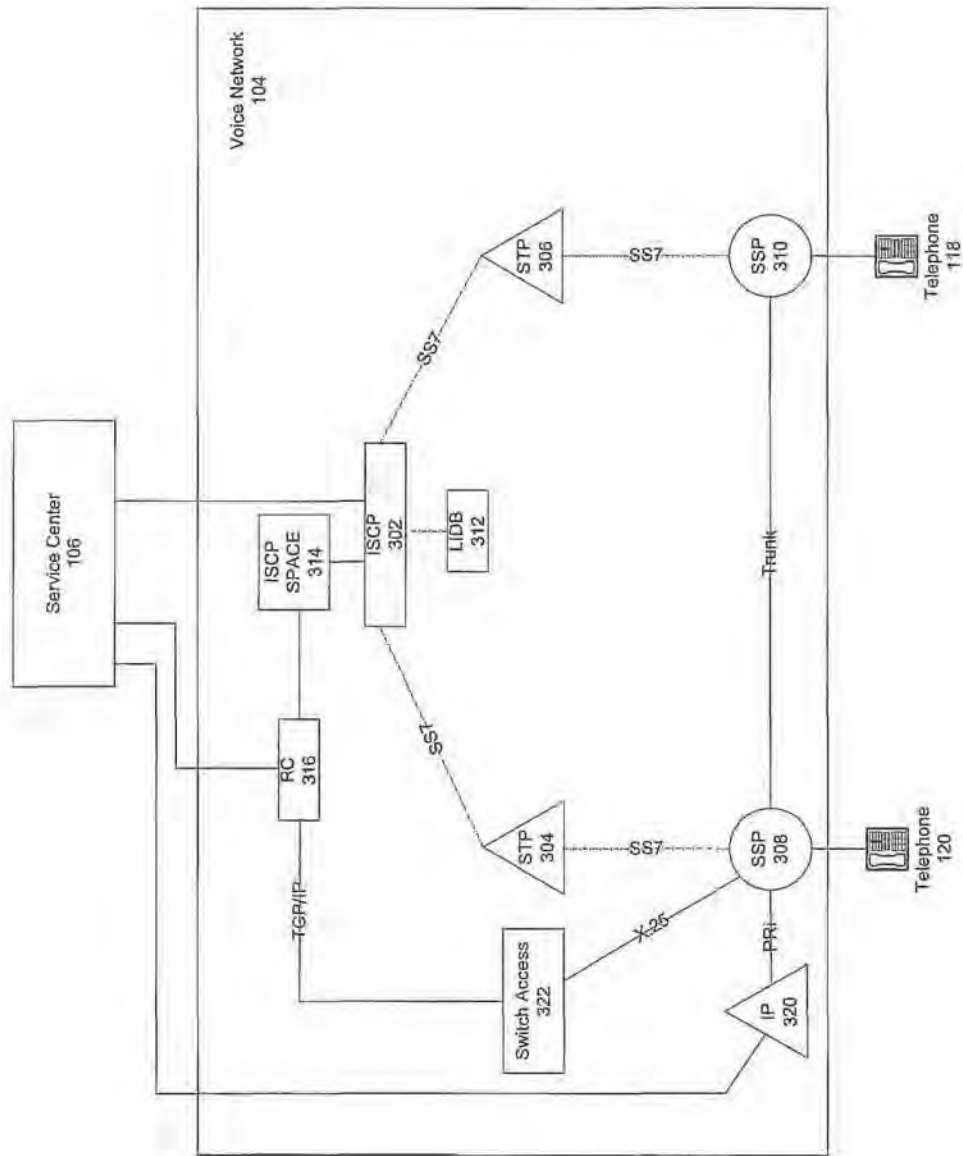


Figure 3

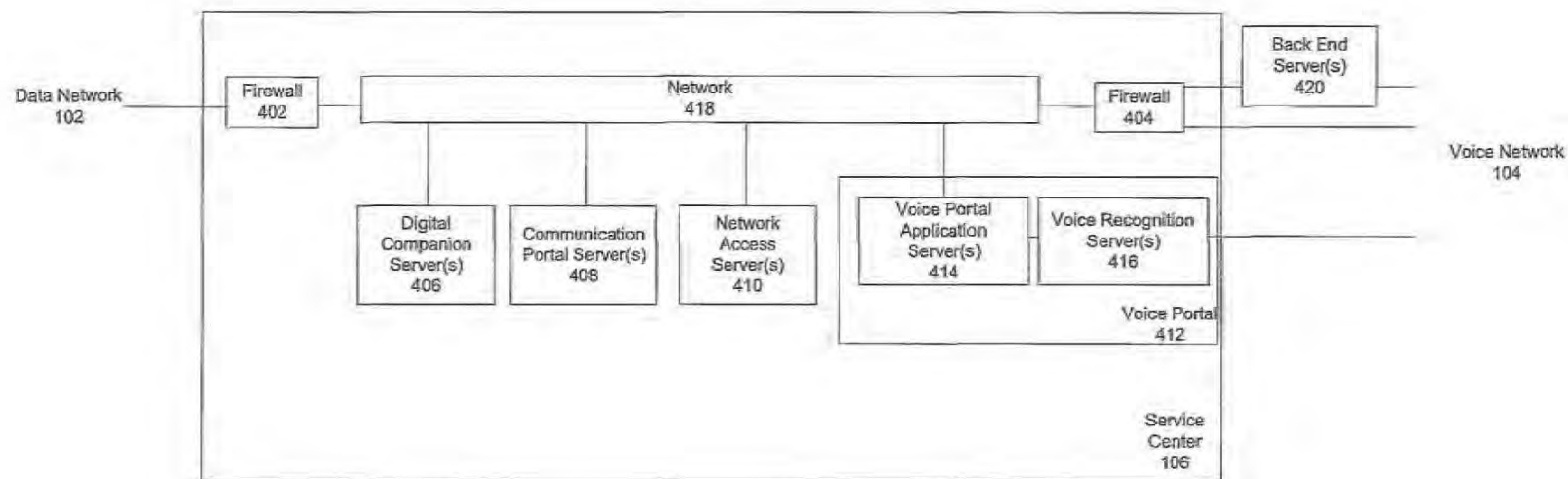


Figure 4

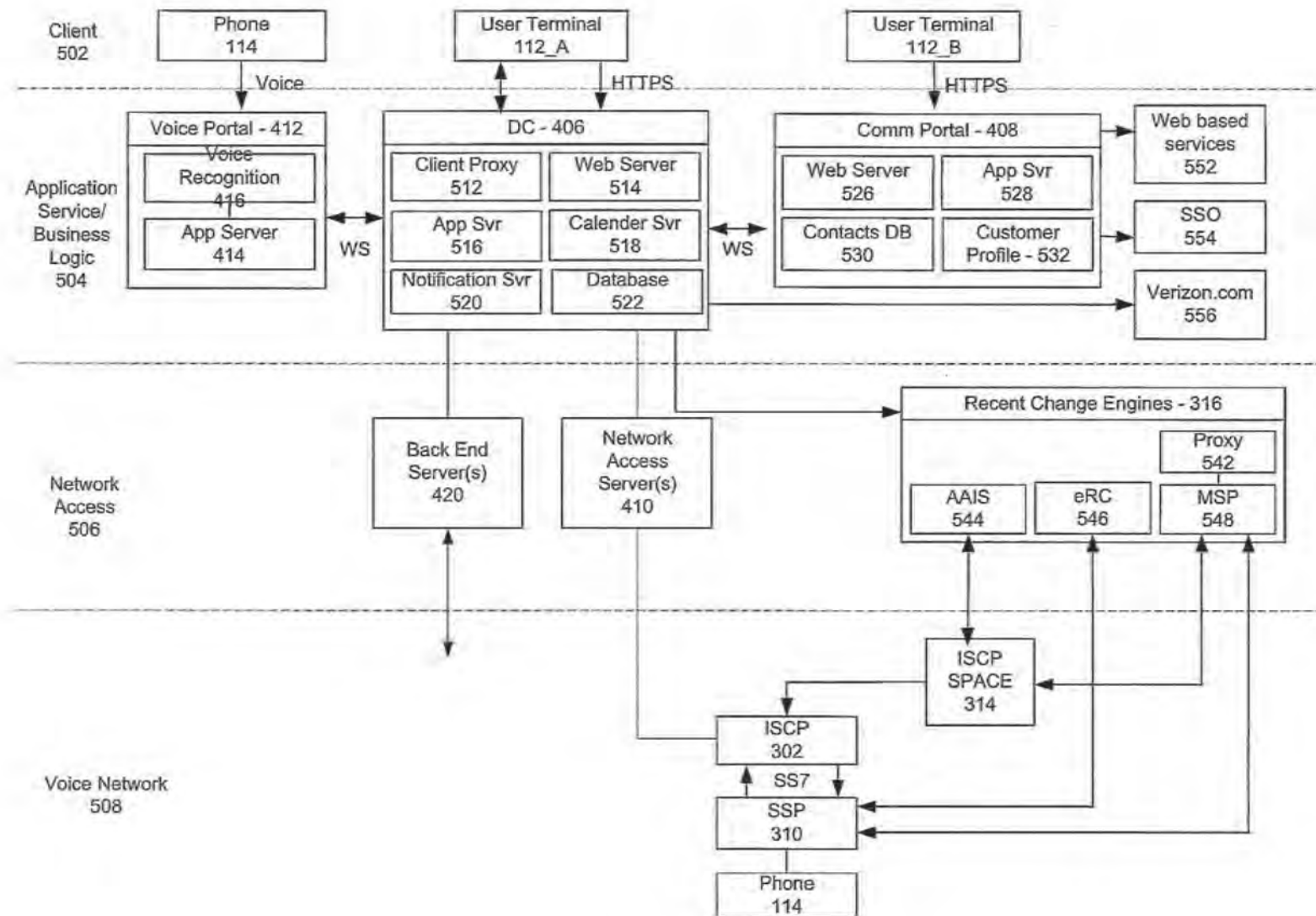


Figure 5

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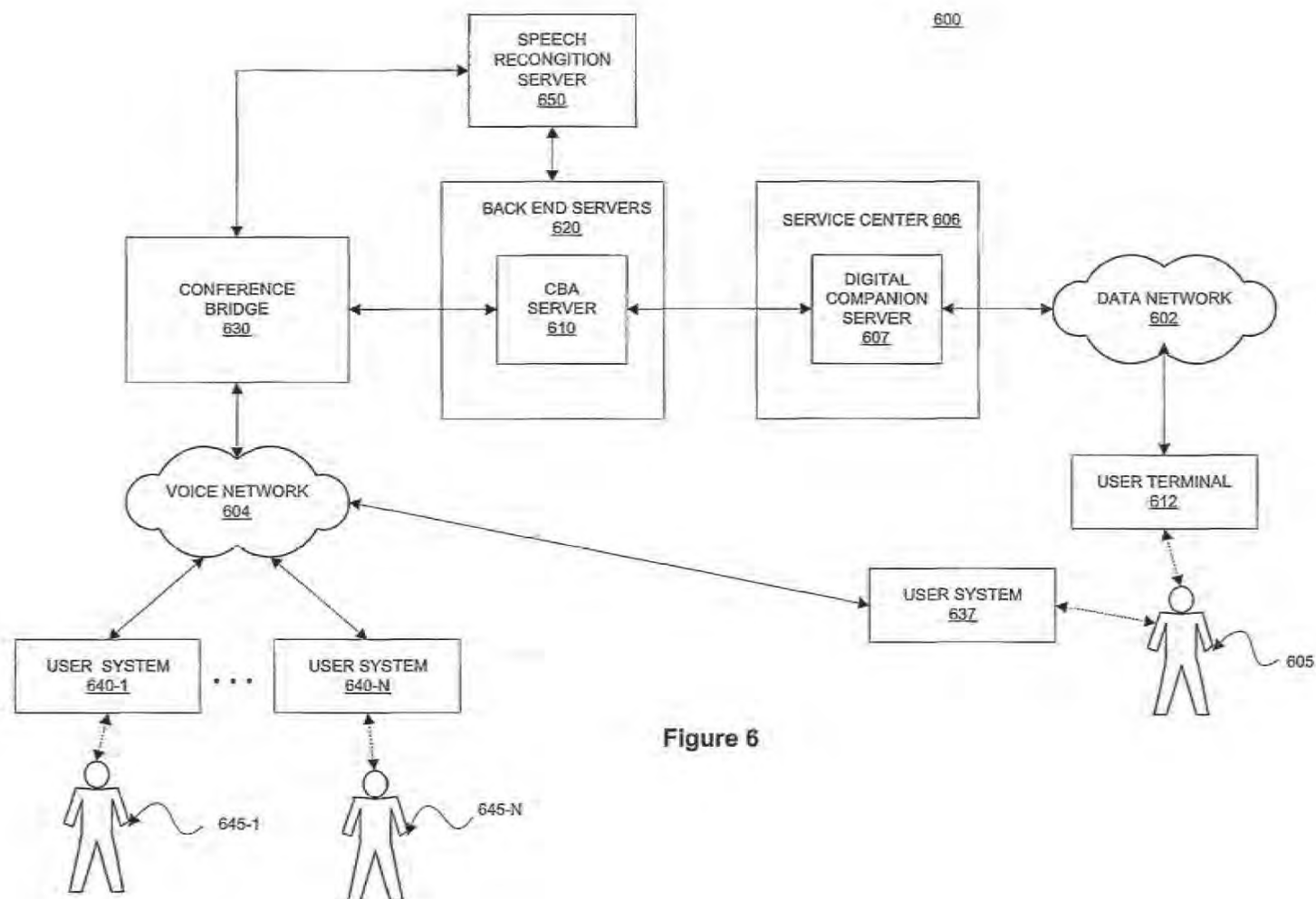


Figure 6

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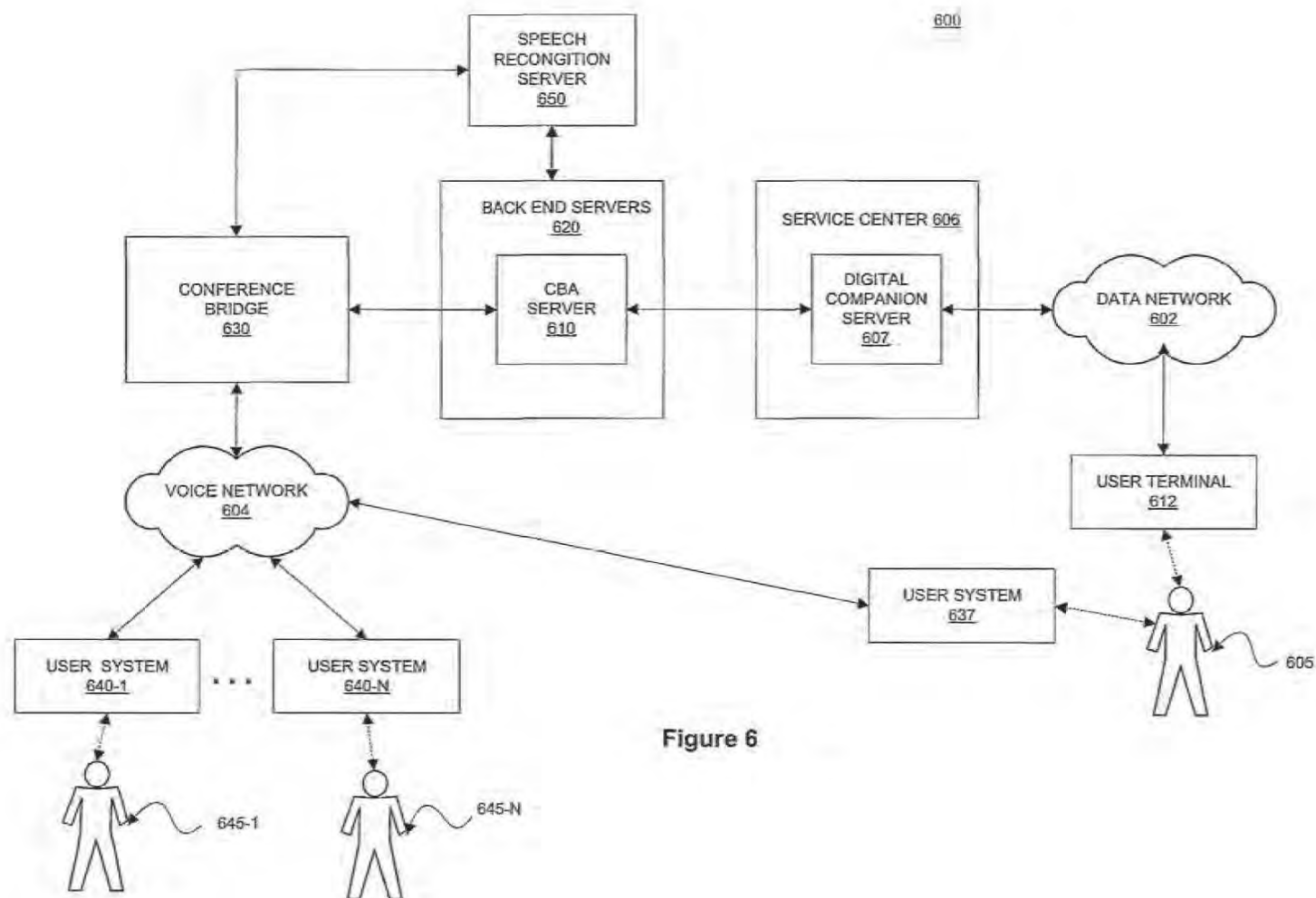


Figure 6

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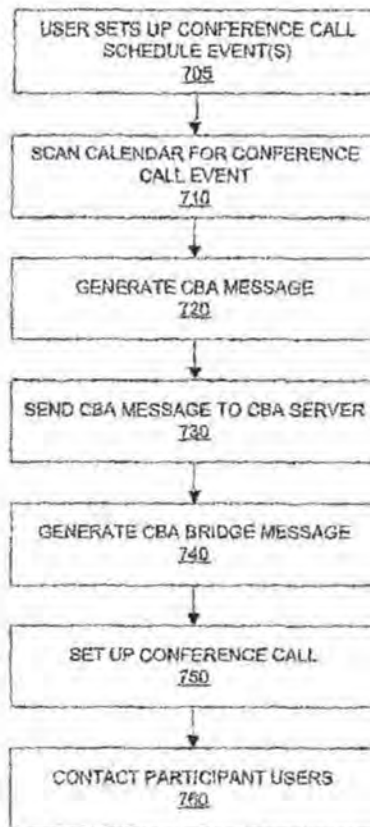


Figure 7

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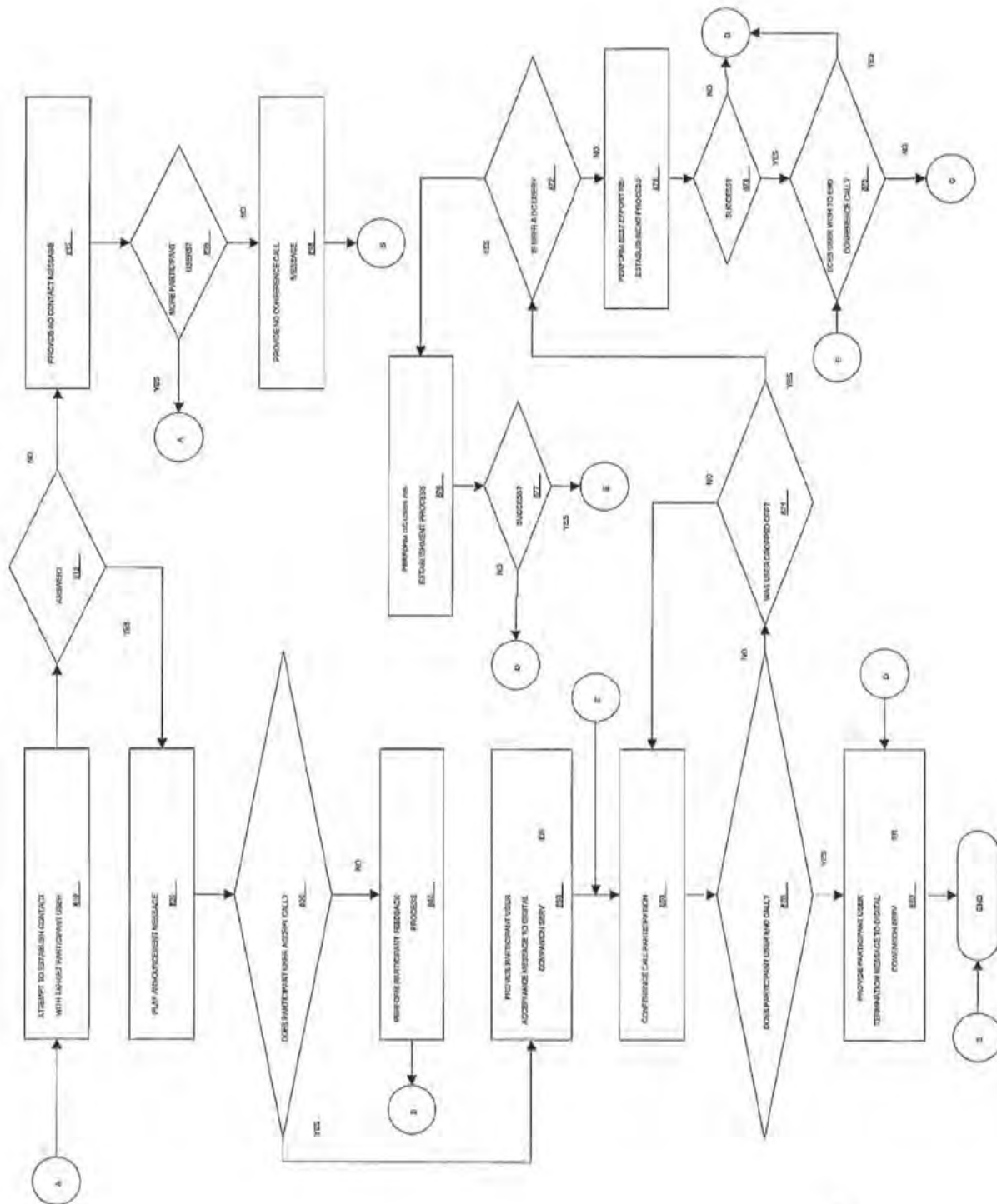


Figure 8

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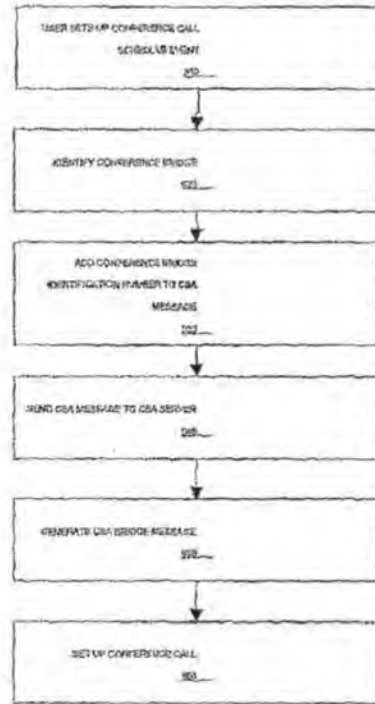


Figure 9

INTERNATIONAL SEARCH REPORT

International application No.

PCT/US03/38135

A. CLASSIFICATION OF SUBJECT MATTER

IPC(7) : HO4L 12/16; HO4M 3/24

US CL : 370/260, 261; 709/204; 379/202.01

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)

U.S. : 370/260, 261; 709/204; 379/202.01

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
Please See Continuation Sheet**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X --- Y	US 6,275,575 A (WU) 14 August 2001 (14.08.2001), column 10, lines 5-55.	1-17, 22-31, 33-57, 62-90
Y	US 5,710,591 A (BRUNO et al) 20 January 1998 (20.01.1998) column 9, lines 34-68.	18-21, 58-61 18-21, 58-61
A	US 5,631,904 A (FITSER et al) 20 May 1997 (20.05.1997), see figures 2 & 3.	1-90
A	US 5,673,080 A (BIGGS et al) 30 September 1997 (30.09.1997), See fig. 8.	1-90
A	US 5,907,324 A (LARSON et al) 25 May 1999 (25.05.1999), See figures 8-13.	1-90

☐ Further documents are listed in the continuation of Box C.☐ 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

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Date of the actual completion of the international search

01 April 2004 (01.04.2004)

Date of mailing of the international search report

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Authorized officer

Andy Lee

Telephone No. 703-305-3900

Form PCT/ISA/210 (second sheet) (July 1998)

INTERNATIONAL SEARCH REPORT

PCT/US03/38135

Continuation of B. FIELDS SEARCHED Item 3:

EAST:

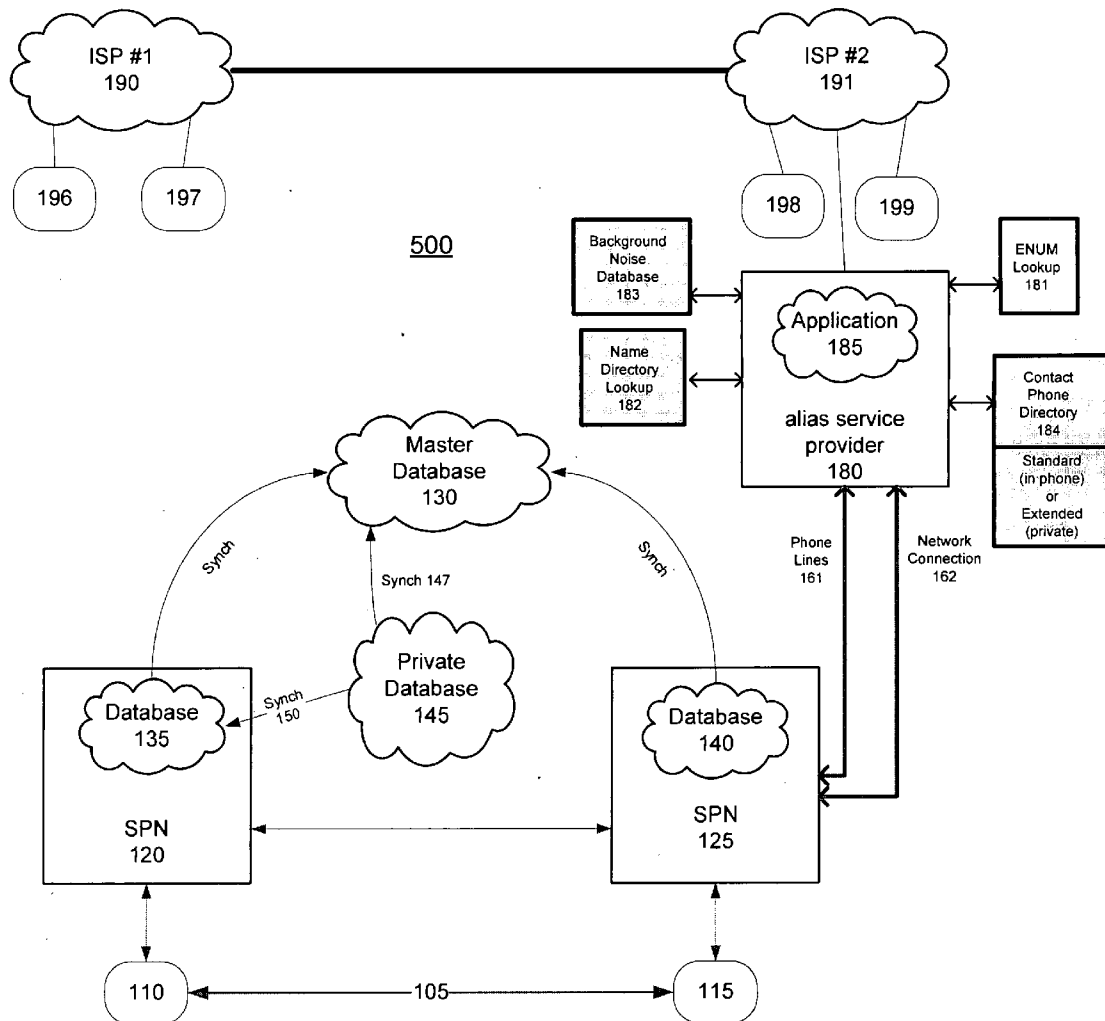
search terms: conferencing, scheduling, calendar, address book, audio text conversion, options

Form PCT/ISA/210 (second sheet) (July 1998)

Exhibit R

(19) **United States**(12) **Patent Application Publication**
Black et al.(10) **Pub. No.: US 2006/0140200 A1**(43) **Pub. Date: Jun. 29, 2006**(54) **USER-CONTROLLED
TELECOMMUNICATIONS SYSTEM****Publication Classification**(51) **Int. Cl.**
H04L 12/56 (2006.01)(52) **U.S. Cl.** **370/401**(76) Inventors: **Jeffery D. Black**, Menlo Park, CA
(US); **Alexander D. Black**, Menlo
Park, CA (US); **John H. Todd**,
Portland, OR (US)Correspondence Address:
FISH & RICHARDSON, PC
P.O. BOX 1022
MINNEAPOLIS, MN 55440-1022 (US)(21) Appl. No.: **11/286,310**(22) Filed: **Nov. 23, 2005****Related U.S. Application Data**(60) Provisional application No. 60/631,002, filed on Nov.
24, 2004. Provisional application No. 60/718,620,
filed on Sep. 19, 2005.(57) **ABSTRACT**

Disclosed are systems and methods that enable an end user to control the parameters, behavior, and identity of network communications regardless of the limitations placed upon these controls by the end user's communications device or communications service provider. An intermediate media proxy resides on a telecommunications network and accepts inbound and outbound communications to and from a network communication device. The intermediate media proxy functions as an intermediary for network communications that occur between the network communications device and other network communications devices. In this manner, the intermediate media proxy can control various features of the communications independent of limitations enacted by the end user's communications device or communications service provider.



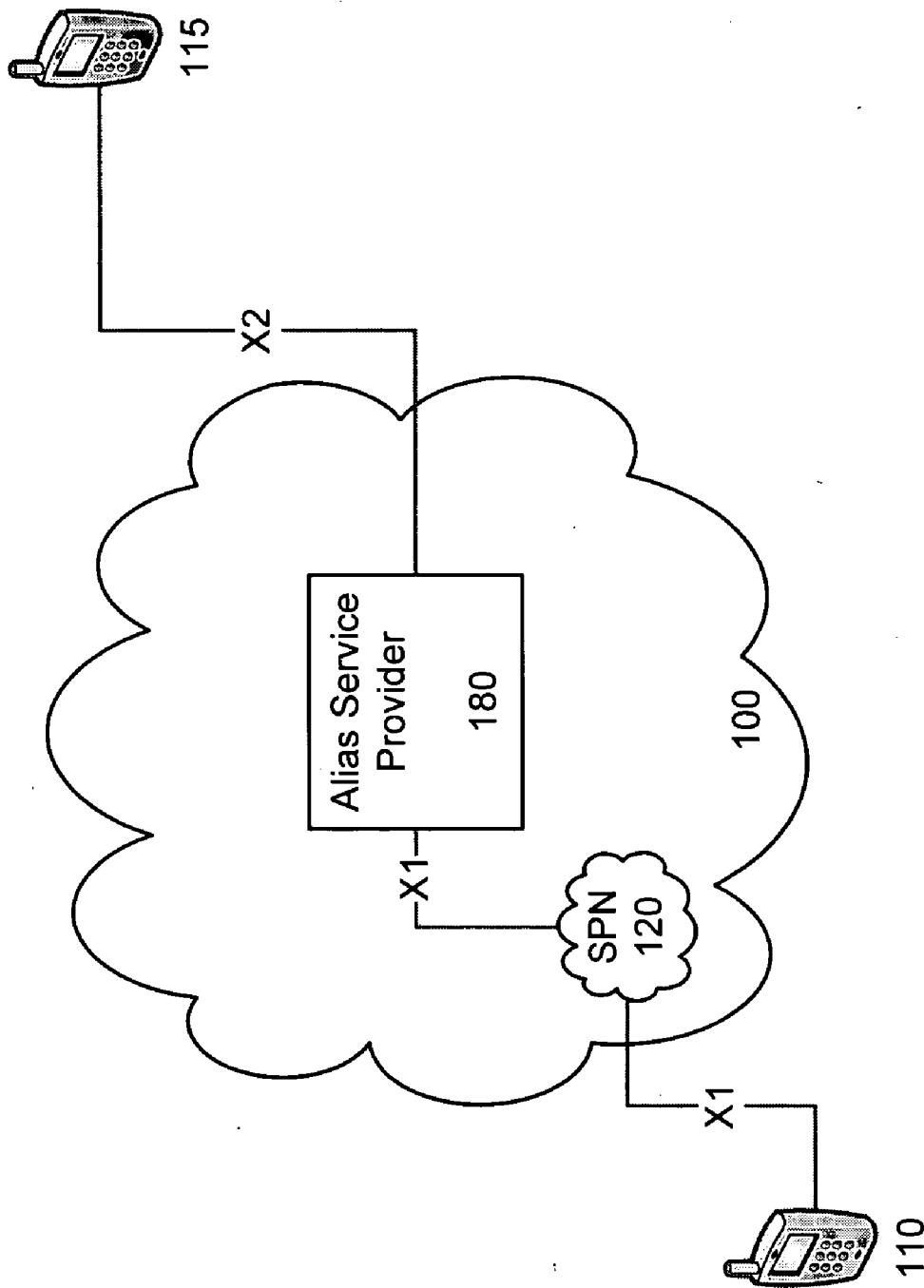


FIGURE 1A

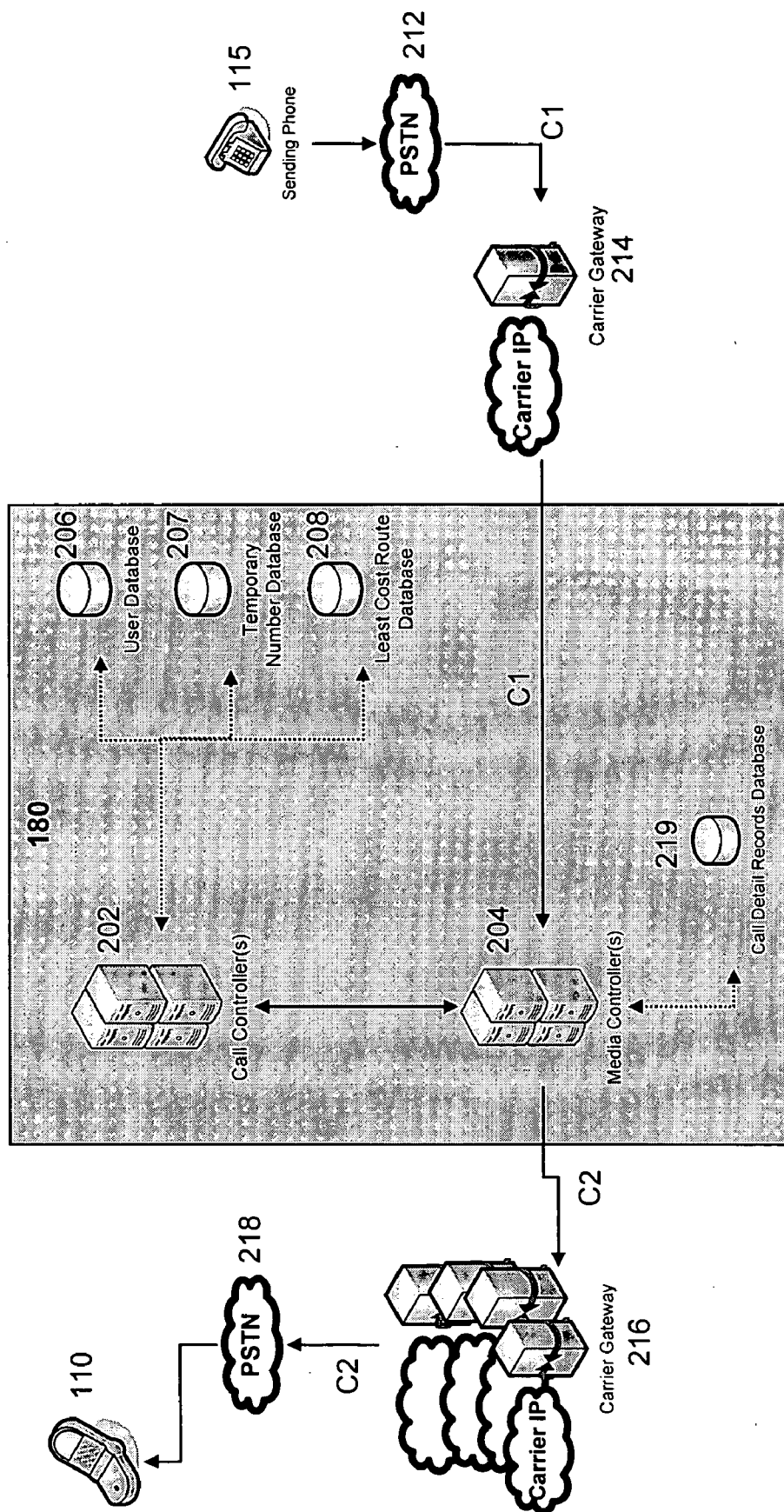


FIGURE 1B

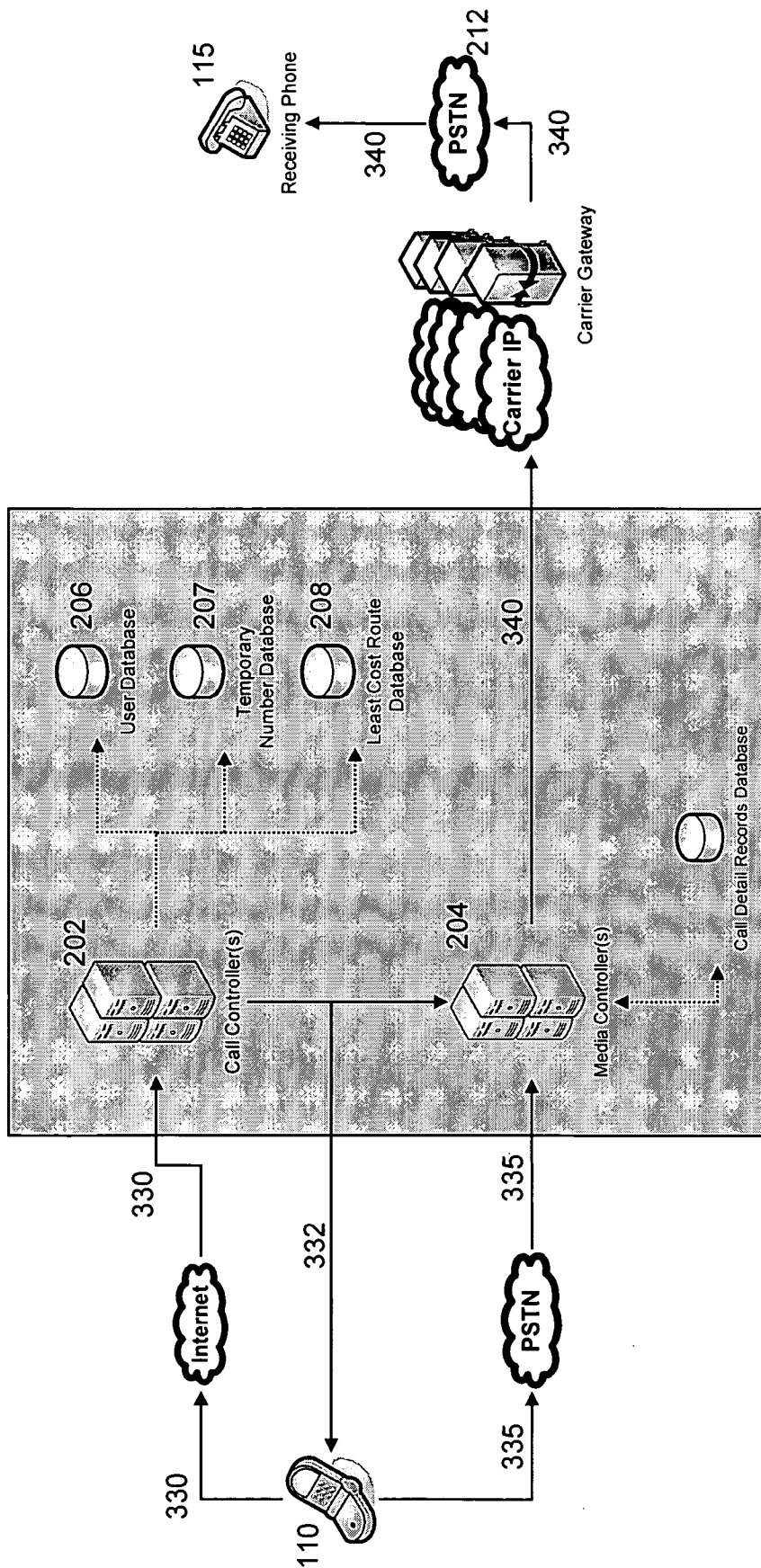


FIGURE 1C

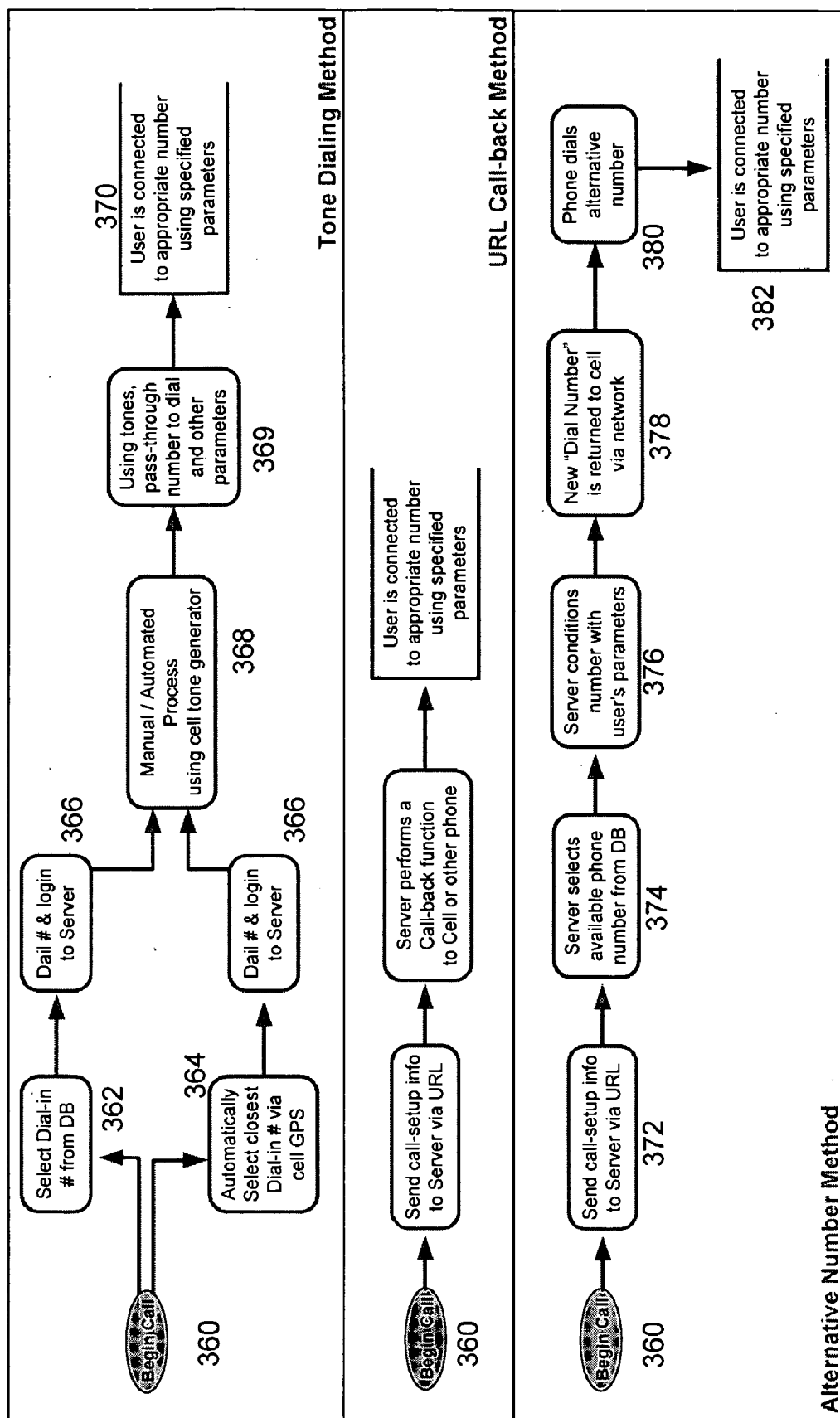


FIGURE 1D

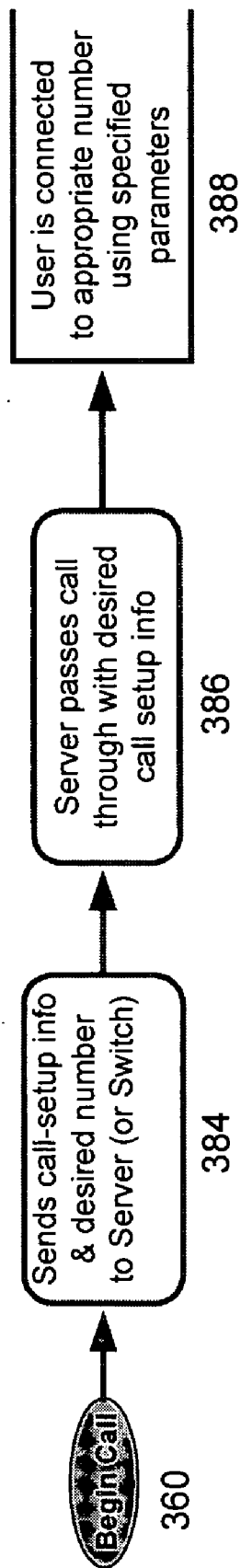


FIGURE 1E

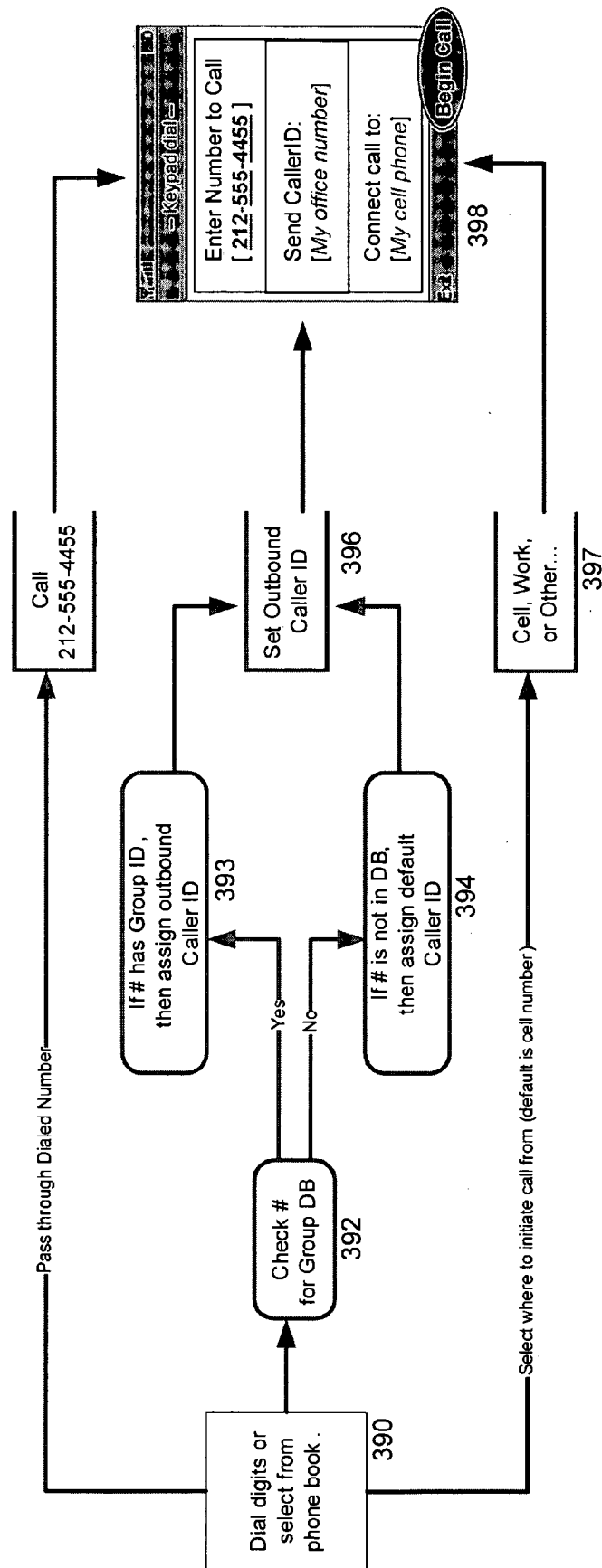


FIGURE 1F

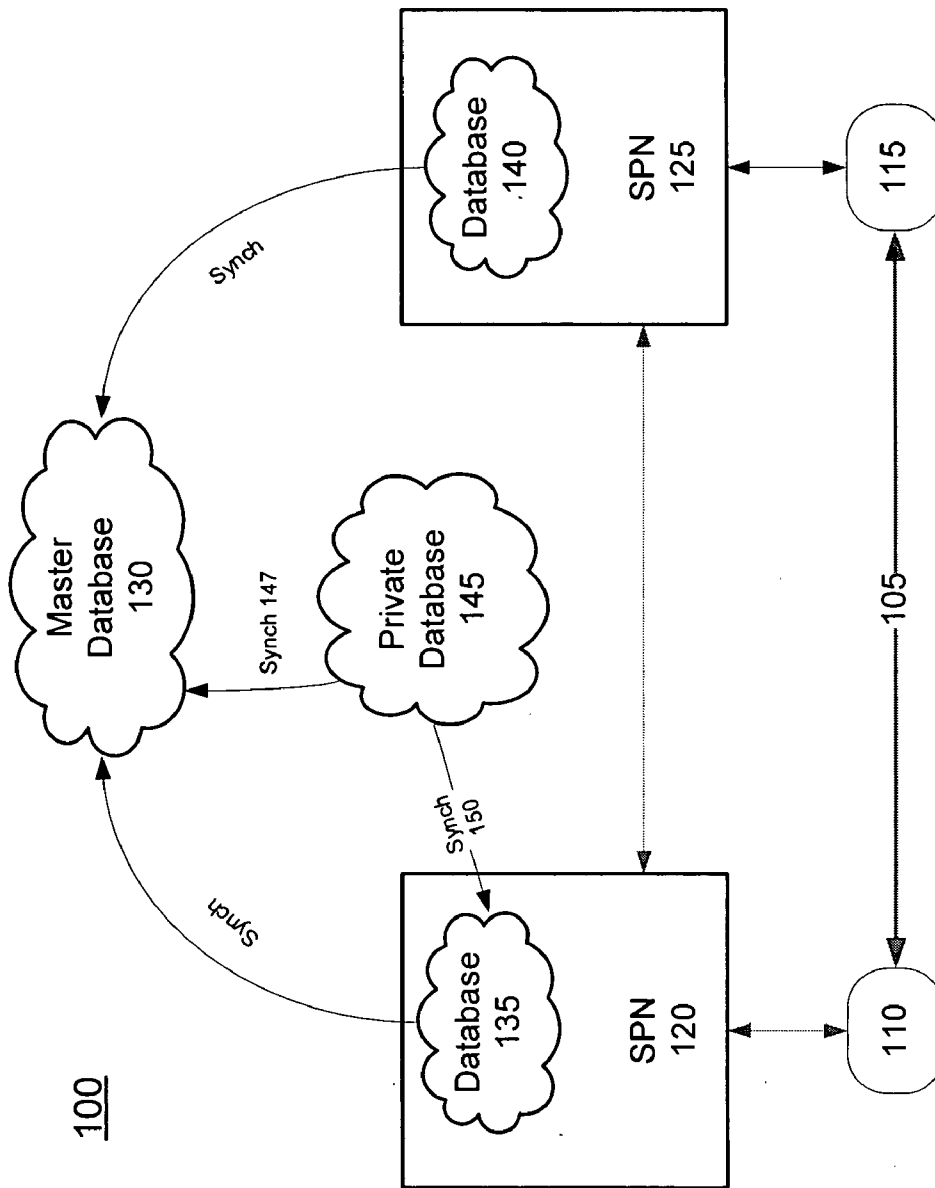


FIGURE 1G

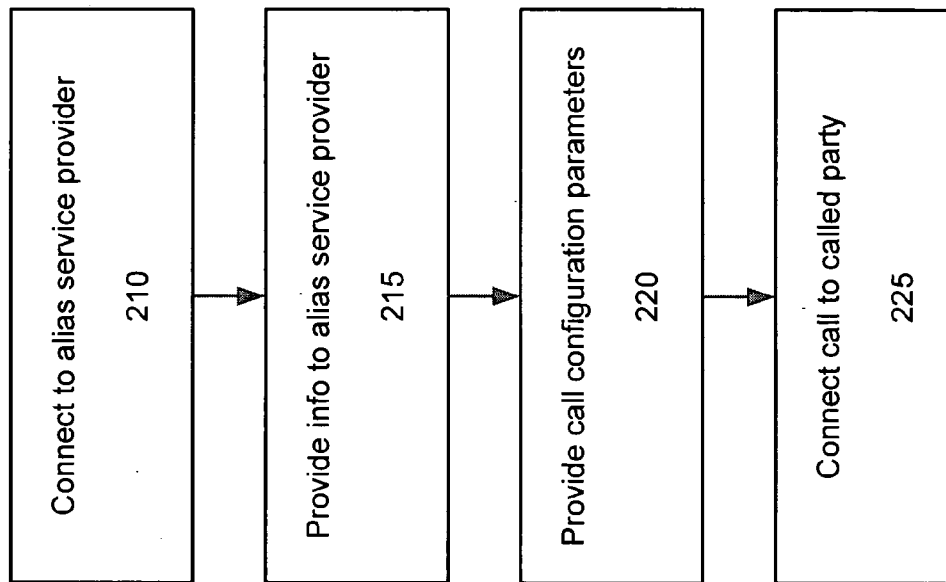


FIGURE 2

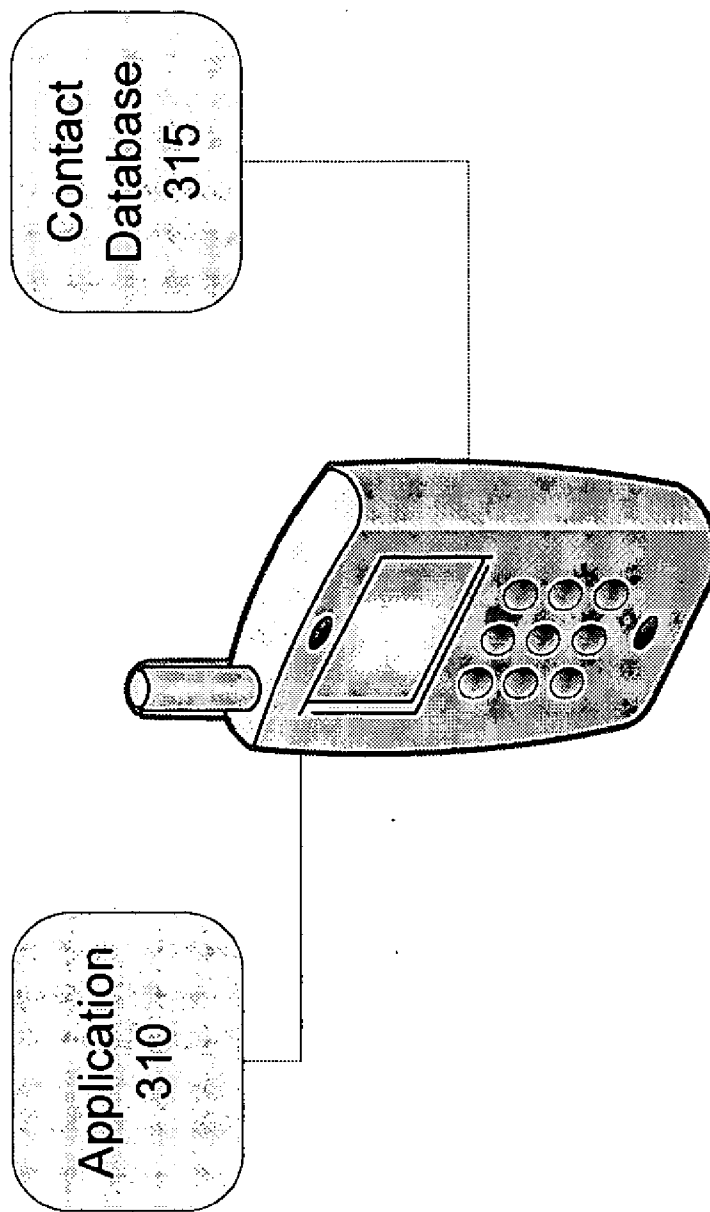


FIGURE 3A

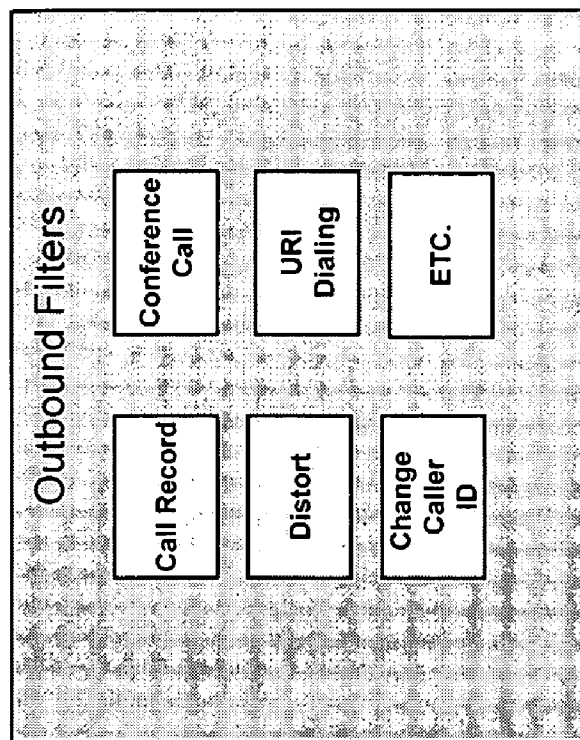


FIGURE 3C

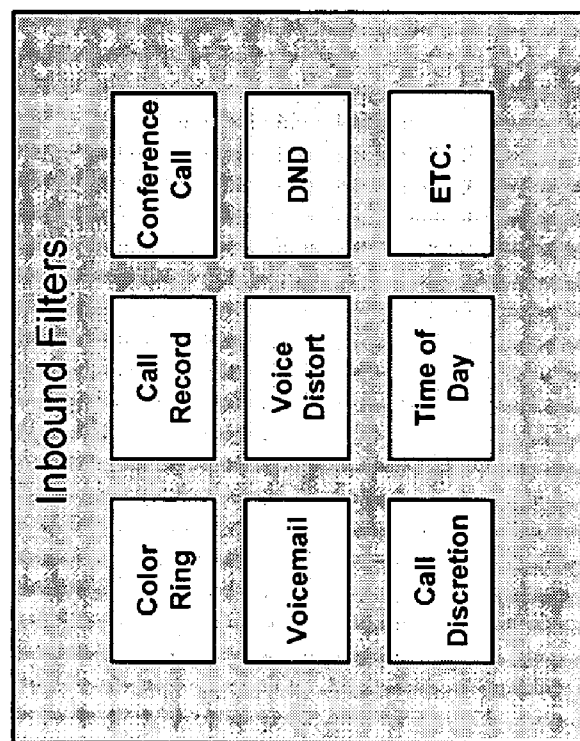
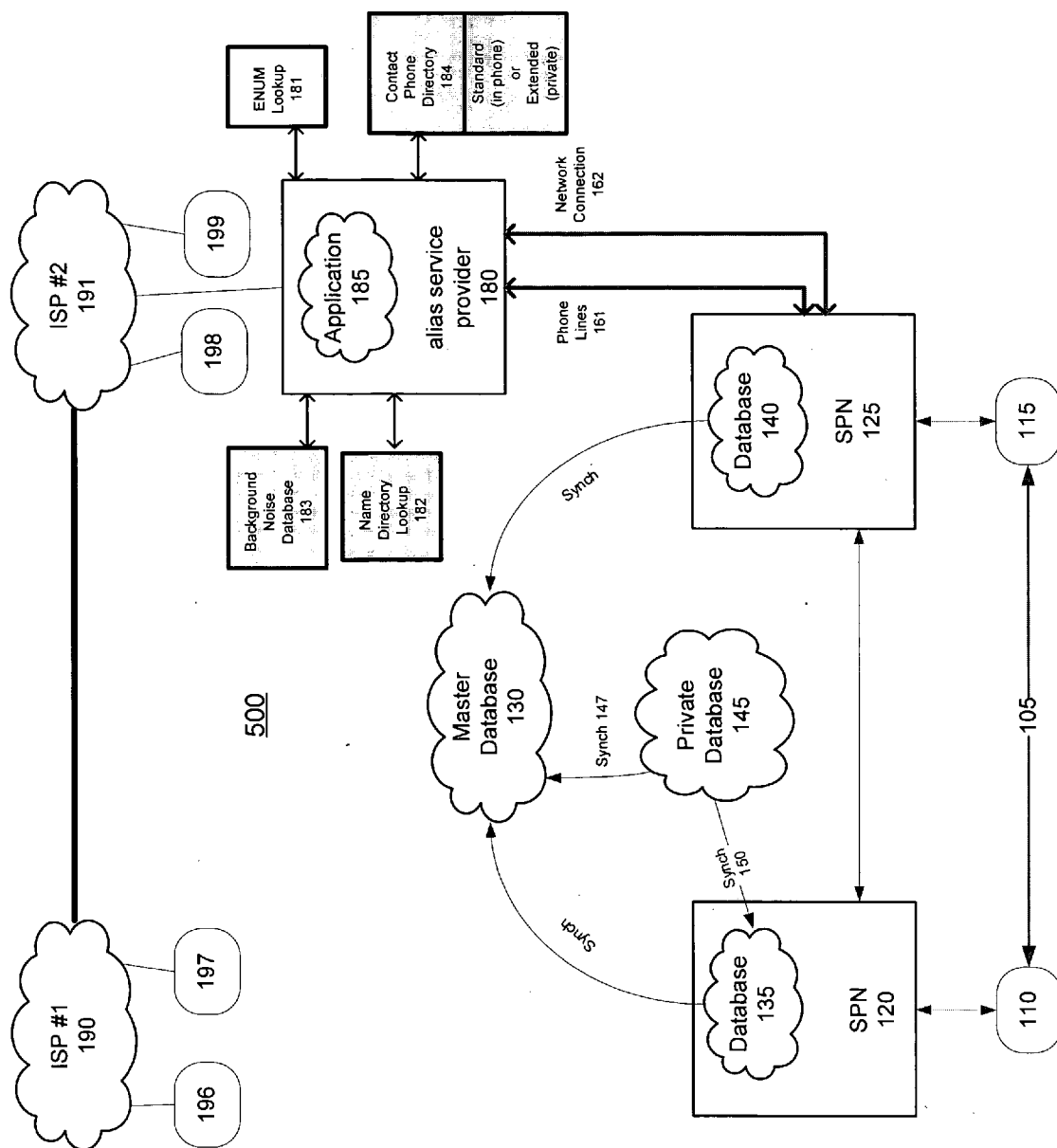


FIGURE 3B

FIGURE 4



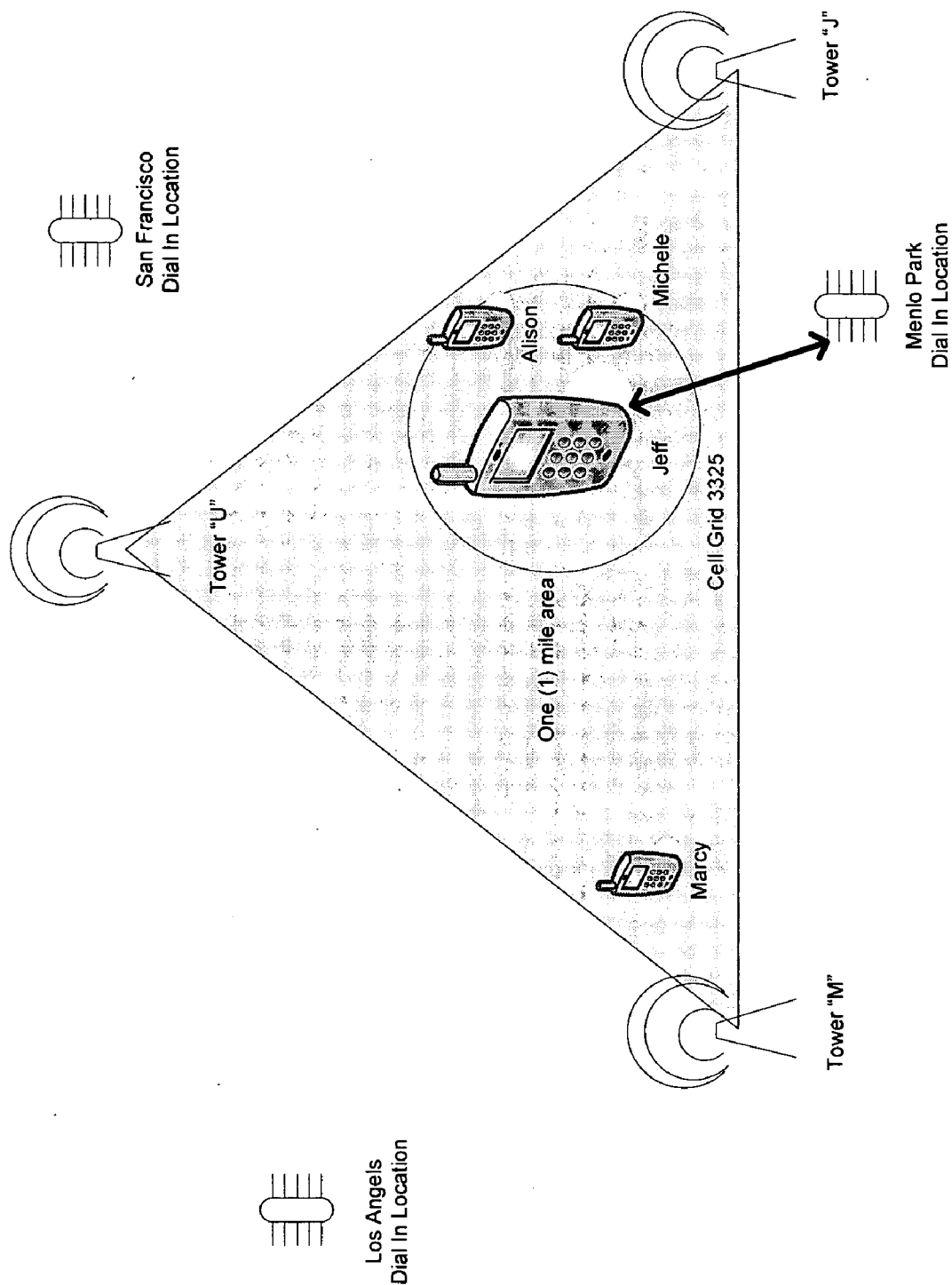


FIGURE 5

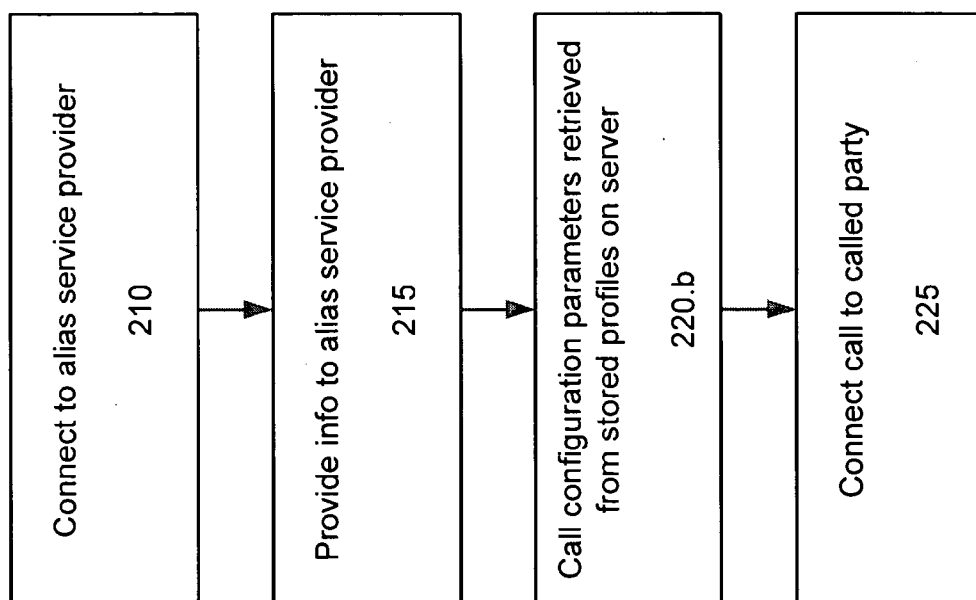


FIGURE 6

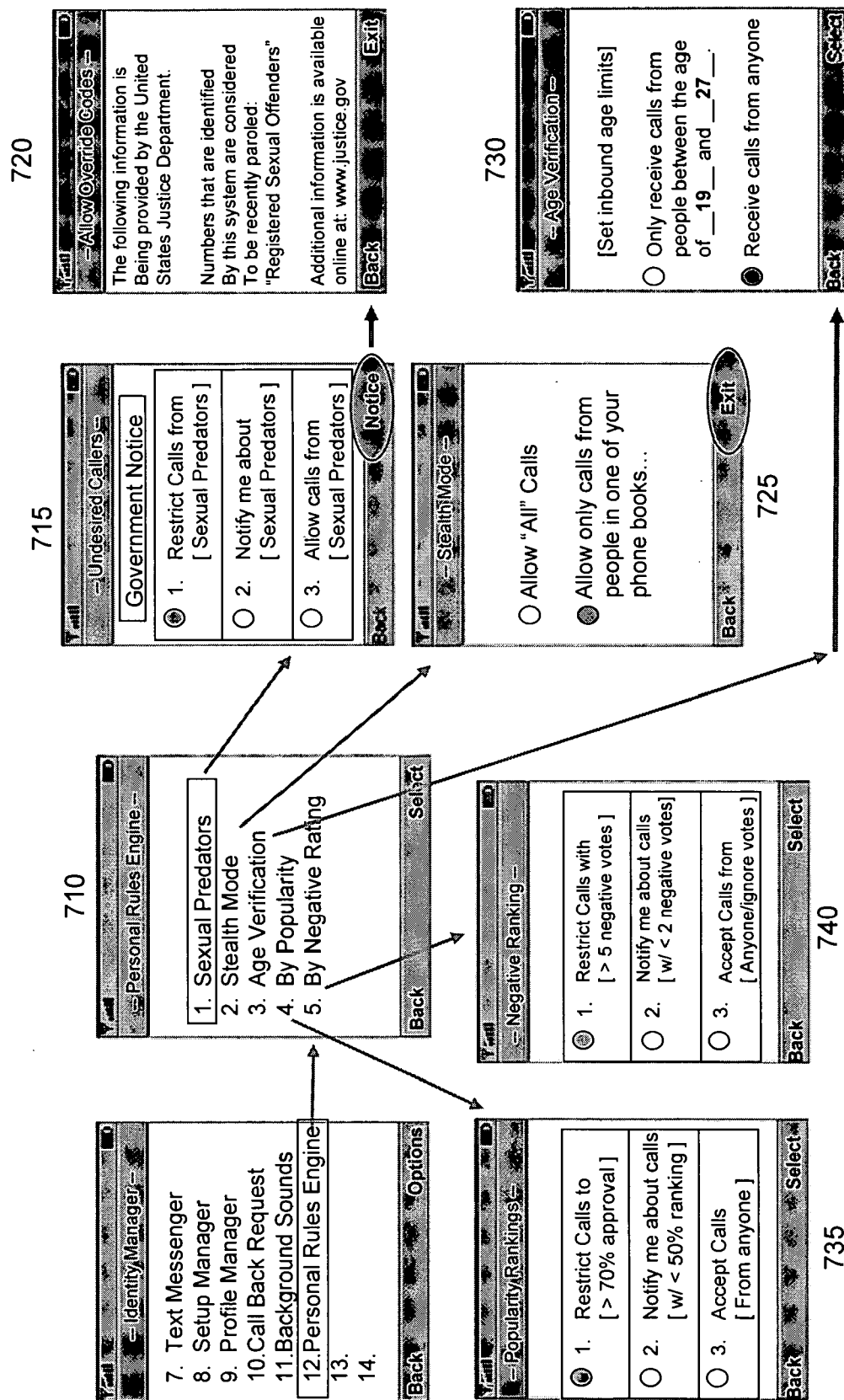


FIGURE 7A

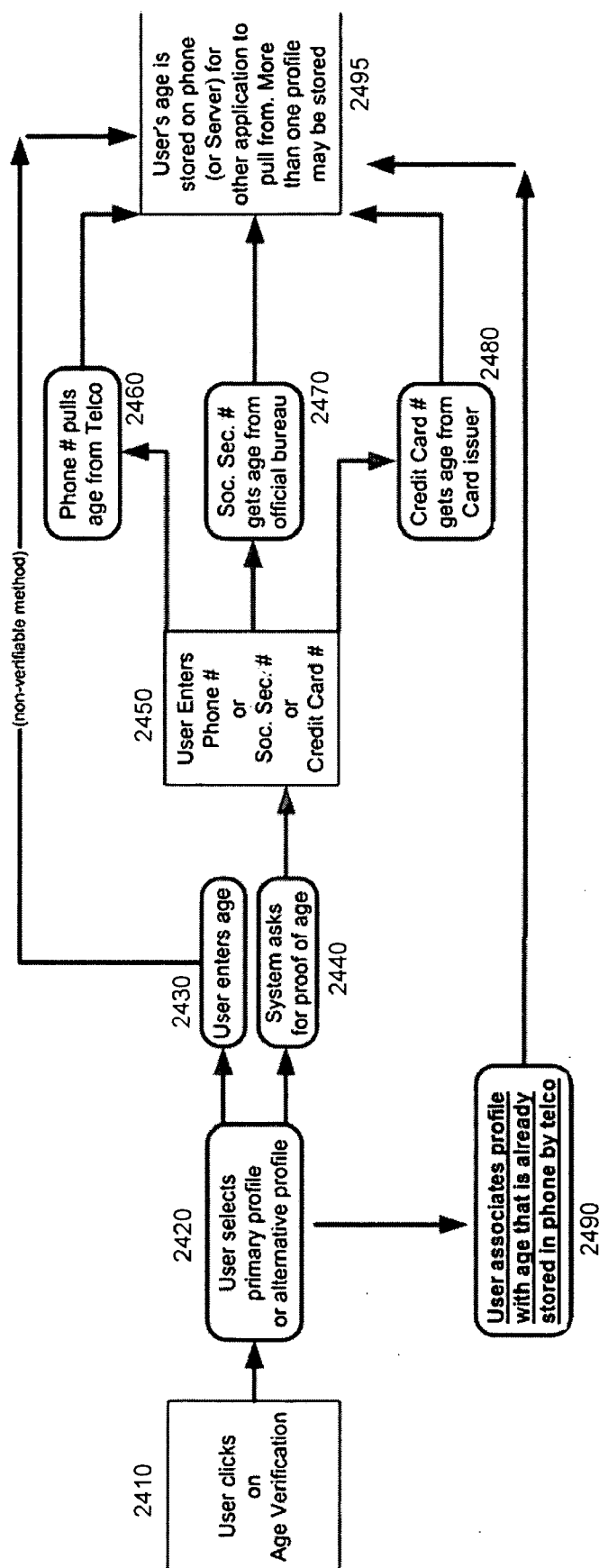


FIGURE 7B

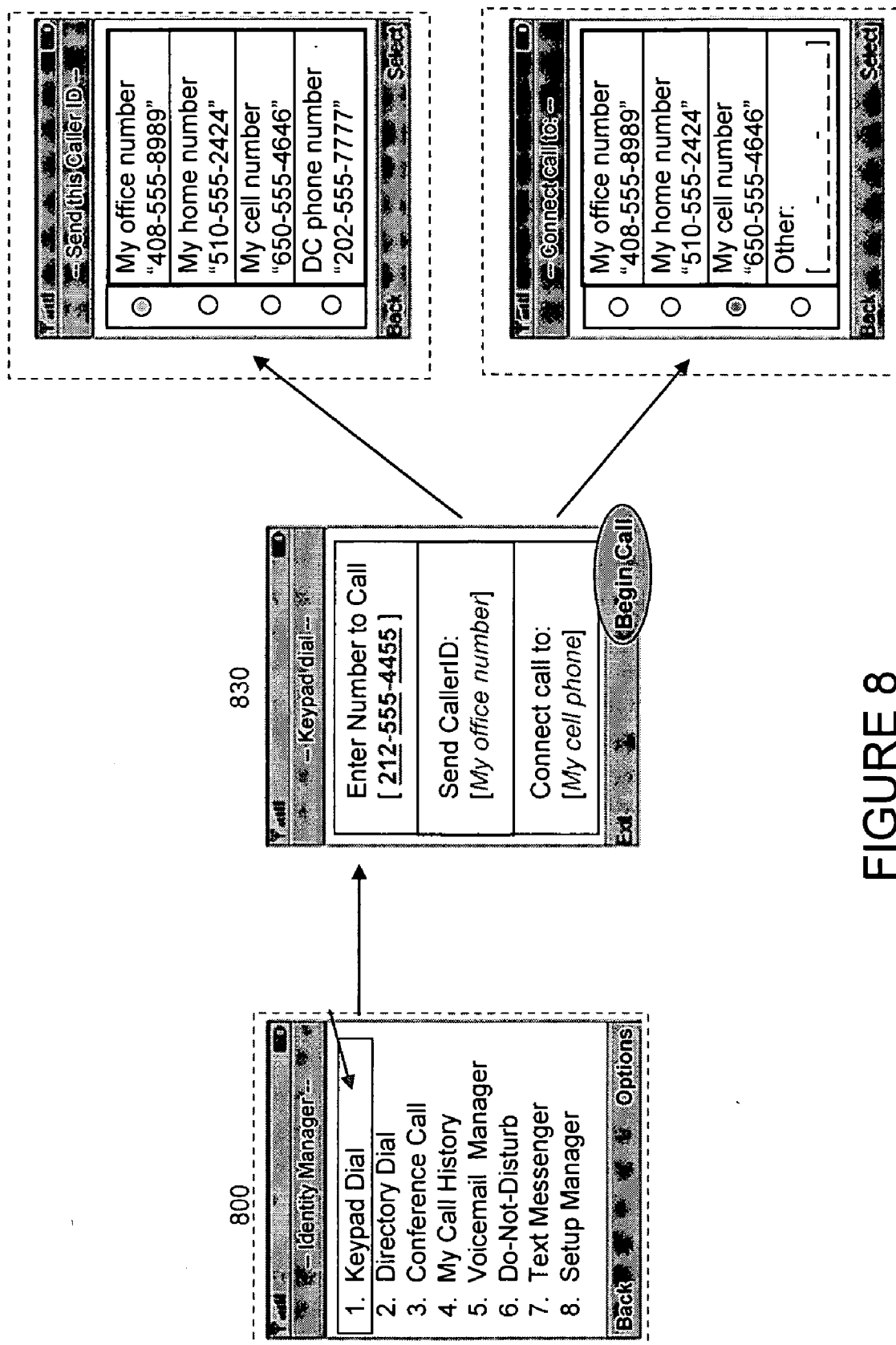


FIGURE 8

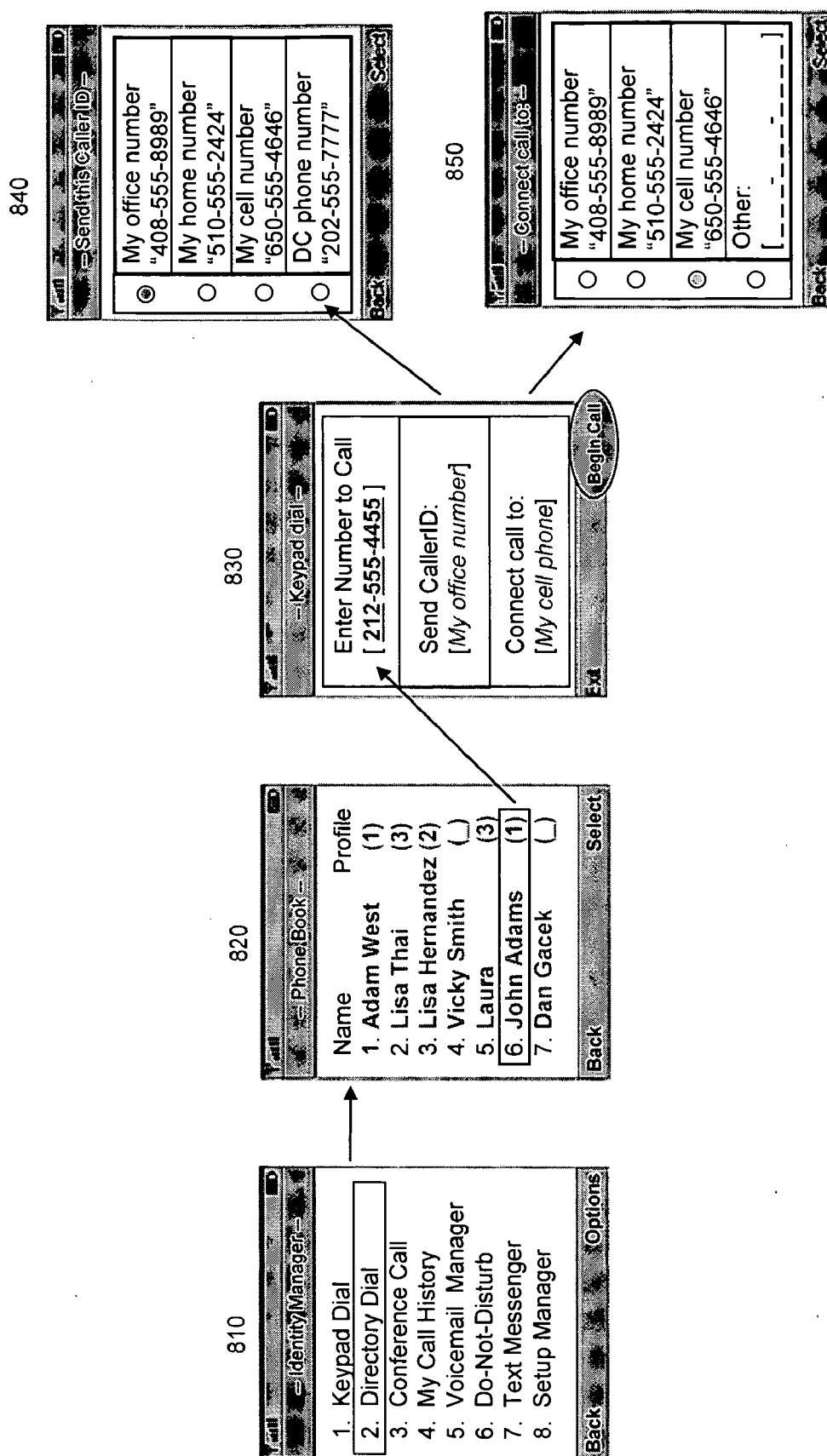


FIGURE 9

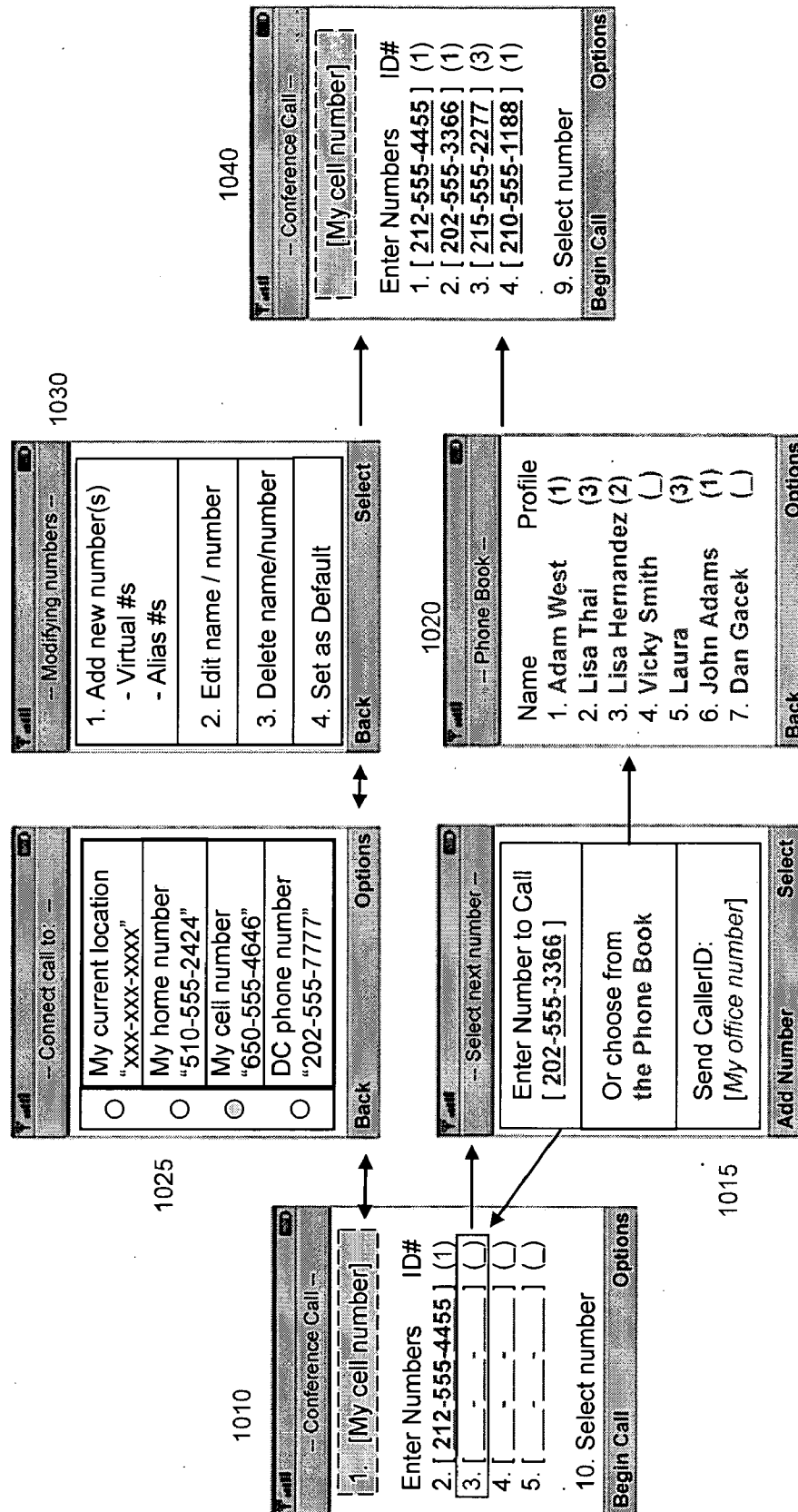


FIGURE 10

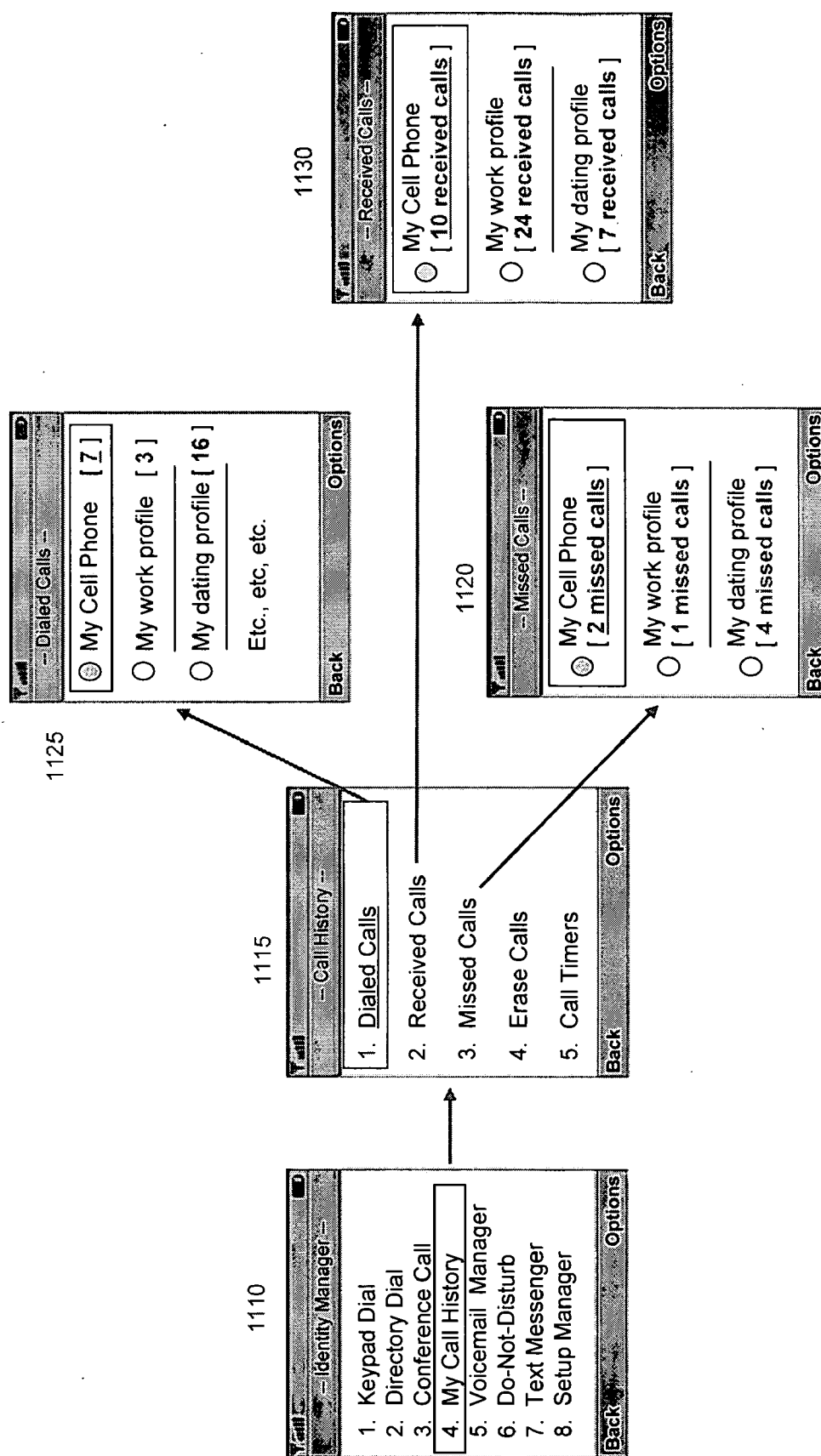


FIGURE 11

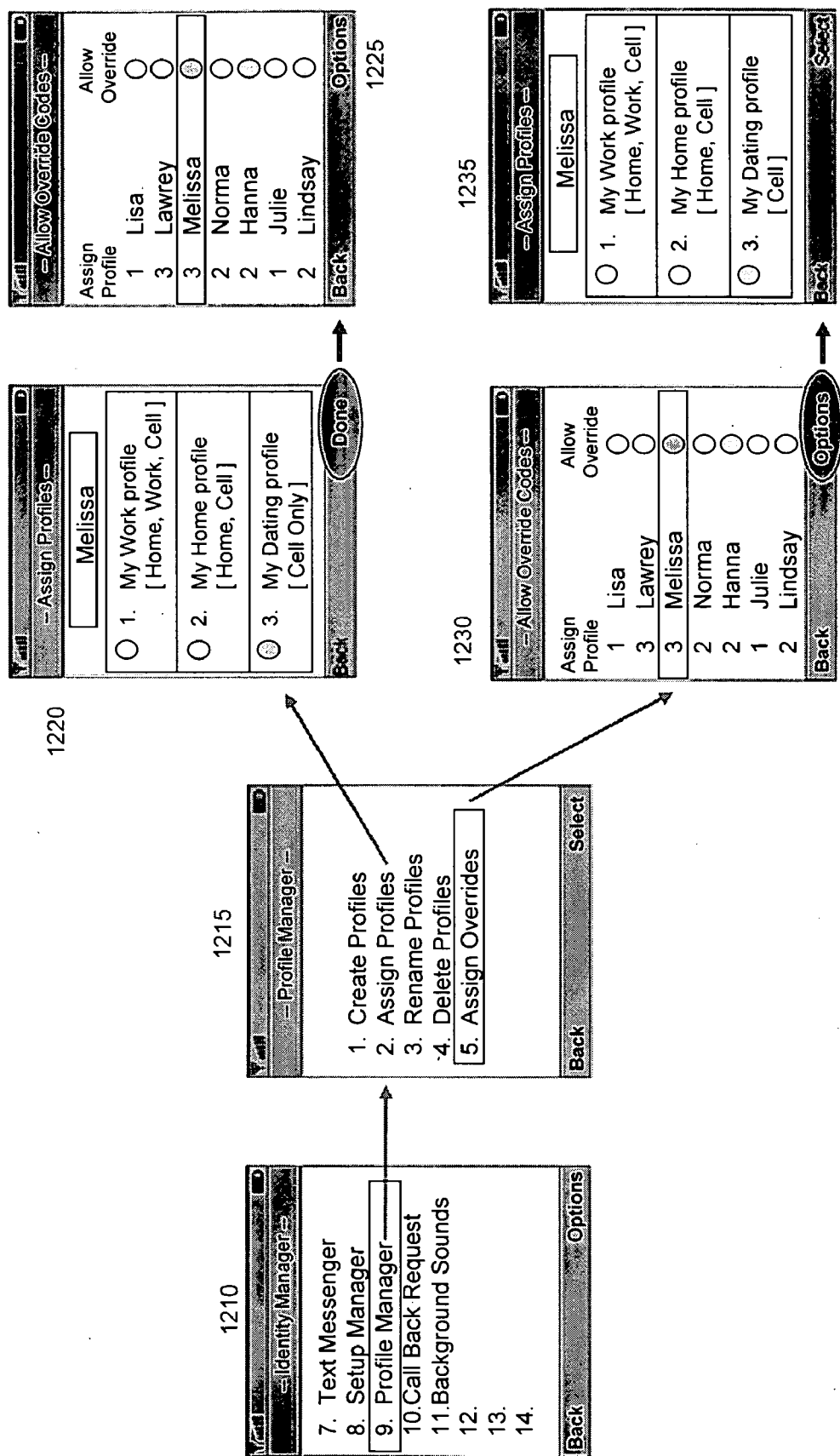


FIGURE 12

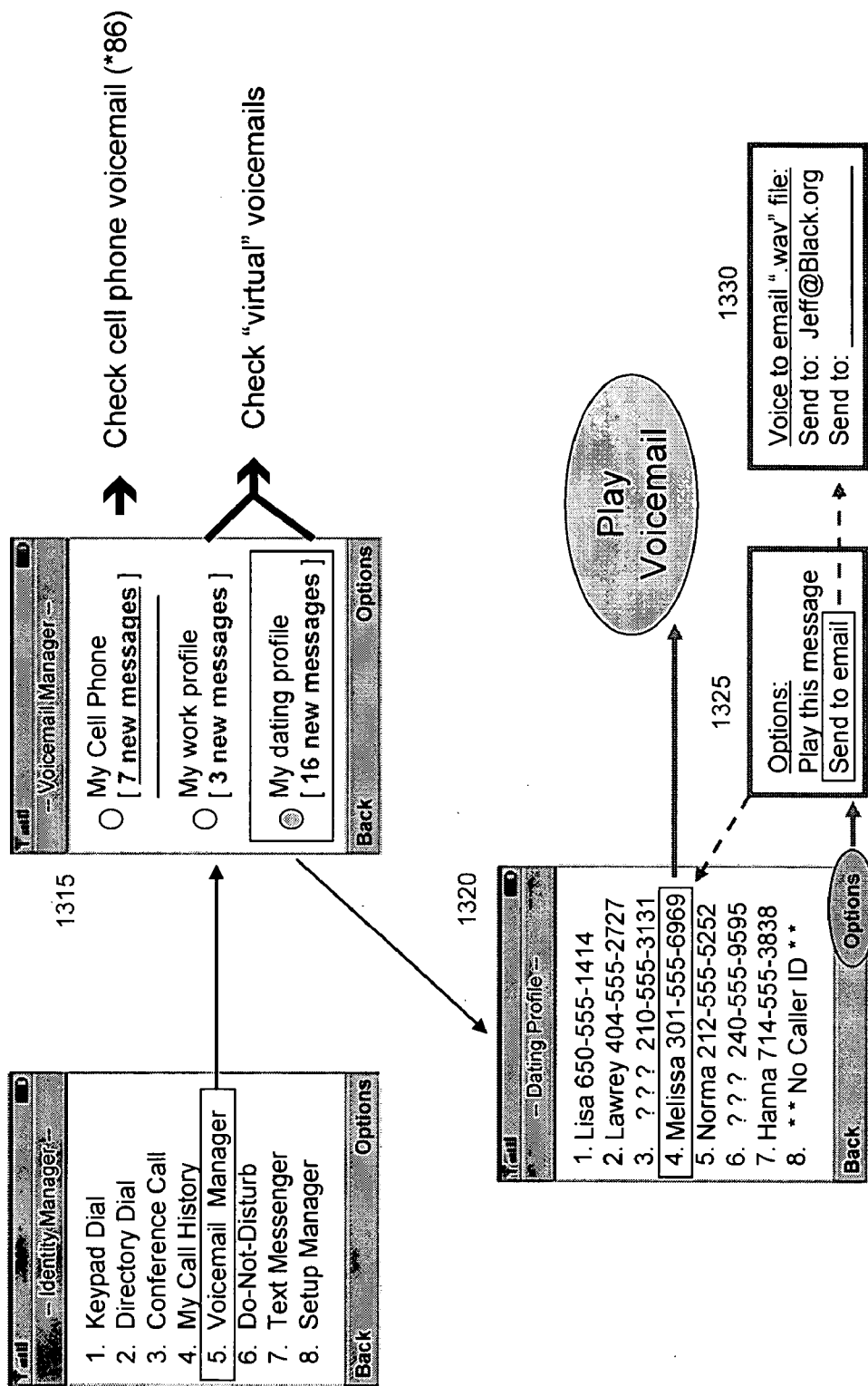


FIGURE 13

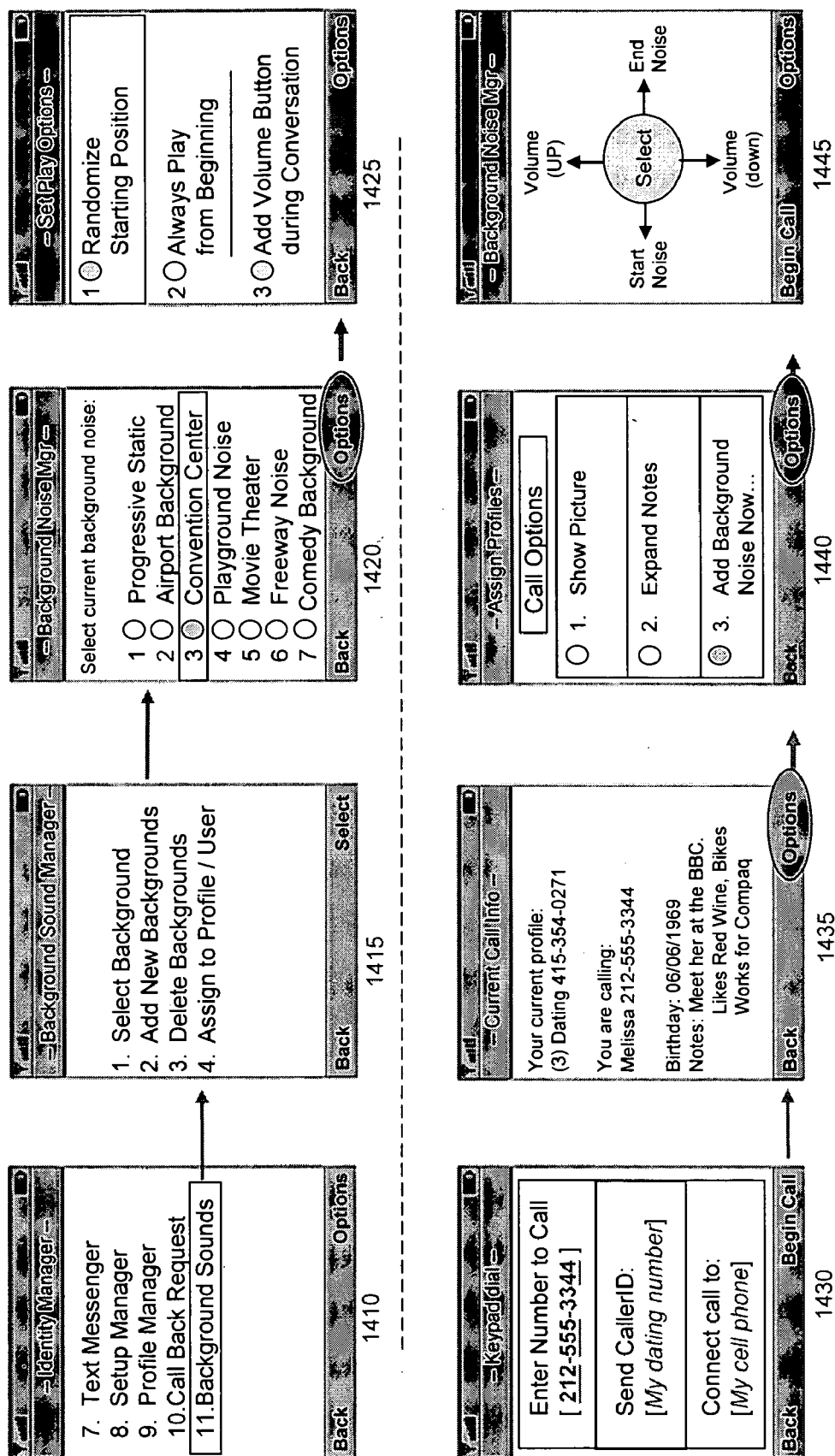


FIGURE 14

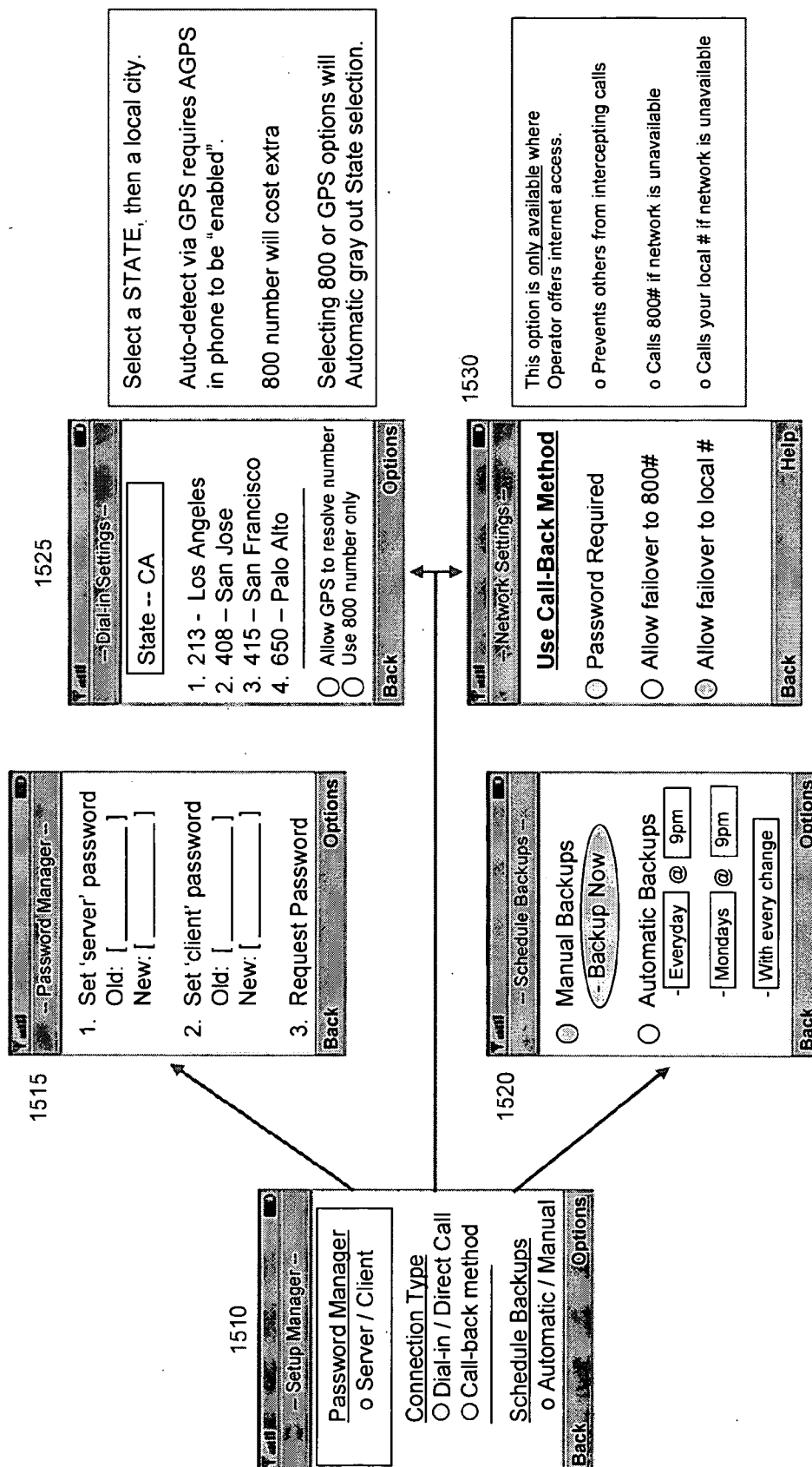


FIGURE 15

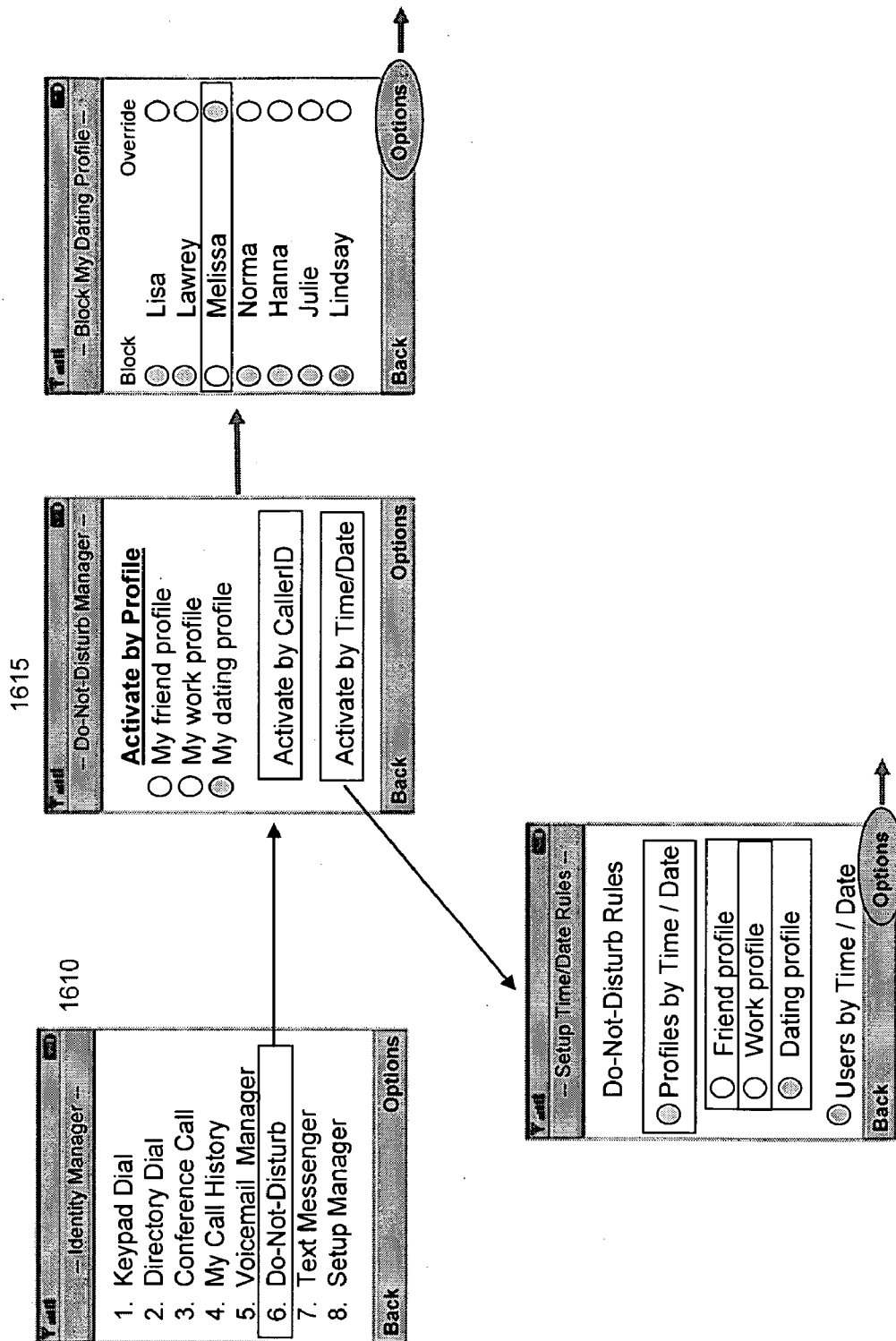


FIGURE 16

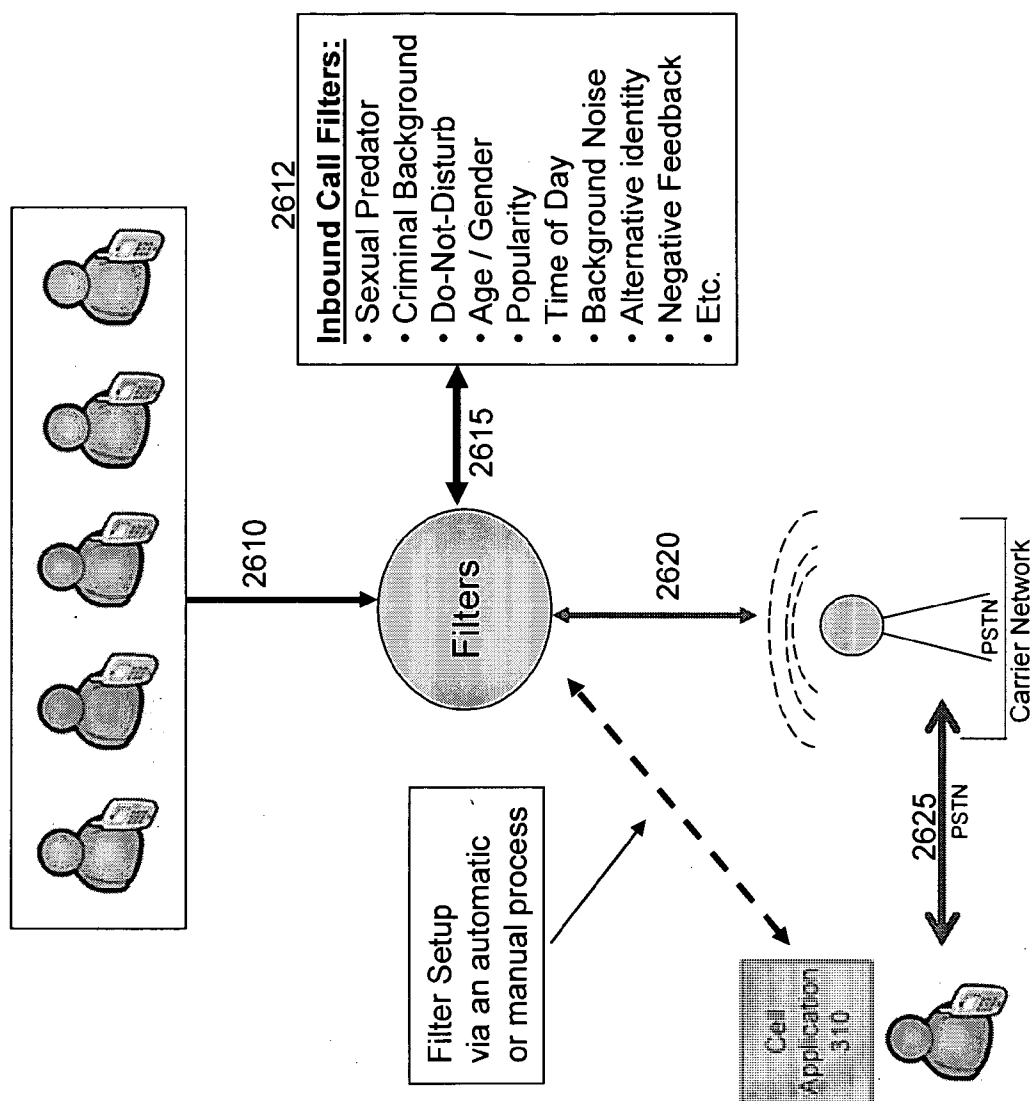


FIGURE 17

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USER-CONTROLLED TELECOMMUNICATIONS SYSTEM

REFERENCE TO PRIORITY DOCUMENT

[0001] This application claims priority of co-pending U.S. Provisional Patent Application Ser. No. 60/631,002, filed Nov. 24, 2004 and co-pending U.S. Provisional Patent Application Ser. No. 60/718,620, filed Sep. 19, 2005. Priority of the aforementioned filing dates is hereby claimed, and the disclosures of the Provisional Patent Applications are hereby incorporated by reference in their entirety.

BACKGROUND

[0002] There is currently an explosive growth in the use of telecommunications devices, particularly wireless telecommunication devices such as mobile phones. Telecommunications services for such devices are usually provided by a communications service provider that provides the devices with access to a telecommunications network for making inbound and outbound communications. In a typical scenario, the telecommunications service provider, such as a cellular carrier, provides a subscriber with the telecommunication device and also provides services associated with the device. For example, the cellular carrier can provide a subscriber with a cell phone and with communications services that permit the cell phone to make and receive telephone calls over the carrier's network.

[0003] One drawback of such a system is that the subscriber is bound by limitations and rules that are set up by the communication service provider. In the example of a cell phone subscriber, the subscriber must typically use a cell phone device that is authorized by the cellular provider. Consequently, the subscriber's use of the cell phone is limited by the capabilities of the devices provided by the cellular provider. Moreover, the cellular service provider controls the behavior and features with respect to inbound and outbound telephone calls made on the service provider's network.

[0004] It would be advantageous for a user of telecommunications devices to be able to control the features of inbound and outbound communications regardless of the limitations placed on such communications by a service provider.

SUMMARY

[0005] Disclosed are systems and methods that enable an end user to control the parameters, behavior, and identity of network communications regardless of the limitations placed upon these controls by the end user's communications device or communications service provider. An intermediate media proxy resides on a telecommunications network and accepts inbound and outbound communications to and from a network communication device. The network communications device can be any device that can communicate with a telecommunications network. The intermediate media proxy functions as an intermediary for network communications that occur between the network communications device and other network communications devices. In this manner, the intermediate media proxy can control various features of the communications independent of limitations enacted by the end user's communications device or communications service provider. The intermediate

media proxy can also re-direct communications between two specified communications devices without the direct knowledge of either party involved in the communication as to the origin, destination, or characteristics of the communication.

[0006] The network that is used by the end user (as an example, a cellular carrier) is used only so far as to connect the communications device (e.g., a cell phone, instant message platform, video terminal, etc.) to the intermediate media proxy over a first communication leg. At that point, the intermediate media proxy is instructed (or has previously been instructed, through manual request or automatically executed rules) to create a second communication leg of the same or similar type to another endpoint. This new communication leg may have a different identity than the first leg per the user's request, or may have certain features which the originating leg did not allow. Such features as applied by the intermediate media proxy may be controlled in-band (via the media, such as spoken words, touch tones, instant message key words, or similar methods) or out of band (via some other network, such as an IP protocol which communicates with a user-driven application on the same device or via a third party interface, or by other interaction) by the user. Moreover, in a conference call scenario, each participant in the conference call can be connected pursuant to a different protocol.

[0007] This method may be used for both outbound and inbound communications in a similar fashion, except that for inbound communications the control is still in the hands of the "customer" (receiving number) and the originating party has no knowledge or control over the methods of communication completion.

[0008] In one aspect, there is disclosed a method and device for controlling communications of an end user, comprising receiving a communication signal at an intermediate media proxy, the communication signal inbound or outbound relative to at least one telecommunications network that is controlled by an entity separate from the intermediate media proxy; and controlling the communication signal, wherein the control is independent of limitations enacted by the entity that controls the at least one telecommunications network.

[0009] In another aspect, there is disclosed an intermediate media proxy, comprising telecommunications equipment communicatively coupled to a telecommunications network, wherein the telecommunication equipment is configured to receive network communication signals over the network, wherein the intermediate media proxy enables an end user to control parameters, behavior, and identity of the network communication signals regardless of limitations enacted by an end user's communications device or communications service provider.

[0010] In another aspect, there is disclosed a method of establishing a conference call, comprising identifying two or more participants of a conference call; providing a conference call notification to each of the two or more participants, wherein the conference call notification includes an identifier descriptive of an originator of the conference call; and varying the identifier for each participant that receives the notification.

[0011] In another aspect, there is disclosed a method of initiating a conference call, comprising presenting a graphi-

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cal user interface (GUI) on handset, wherein the GUI includes at least one field for providing a conference call participant and at least one field for providing an originating identification to be displayed in a notification to the conference call participant, the originating identification associated with an originator of the conference call.

[0012] The details of one or more implementations are set forth in the accompanying drawings and the description below. Further features, aspects, and advantages will become apparent from the description, the drawings, and the claims.

BRIEF DESCRIPTION OF THE DRAWINGS

[0013] **FIG. 1A** shows a high-level, schematic diagram of an exemplary embodiment of a telecommunications network that enables communications between devices communicatively connected to the network in accordance with the present disclosure.

[0014] **FIG. 1B** shows a schematic diagram that illustrates an exemplary scheme for handling incoming communications to a telecommunications device.

[0015] **FIG. 1C** shows a schematic diagram that illustrates exemplary process for handling outbound communications from a telecommunications device.

[0016] **FIGS. 1D-1F** shows several flow charts that describe alternate methods for initiating outbound communications via an alias service provider.

[0017] **FIG. 1G** shows an exemplary embodiment of a telecommunications network that can effectuate user-selective call.

[0018] **FIG. 2** shows a flow diagram that represents an exemplary method that enables a subscriber of an alias service to configure the information that is provided to a called party pursuant to a caller configuration system.

[0019] **FIG. 3A** shows an exemplary wireless telephone equipped with an application and a contact database that reside on the telephone.

[0020] **FIGS. 3B and 3C** show an exemplary configurations of inbound and outbound filter databases.

[0021] **FIG. 4** shows another exemplary embodiment of a telecommunications network that can effectuate selective call configuration in accordance with the present disclosure.

[0022] **FIG. 5** shows a schematic representation of a proximity-based predictive dialing scenario.

[0023] **FIG. 6** shows a flow diagram that represents an exemplary method that enables a subscriber of the alias service to configure the information that is provided to a called party pursuant to a caller configuration system.

[0024] **FIG. 7A** shows an exemplary series of user interfaces that define a Rules Engine for the alias service.

[0025] **FIG. 7B** shows an exemplary flow diagram relating to age verification features of the alias service provider.

[0026] **FIGS. 8 and 9** show a series of exemplary user interfaces that can be shown on a display screen of the phone to effectuate the user provided configuration information for an outgoing call.

[0027] **FIG. 10** shows an exemplary series of user interfaces that allow a phone user to initiate a multi-way conference call.

[0028] **FIG. 11** shows an exemplary user interface for a call history manager that displays call history information that can be grouped pursuant to received calls that pertain to a call profile or group.

[0029] **FIG. 12** shows series of exemplary user interfaces that can be used to set-up and manage profiles.

[0030] **FIG. 13** shows a series of user interfaces that permit a user to access and manage voicemail messages.

[0031] **FIG. 14** shows a series of exemplary user interfaces for configuring background noise for calls implemented by the disclosed system.

[0032] **FIG. 15** shows an exemplary series of user interfaces for setting up proximity based dialing parameters.

[0033] **FIG. 16** shows a series of user interfaces that can be used to manage a Selective Do-Not-Disturb feature.

[0034] **FIG. 17** is a schematic diagram that illustrates the implementation of filters for inbound calls.

DETAILED DESCRIPTION

[0035] **FIG. 1A** shows a high-level, schematic diagram of an exemplary embodiment of a telecommunications network **100** that enables communications between devices communicatively connected to the network in accordance with the present disclosure. The network **100** can comprise, for example, a common or private bidirectional telecommunications network (e.g., a public switched telephone network (PSTN), cellular network, a cable-based telecommunication network, a LAN, a WAN, a wireless network, or combinations thereof), coupled with or overlaid by a TCP/IP network (e.g., the Internet or an intranet).

[0036] At least a first telecommunications device **110** and a second telecommunications device **115** are communicatively linked to the network **100**. The telecommunications devices can be any type of devices that are configured to communicate with the network and to establish communication links with one another over the network **100**. The telecommunications devices are described herein in the context of being a first wireless telephone **110** and a second wireless telephone **115**, although the type of communications device can vary. For example, the telecommunications devices can be anything (e.g., computers, personal digital assistants, land-line telephones, etc.) configured to transmit and receive any type of data over a network.

[0037] Each telecommunications device has at least one address that points to a network location where a communication link can be established with the telecommunications device via the network **100**. Throughout this disclosure, the network devices are sometimes described in the context of being telephones with one or more addresses comprised of telephone numbers. It should be appreciated, however, that the addresses of the communications devices are not limited to telephone numbers, but can include other types of addresses, such as, for example, a Session Initiated Protocol (SIP) Uniform Resource Identifier (URI), an instant message nickname, a shorthand descriptor of a group of other

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addresses, or any identifier that points to a location or node on the network where a communication link can be established.

[0038] The network **100** includes or is coupled to one or more Service Provider Networks (SPNs) **120** that provide communication services to the telephones **110** and **115**. An SPN may be, for example, a Local Exchange Carrier (LEC), a Competitive Local Exchange Carrier (CLEC), a wireless telephony carrier, a VoIP provider, or any other network which provides endpoint-to-endpoint communications between subscribing devices such as the telephones **110**, **115**. Each telephone **110** and **115** is associated with an SPN **120** such that the SPN provides communication access to and over the network **100**. The telephones **110** and **115** can both be associated with a common SPN or can each have a different SPN. For each telephone, the respective SPN provides one or more addresses (referred to as base addresses), such as base telephone numbers, that point to the telephones and that can be used to establish communication links over the network. The SPN typically limits the configuration or behavior of communications links that the telephones establish over the network **100**.

[0039] With reference still to **FIG. 1A**, an alias service provider **180** (also referred to as an intermediate media proxy) resides within the network **100** or is otherwise communicatively linked to the network **100**. The alias service provider **180** can include equipment to enable network communication, such as, for example, servers, call controllers, database equipment, computer processors, memory, etc. The equipment can include computer-readable medium having computer-readable data that includes instructions for performing the processes described herein. The alias service provider **180** is configured to accept communications that the telephones **110** or **115** make over the network to provide control over the communication regardless of the limitations placed upon these controls by the SPN **120** associated with the telephones **110** and **115**. Such control can include, for example, control over the parameters, behavior, and identity of the communications, as described more fully below.

[0040] In order to provide such communications-related services, the alias service provider **180** establishes a service contract with an owner of a telecommunications device such as the telephone **110** or with a SPN. The telephone **110** can optionally be equipped with one or more applications provided by the alias service provider **180** (such as an application **310** shown in **FIG. 3A**) to enable the services. In addition, the alias service provider **180** can provide the telephone **110** with one or more additional addresses (referred to as alias addresses), such as alias telephone numbers, that are controlled by the alias service provider **180** rather than the SPN associated with the telephone **110**. This enables the alias service provider **180** to control communications related to the alias addresses.

[0041] In an alternate embodiment, the alias service provider **180** and the SPN **120** are the same entity or are both controlled by the same entity. In such a case, both the base address(es) and the alias address(es) are provided by that entity. In other embodiments, the application **310** resides on a device other than the telephone **110**, such as on a personal computer. A subscriber is able to access the alias service provider via the personal computer using the application.

The subscriber would be able to generate and define communications using the application.

[0042] In an exemplary context, the alias service provider **180** provides the telephone **110** with one or more alias telephone numbers and an entity controlling the SPN **120** provides one or more base telephone numbers. Each of the base telephone number and the one or more alias telephone numbers can be used by the telephone **110** to access the network **100** for making and receiving communications such as inbound and outbound calls. However, inbound and outbound calls for the alias telephone numbers can be uniquely configured via the alias service provider **180**, as described below. As mentioned, the addresses are not necessarily telephone numbers and the communications are not necessarily telephone calls. Other types of addresses and communications over the network are within the scope of this disclosure.

[0043] **FIG. 1A** provides a high level representation of how the alias service provider enables user-defined control of outbound and inbound communications of the telephone **110** regardless of the limitations placed upon the communications by the SPN **120** of the telephone **110**. An outbound communication **X1** is initiated by the telephone **110** to an address, such as a telephone number, associated with the telephone **115**. As mentioned, the address being a telephone number is merely an example. The network devices do not have to be telephones and the address is not necessarily a telephone number but can be some other type of address.

[0044] The outbound communication **X1** at least initially passes through the SPN **120**. Prior to reaching the telephone **115**, the outbound communication **X1** is at least partially controlled by the alias service provider **180**, which lies somewhere in the network **100** between the telephones **110** and **115**. In an exemplary embodiment, the telephone **110** establishes an initial communication link to the alias service provider prior to sending the outbound communication to the address on the telephone **115**. Pursuant to the initial communication link, the telephone provides the alias service provider with information relating to the upcoming outbound communication to the telephone **115**. The telephone **110** then initiates (via the alias service provider) the outbound communication **X1** to the telephone **115**. An exemplary scheme for enabling the alias service provider to control the outbound communication is described below.

[0045] The alias service provider **180** thus establishes control over the outbound communication **X1**. In this regard, the alias service provider **180** can configure the communication **X1** to enable various features described below, some of which are not permitted or enabled if the communication **X1** were handled exclusively by the SPN **120**. The alias service provider **180** then establishes a communication link **X2** with the telephone **115**. The communication link **X2** can pass through an additional SPN associated with the telephone **115**.

[0046] At this stage, a communication link **X1** exists between the telephone **110** and the alias service provider **180**, and a communication link **X2** exists between the alias service provider **180** and the telephone **115**. The alias service provider **180** then establishes a two-way communication link between the telephone **110** and the telephone **115** with the alias service provider having some control over the configuration of the communication link. Advantageously,

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this scheme permits the alias service provider **180** to provide communication-related features regardless of the limitations placed by the SPN **120**. As mentioned, a specific scheme for handling outbound communications from the telephone **110** is described in more detail below.

[0047] The alias service provider **180** can also control inbound communications to an address associated with the telephone **110**, such as an inbound call to a telephone number of the telephone **110**. The telephone number can be an alias number that the alias service provider **180** assigned to telephone **110** wherein the alias service provider has control over that alias telephone number. Inbound calls to the alias telephone number automatically link to the alias service provider.

[0048] For example, with reference to **FIG. 1A**, the telephone **115** initiates a communication **X2** toward an alias telephone number of the telephone **110**. The communication **X2** is directed to the alias service provider **180** because the alias service provider **180** has previously set-up the alias number to act in such a manner. The alias service provider **180** now has control over the inbound communication **X2**. Next, the alias service provider **180** establishes an inbound communication link **X1** to the telephone **110**, wherein the inbound communication link **X1** may pass through the SPN **120**. The alias service provider **180** then establishes a communication link between the telephones **110** and **115** with the communication link having features enabled by the alias service provider.

Exemplary Schemes for Inbound Communications

[0049] **FIG. 1B** shows a schematic diagram that illustrates an exemplary scheme for handling incoming communications to the telephone **110**. For purposes of this example, an owner of the telephone **110** has subscribed to services provided by the alias service provider **180**, which is positioned in a communication network so as to be able to control at least some communications between telephone **110** and telephone **115**. The telephone **115** is not necessarily a service subscriber of the alias service provider **180**.

[0050] **FIG. 1B** shows a schematic diagram that illustrates an exemplary scheme for handling incoming communications to the telephone **110**. For purposes of this example, an owner of the telephone **110** has subscribed to services provided by the alias service provider **180**, which is positioned in a communication network so as to be able to control at least some communications between telephone **110** and telephone **115**. The telephone **115** is not necessarily a service subscriber of the alias service provider **180**.

[0051] As mentioned, the telecommunications devices **110** and **115** do not necessarily have to be telephones. Moreover, although the example is sometimes described herein in the context of communication via telephone numbers, it should be appreciated that the disclosed processes can readily be used with other types of addresses, such as a SIP URI, an instant message nickname, etc. that are assigned to the telecommunications devices.

[0052] As mentioned, the alias service provider **180** assigns one or more, two or more, three or more, or any quantity of alias addresses to the telephone **110**. In an exemplary context, the alias service provider **180** has assigned the telephone **110** with one or more alias telephone numbers that can be used to receive and send calls on the

telephone **110**. The alias numbers are in addition to a base telephone number that number(s) that was/were **120**. It should be appreciated that the alias service provider control over a pool of alias addresses that can assign to subscribers. This permits the alias service provider to configure the addresses such that inbound communications to the addresses are automatically directed to the alias service provider. The alias service provider can then assign the addresses on a temporary or permanent basis to its subscribers.

[0053] With reference to **FIG. 1B**, the alias service provider **180** is located in communication network such that the alias service provider **180** can manage or access inbound and outbound communications to the telephone **110** particularly with respect to the alias numbers that are assigned to the telephone **110**.

[0054] The alias service provider **180** includes various equipment for effectuating management of inbound and outbound calls. For example, the alias service provider **180** includes one or more call controllers **202**, one or more media controllers **204**, and one or more databases, such as a user database **206**, a temporary or alias number database, a least cost route database **207**, and a call detail records database **208**. The databases can vary and can be used to store information related to the subscriber and the subscriber's telephone **110**, as well as to store data that defines how the subscriber wants to manage inbound and outbound calls. The subscriber can advantageously transfer data related to subscriber's alias number(s) and call configuration data to a different telephone by connecting to the alias service provider over the network.

[0055] In the exemplary embodiment, the sending phone **115** initiates a communication, such as a telephone call **C1**, to the telephone **110**. The call is initiated to an alias number that is assigned to the telephone **110** by the alias service provider **180**. Because the alias service provider **180** assigned the number, the alias service provider **180** can control access to calls received through the number. The call can be routed, for example, through a PSTN **212** and a carrier gateway **214**.

[0056] Upon receiving the call, the alias service provider **180** initially obtains information related to the subscriber that is associated with the alias service number that received the call. For example, the media controller **204** (or any device associated with the alias service provider) contacts the call controller **202** and sends information, such as the alias number, relating to the call. The call controller **202** communicates with one or more of the databases to obtain information that defines how to handle the call based on the alias number being called. In this regard, the controllers can consult a database of media filters that are used to configure various aspects of the inbound call once a communication link between the telephones **110** and **115** is established. The media filters define one or more features that are to be implemented on the inbound call. The features can vary, although some exemplary features are described below. The call controller **202** and media controller **204** can communicate back and forth as required. The media controller **204** can then act on the call based on the obtained information.

[0057] The media controller **204** then establishes a communication **C2**, such as a telephone call, to the telephone **110**. The communication **C2** can be routed through a carrier

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gateway **216** and a PSTN **218** associated with the telephone **110** so that the alias service provider **180** can communicatively connect to the telephone **110**. If desired, the media controller can update a call details records database **219**, such as for billing purposes.

[0058] When the telephone **110** receives the call, the alias service provider **180** provides a notification to the telephone (such as voice or text message) that provides an indication regarding which alias number the inbound call is associated with. As mentioned, the telephone **110** can have several alias address, so it is helpful for the subscriber to identify which alias address is receiving the call. A user may choose to turn off the profile notification message or turn on a prompt that asks the user if he/she would like to take the call. The user can also define rules that govern how the call is handled, such as to force the call to voicemail or automatically play a busy signal, etc.

[0059] In one embodiment, the telephone **110** rings and an identifying auditory message plays. For example, the message can say "You are receiving a call on line **3**" (wherein line **3** is associated with an alias number). Alternately, there can be a unique ringtone associated with each base number and each alias number. Any other type of identifier can be used, such as a unique graphic being displayed, a unique vibration, an automatic auditory signal, etc.

[0060] Depending on whether the subscriber takes the call, the alias service provider **180** then establishes a two-way communication link between the telephone **110** and the telephone **115**, or acts on the call based on filters that were previously defined by the subscriber and/or the alias service provider. During the pendency of the communication between the telephones **110** and **115**, the alias service provider can exert various levels of control over the call.

[0061] FIG. 3B shows an exemplary configuration of an inbound filters database. The inbound filters database includes data related to how the subscriber desires the alias service provider to handle inbound communications to addresses controlled by the alias service provider. The subscriber can specify on an address-by-address basis or rules based on groups of addresses, wherein the rules specify how an inbound communication for that address is handled. The filters database can include various data related to call handling, such as call recording (subject to legal requirements for recording calls), conference call features, voicemail treatment, etc., as described in detail below.

[0062] For example, a filter can specify that an inbound calls for alias number X is sent to a specific voicemail box with a predetermined greeting, while inbound calls for alias number Y is sent to a different voicemail box. In another example, a filter may be set to send all incoming calls for several different alias numbers (or specific calling numbers) to a single voicemail box where alias user or inbound number may receive a different voicemail greeting that was customized for that inbound number or alias number. In other words, a subscriber could have one voicemail box with multiple greetings that vary based on the alias number to which the call is directed.

[0063] For each address or group of addresses associated with the telephone, the user can define rules that govern handling of the incoming communication to that address. Some actions that can be performed include, for example,

recording a call, automatically entering the call into a conference mode, inserting background noise into the call, causing the user's voice to distort during the call, setting a do-not-disturb for the call, varying the call response based on time of day or location of the telephone **110**, etc. This permits the user to use a single telephone **110** to handle multiple telephone numbers while also varying the action taken for inbound calls based on the alias number to which the call is directed.

[0064] FIG. 17 is a schematic diagram that illustrates the implementation of filters for inbound calls. Outbound call filters can be implemented in a similar manner. At operation **2610** an inbound call is initiated from one or more callers. The inbound call is accepted by the alias service provider (as described above) and a filter database **2612** is accessed at operation **2615**. As mentioned, the filters can vary and can include, for example, filters relating to sexual predator information, criminal background information, do-not-disturb information, age/gender information, popularity information, time of day information, background noise information, alternative identity information, and negative feedback information. The filters are used to provide features to the inbound calls as described in more detail below. The filters can be implemented based on various criteria, such as the address to which the call is directed, the address of where the call originated, etc. some of which are defined in the filters database. The filters can be set-up automatically by the alias service provider or in conjunction with user interaction, such as via the application **310**. In operation **2620** and **2625**, the inbound call is directed to the user.

[0065] Thus, a plurality of addresses, such as telephone numbers, are allocated or otherwise mapped to a single telecommunications device, such as a mobile telephone. The plurality of telephone numbers can include one or more numbers that are assigned to the telephone by a conventional mobile phone service provider, as well as additional numbers that are assigned by an alias service provider, as described below. The telephone can receive and/or make calls via any of the telephone numbers assigned to the telephone.

[0066] An exemplary use of the system is where a user wants to remain anonymous during a call or establish other controls to protect his or her privacy. The user can give one of the alias numbers to a limited number of people while giving the telephone's base number or another alias number to a different group of people. The user receives calls for all of the numbers on the same phone but can treat each call differently based on the number to which the call is made.

[0067] For example, a woman wishing to protect her privacy and safety while she pursues online dating relationships can give out an alias number that is reserved for new acquaintances. The woman may wish to stay anonymous during the beginning of dating communications, not revealing her name or other information that would personally identify her. Yet, she still wants to be able to speak with a potential partner while in anonymous mode. Thus, the woman gives new acquaintances a certain alias number, rather than the base number. When the woman receives a call via the "new acquaintance" number, she can either choose to accept it, ignore it, or to have the rule set automatically treat the inbound call in a predetermined manner.

[0068] If the woman no longer wants to be accessible to a particular acquaintance, she can have the rule set automati-

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cally prevent calls from that acquaintance's phone number from being connected to the alias number that she had given the acquaintance. The woman can also cancel the "new acquaintance" number altogether and have it removed as one of the numbers she had acquired from the alias service provider, thereby preventing all the new acquaintances to whom she had provided that number from contacting her further in any manner.

[0069] In another example, a professional, such as a lawyer, can have several alias numbers associated with a telephone. One of the alias numbers can be used for work-related calls such that the lawyer can define rules on how the work related calls are handled and can also know when a call is received that it is work-related. Assume that the lawyer wants a weekend away from work calls but wants the freedom to answer all other calls whether from alias or base numbers. In such a case, the lawyer can specify a rule set that causes all inbound calls to the "work related" alias number to automatically go into a voice mail or do not disturb mode.

[0070] Yet another example is where a user places a "for sale" advertisement in an online classified listings service or a local newspaper or on an online auction service, but does not want to give out her base number in the advertisement for security and privacy reasons. In such a case, the user can be assigned an alias number by the alias service provider, wherein the alias number is only in effect for the duration of the advertisement and is canceled after the item is sold. The user can list the alias number in the advertisement and can receive calls on the alias number on the same telephone as the user's base number. After the item is sold or the user otherwise cancels the advertisement and the alias number, the user will no longer be contactable by members of the public who had retained the alias number.

Exemplary Scheme for Outbound Communications

[0071] FIG. 1C shows a schematic diagram that illustrates exemplary process for handling outbound communications from the telephone 110 to the telephone 115. As described, the alias service provider 180 has assigned the telephone 110 with one or more alias addresses, such as alias telephone numbers, that can be used to receive and send calls on the telephone 110. The subscriber can make outbound calls in connection with any of the alias numbers or with the base number. The alias service provider includes equipment and/or databases as described above with reference to FIG. 1B.

[0072] In an exemplary embodiment, the subscriber desires to initiate an outbound call on the telephone 110 to a target contact identifier, such as a telephone number. For purposes of this description, the telephone number to which the subscriber initiates the call is referred to as the "target telephone number" and the target telephone number points to the telephone 115 in FIG. 1C. The subscriber desires to have the call uniquely configured via the alias service provider.

[0073] In an initial operation, the subscriber launches the application 310 that resides on the telephone 110 wherein the application is configured to initiate a communication link with the alias service provider 180 in a predetermined manner. As described below, the subscriber can use the application 310 to define various parameters relating to how an outbound communication is to be handled by the alias service provider. Upon launch, the application 310 can

optionally establish an initial administrative-related communication link with the alias service provider such as to perform initial administrative tasks. Such tasks can include, for example, downloading software updates, verifying the identity of the subscriber, attending to billing tasks, etc. The initial communication link can occur in various manners, such as via a telephone network or via a Internet connection. While the administrative communication link is being established, the subscriber can be interacting with the application 310 to define the outbound call configuration, as described in detail below.

[0074] In a next operation, the subscriber instructs the application to initiate an outbound communication to a target address, such as an outbound telephone call to a target telephone number. This can occur, for example, by pressing a send key on the telephone or taking some other action. The application establishes a communication link 330 with the alias service provider 180. The communication link 330 can be implemented in any manner, such as through a PSTN or via the Internet. At least one purpose of the communication link 330 is for the application to provide the alias service provider 180 with information regarding the target telephone number and how the user desires to configure the outbound call.

[0075] It should be appreciated that a communication link between the telephone 110 and the telephone 115 has not yet been established. Prior to establishing such a link, the alias service provider 180 takes various actions to configure the call.

[0076] In the next operation, the call controller 202, consults the various databases 206, 207, 208 using information provided by the application 310, such as configuration information specified by the subscriber. The call controller obtains a temporary address, such as a temporary telephone number, that points to the alias service provider 180 and that will be provided to the telephone 110 for establishing a subsequent communication link 335 with the media controller 204. The call controller 202 then transmits (via the communication link 332) the temporary telephone number to the application 310 on the telephone 110.

[0077] In the next operation, the application provides the temporary telephone number to a dialer application on the telephone 110. Using the temporary telephone number, the dialer application then establishes a communication 335 with the alias service provider 180 such as by placing a call over a PSTN 218. In an alternate embodiment, the initial step of the application connecting to the alias service provider 180 via the communication link 330 is eliminated. The subscriber can simply place a telephone call to the alias service provider 180 over the PSTN 218 or in some other manner.

[0078] The media controller 202 can then perform certain actions relating to the outgoing communication, such as by implementing filters contained in user-defined profiles. For example, the media controller can set a predefined user identity, such as caller ID, for the call or can automatically set up recording of the call (subject to legal requirements for recording calls.)

[0079] At this stage of the process, there is still no communication link between the telephone 110 and the telephone 115. The media controller 204 then establishes a

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communication link **340** over a PSTN **212** (or other form of communications link) to the telephone **115** using the target telephone number. A communication link **340** now exists between the alias service provider **180** and the telephone **115**. A communication link **335** also exists between the telephone **110** and the alias service provider **180**. The alias service provider **180** then establishes a two-way communication link between the telephones **110** and **115** with the alias service provider **180** having at least some control over the communication link. If desired, the media controller can update a call details records database, such as for billing purposes.

[0080] The alias service provider can establish the communication link **335** in various manners and protocols. Advantageously, the communication link can be established pursuant to a protocol or network that differs from the protocol or network of the link **335**. For example, the link **335** can be pursuant to any type of service or protocol, such as Skype, Vonage (TM), AIX, PBX) while the call originated from a different protocol.

[0081] In addition, the subscriber can set up one or more profiles that specify how outbound calls will be connected on the termination link **335**. The profiles can be associated with specific number or groups of numbers to use a specific termination service or protocol based on a predetermined rule. In this regard, the subscriber can set up a database of passcodes (such as username/passwords) that can be readily accessed by the alias service provider when the alias service needs a passcode to complete the link **335**.

[0082] FIG. 3C shows an exemplary configuration of an outbound filters database. The outbound filters database includes data related to how the subscriber desires the alias service provider to handle outbound calls. As mentioned, the subscriber can specify rules on a number-by-number basis or rules based on groups of numbers, wherein the rules specify how an outbound call for that number is handled. The filters database can include various data related to outbound call handling, such as call recording (subject to legal requirements for recording calls), conference call features, distortion of voice, piping in of background noise, caller ID, etc.

[0083] FIGS. 1D shows several flow charts that describe alternate methods for initiating outbound communications via the alias service provider. A first method is Tone Dialing method. In an initial operation **360**, the subscriber decides to initiate an outbound call. The method can proceed in either of two ways. In operation **362** the subscriber selects a dial-in number to the alias service provider from a list of dial in numbers on the telephone directory of the telephone. Alternately, in operation **364**, a geographically -closest dial in number is automatically selected using GPS. In the next operation **366**, the telephone dials into the alias service provider using the log-in number. In operation **368**, **369**, and **370**, the subscriber provides instructions to the alias service provider either manually or through an automated process. Using tones, the outbound call is passed through to the target number and the user is connected to the target number using specified parameters.

[0084] Another method is a URL call back method, which was described above with reference to FIG. 1 C. Another method is the alternative number method. After the subscriber decided to initiate an outbound call at operation **360**, the telephone sends call set-up information to the alias

service provider via a uniform resource locator (URL) at operation **372**. At operations **374** and **376**, the server selects an available phone number from a database and then conditions the outbound call with parameters defined by the user. At operation **378**, the alias server provides the telephone **110** with a temporary number and the telephone dials the temporary number at operation **380**. The user is then connected to the target number using the specified parameters at operation **382**.

[0085] With reference to FIG. 1 E, there is shown a carrier or in-network method. Pursuant to this method, the outbound call is made through the SPN of the telephone. In a first operation, **360**, the user decides to make an outbound call. In operation **364**, the telephone sends call-setup information and the target number to the a server or switch operated by the SPN. In operation **386**, the server or switch passes the outbound call through with the desired configuration parameters. The user is then connected to the target number using the specified parameters at operation **388**.

[0086] FIG. 1F shows a flow diagram that describes an exemplary process for initiating outbound calls from the telephone. As shown in operation **390**, the user can initiate the outbound telephone call by either entering an alphanumeric string of digits on the keypad or by selecting a telephone number from an address book on the telephone. Pursuant to one procedure A, the telephone number to be accessed is, for example, 212-555-4455. That number is entered into a User Interface (UI) **398**.

[0087] In a next operation **392**, the telephone checks to see if the outbound number is in a database and is associated with a profile that has predetermined features associated with that caller profile. The check can be performed locally at the telephone or can also be performed by the alias service provider. For example, the profile can specify that any outbound calls for any number associated with the profile are provided with a predetermined identification (such as caller ID). Thus, in operation **393**, a check is performed to determine whether the telephone number has a Group profile ID and, if so, the outbound identification (such as caller ID) is set at operation **296**. Alternately, in operation **394**, the telephone number is not in the database then the user can manually assign an identification to the call. It should be appreciated that the user can override parameters that are defined in profiles related to alias numbers. The UI **398** identifies the caller ID that will be used for the outbound call.

[0088] In another operation **397**, the user can specify the telephone number where the outbound call will originate from. In other words, the user can make an outbound call from a telephone **110**, but the alias service provider will configure the outbound communication such that it appears that the call originated from a number of the user's choosing.

[0089] The user can also configure a profile such that the alias service provider automatically sets the caller ID for an outbound call to a predetermined number based on the outbound number being called. For example, the subscriber could set up a profile such that whenever the subscriber calls number X, the caller ID is set to number Y regardless of which number the subscriber is actually calling from. This could prevent a subscriber from calling someone and the caller-ID showing a work-related alias number when the subscriber only intended to show a dating related alias number on caller ID.

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Various Communication Configurations

[0090] As discussed above, the alias service provider can control inbound and outbound communications for the telephone 110 and provide features to the communications that are not governed by limitations of the SPN. Described below are various features that can be implemented for the communications. For sake of example, the features are sometimes described in the context of the communications being telephone calls between telephones. It should be appreciated, however, that the described features can be modified for use with other types of communications between non-telephone addresses.

[0091] In one embodiment, the subscriber can configure one or more rules that govern how the alias service provider provides such features. The rules can be contained in a Rules Engine that resides in the local application 310 or in the filters databases at the alias service provider. FIG. 7 shows an exemplary series of user interfaces that define the Rules Engine. The rules engine allows the subscriber (or other authority) to preset certain rules that govern use of the alias service. An exemplary basic list of functions is as follows. A Personal Rules Engine interface 710 can be used to access management pages for governing the Rules Engine.

User Identification for Outbound Communications

[0092] FIG. 1G shows another exemplary embodiment of a telecommunications network 100 that can effectuate various communication-related features in accordance with the present disclosure. The telecommunications network 100 is illustrated in a different manner than in FIG. 1A, although the features described with respect to FIGS. 1A and 1B can be implemented in either embodiment. FIG. 1G is used to describe exemplary features that can be implemented with respect to outbound telephone calls from the telephone 110.

[0093] The telecommunications network 100 can include components that are typically associated with a public switched telephony network (PSTN), as well as components associated with a wireless telephony network, such as a cellular telephony network. The telecommunications network 100 can also be configured to include components of an Internet Protocol (IP) network to effectuate corresponding functionality and services, such as, for example, voice over IP (VoIP).

[0094] As shown in FIG. 1G, the telecommunications network 100 can be used to effectuate a voice trunk 105 between a calling party having a first telecommunications device, such as a telephone 110, and a calling party having a second telecommunications device, such as a telephone 115. The telecommunications network 100 can also effectuate conference calling wherein voice trunks are established between more than two telecommunications devices.

[0095] The telecommunications devices are described herein in the context of being telephones (which includes wireless telephones and Voice over IP phones (e.g., web phones) and hybrid wirelessNoIP phones), although it should be appreciated that the telecommunications devices can be any type of device through which voice and/or data communications can be implemented. The telecommunications network 100 further includes one or more Service Provider Networks (SPNs) 120,125 that provide local telephone service to the telephones 110 and 115. As mentioned, an SPN may be a LEC, (Local Exchange Carrier), a CLEC

(Competitive Local Exchange Carrier), a wireless telephony carrier, a VoIP provider, or any other network which provides endpoint-to-endpoint communications between subscribers. The SPNs 120 and 125 can be interconnected via one or more interexchange carriers (IXCs) in a well-known manner, though it is possible that an SPN does not use any IXC or other resource to complete end-to-end calls.

[0096] The telecommunications network 100 includes a master database 130, such as a Calling Name Database (CNAM), that includes profile data relating to subscribers of the SPNs' services. Although FIG. 1G shows only a single master database, it should be appreciated that the network 100 can include more than one master database. The profile data can be accessed by the SPNs 120,125 and provided to a called party pursuant to a caller ID system. In this regard, the master database 130 can include, for example, a name or other indicia associated with each subscriber. For each subscriber, there is also one or more station IDs, such as a telephone number (e.g., a ten-digit number which comprises a three-digit area code defining a specific geographic area followed by a three-digit office code identifying a particular SPN switch, that is followed in turn by a four-digit customer line). For example, Table 1, below, shows exemplary fields that are associated with each subscriber in the master database 130. It should be appreciated that the master database 130 can include additional information or information other than that shown in Table 1. Thus, Table 1 is merely an example and does not limit the configuration of the master database 130.

TABLE 1

Telephone Number	Subscriber Indicia
888-555-1111	John Doe
...	...
777-555-1212	ACME Products

[0097] The master database 130 is maintained by a master administrator that has both reading and writing authority to the contents of the master database 130. The administrators of the SPNs 120, 125 have the ability to read data from the master database 130, but can only write data to the master database 130 in certain circumstances, as described more fully below.

[0098] With reference still to FIG. 1G, each SPN maintains a local subscriber database, such as a local database 135 for the SPN 120 and a local database 140 for the SPN 125. At least some of the data fields in the local databases correspond to the data fields in the master database 130. In this regard, the local databases contain data, such as subscriber indicia and telephone numbers, for the respective subscribers of the SPNs. Thus, each SPN has sole authority for maintaining the data contained in its respective local database. The master database 130 includes a compilation of all of the data contained within the collective local databases or simply a compilation of all data associated with a particular phone numbers issued by a specific SPN 120. In some cases, a SPN 120 may only maintain one database to resolve phone numbers against as in a traditional CNAM (or related) lookup. In the case where only one database is maintained, the SPN's 120 Database 135 and the Master Database 130 would simply be one single database (either 135 or 130) to the rest of the telephony network. It is also understood that

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each SPN **120** and **125** may maintain a single Database **135** or **140** as its official Master Database **130** (or CNAM database). So, the Master Database **130** and the SPN **120**, **125** Databases **135**, **140** may be completely interchangeable in the figures contained herein.

[0099] As mentioned, the SPNs can provide data to the master database **130** pursuant to limited circumstances. For example, the master database **130** can periodically perform a synchronization process wherein the master database **130** queries the local databases for an upload of data from the local databases, which upload can include the entire local database or only modified data. Such a synchronization can occur on a regular basis, such as regularly on passage of a predetermined time span. Another way that the data can be synchronized is by the SPNs sending a prompt to the master database **130** upon any data being modified or updated in the respective local database. Upon receiving such a prompt, the master database **130** performs a data synchronization. Another way that the data can be synchronized is by the SPN pushing modifications (such as Adds, Deletes, and /Changes) to synchronize its database **135** or **140** with the master database **130**.

[0100] Atypical caller ID process works as follows. First, a calling party (who is a subscriber of SPN **120**) at the telephone **110** initiates a call to called party (who is a subscriber of SPN **125**) at the telephone **115**. The telephone **110** establishes a connection with SPN **120**, which establishes a connection with SPN **125**. As mentioned, the connection between the SPNs can occur via an interexchange carriers (i.e., a long distance provider).

[0101] Assuming that the called party subscribes to a caller ID service, and the calling party is not blocking or restricting his phone number from being displayed, the SPN **125** queries the master database **130** for relevant information (such as subscriber indicia (e.g., subscriber name) and/or telephone number associated with the telephone number of the calling party, which information is contained in the master database **130**. As mentioned, the master database **130** contains such information pursuant to the synchronization process that occurs between the local database **135** of the SPN **120** and the master database **130**. The SPN **125** of the called party then obtains the relevant information of the calling party from the master database **130** and provides such information to the called party's telephone **115** for display. For example, the telephone **115** can display the calling party's name and telephone number, which the SPN **125** obtained from the master database **130**. In this manner, the called party has been provided with the calling party's relevant information pursuant to a caller ID service.

[0102] It should be appreciated that the local databases are maintained by their respective SPNs, which generally comprise business entities such as corporations. An individual user (i.e., a subscriber) to the SPN does not have authority or the ability to arbitrarily modify the data contained in the SPN. If a subscriber desired to change its information contained in the local database of the SPN, the subscriber would have to contact the SPN and request that the information be changed. The subscriber then has to wait for the SPN to implement the requested changes. Thus, it can be appreciated that subscribers can be generally reluctant to modify their caller ID data given the potential tedium associated with such a process. Lastly, it is up to the SPN

120, or the party that controls the local phone number database **135** or **140**, to approve any such requested change. In general, SPNs (**120**, **125**) and service providers prefer to use a subscriber's billing information to complete fields found within the local database **135**, which in turn is replicated in the master database **130** later.

[0103] There is now described systems and methods for providing caller ID services that are much more user-friendly for subscribers of SPNs. With reference still to **FIG. 1G**, the telecommunications network **100** includes a private database **145** that can be accessed by subscribers without having to go through a SPN. The private database **145** includes data associated with one or more blocks of telephone numbers, wherein the telephone numbers have been previously purchased or otherwise acquired from one or more SPNs. For example, the telephone numbers associated with the private database **145** can include several hundred telephone numbers acquired from the SPN **120** and several hundred or thousand telephone numbers acquired from the SPN **125**. Lastly, the telephone number associated with the private database **145** can include numbers that a subscriber already has control over (or access to or permission to use) like his home, cell or work phone number. For each telephone number in the private database, there is an associated primary indicia such as the primary indicia described above with reference to table 1. The primary indicia is the indicia that is displayed as a default in connection with caller ID when a subscriber associated with the telephone number makes a telephone call. Each telephone number can also have additional data associated therewith, as described further below.

[0104] In one embodiment, each telephone number in the private database **145** also has one or more alternate indicia associated therewith. The alternate indicia comprise subscriber-specified indicia that can alternately be displayed in connection with caller ID, or that can be displayed in addition to the primary indicia. In other words, the alternate indicia comprise various "aliases" that can be associated with a telephone number in the private database. Table 2, below, shows an exemplary embodiment of a private database having fields for telephone numbers, primary indicia, alternate indicia, and "other" data fields, which are described more fully below.

TABLE 2

	Telephone Number	Primary Indicia	Alternate Indicia 1	Alternate Indicia 2	Other
1	888-555-1111	John Doe	Jane Doe	ACME Products	
2	888-555-6161	Jim Wilson	John Smith	Dad	
3	888-555-3434	Jane Smith	Jane Wilson	Mom	

[0105] In the example of the private database shown in Table 2, the telephone number "888-555-1111" has a primary indicia of "John Doe", a first alternate indicia of "Jane Doe", and a second alternate indicia of "ACME Products". It should be appreciated that although Table 2 shows the private database as having two alternate indicia, that the quantity of alternate indicia uses (if any) can vary.

[0106] The private database **145** of the telecommunications system is set up as follows. First, an entity, such as an

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individual or a business entity, acquires the rights to one or more telephone numbers from telephone service providers such as local exchange carriers or other SPNs. All of the acquired telephone numbers are assigned an entry in the private database 145. The telephone numbers are then assigned, such as for a fee, to subscribers of a service that maintains the private database 145. Henceforth, the subscribers can use the telephone numbers to make and receive telephone calls on the telecommunications network.

[0107] The subscribers to the private database service can populate the fields of the private database as the subscriber sees fit. Thus, the subscriber can specify the contents of the primary indicia and the contents of the alternate indicia and can vary such contents at will. As described more fully below, the subscriber can edit the contents of the private database in a variety of manners, such as by accessing the private directory through the Internet or using a special application that has been loaded into his phone. Accordingly, the subscriber does not have to obtain permission of any SPN to view or edit the contents of the private database.

[0108] The private database 145 is configured to synchronize, either directly or indirectly, with the master database 130. The purpose of the synchronization between the private database 145 and the master database 130 is to provide the master database with the most recent telephone numbers and corresponding primary indicia from the private database 145. In this manner, when a subscriber makes a call associated with one of the telephone numbers, the corresponding caller ID indicia is the primary indicia of the private database. It should be appreciated that the private database 145 can be organized in a variety of manners, and that the organization shown in Table 1 is merely exemplary. For example, the Phone Number and the Primary Indicia may be set to different fields (such as fields 1 and 2) of the database.

[0109] As mentioned, the synchronization between the private database and the master database can occur in either a direct or indirect manner. A direct synchronization can occur in the same manner that the SPN local databases synchronize with the master database 130, such as was described above. That is, the private database 145 communicates directly with the master database 130, as represented by the synch line 147 connecting the private database 145 and the master database 130 in FIG. 1G. When the synchronization 147 occurs, the master database 130 is updated with the telephone number and primary indicia data from the private database 145.

[0110] Alternately, the synchronization between the private database and the master database can be indirect. In the indirect scenario, the private database 145 conducts a direct synchronization with one or more associated SPN local databases, such as with the local database 135, as represented by the synch line 150 in FIG. 1G. The private database 145 provides the local database with updated telephone number and the user-selected primary indicia data. Then, when the local database 135 synchronizes with the master database 130, the latest data from the private database is effectively updated into the master database 130. In any event, for either the direct or the indirect synchronization scenario, the master database is regularly updated with data from the private database 145.

[0111] In use, a subscriber to the private database makes a telephone call in connection with one of the telephone

numbers that the subscriber has rightfully acquired access to via the private database. For such a telephone number, the corresponding primary indicia resides in the private database, the master database, and possibly in one or more of the local databases of the SPNs as a result of previous synchronizations. The telephone call is routed through the telecommunications system as described above. If the called party has a caller ID service, then the called party's SPN will query the master database 130 for the calling indicia for display on the called party's telephone equipment. As mentioned, the indicia in the master database 130 is the same as the primary indicia in the private database as of the most recent synchronization between the master and private databases.

[0112] In one embodiment, subscribers of the private database 135 can remotely access the private database 140, such as to view and/or modify the information connected with their telephone number(s). The subscribers can also amend the contents of the private database, as described more fully below. The private database 135 can be remotely accessed, for example, via a communications network such as the Internet or a special application that has been loaded within a telephone. With respect to amending the private database, an owner of a telephone number in the private database can change the contents of the primary indicia field, change a primary indicia to an alternate indicia, or change an alternate indicia to a primary indicia. Upon the next synchronization between the private database and the master database, the changes that the subscriber made will be reflected in the master database. In this manner, subscribers can update the identification indicia associated with their telephone numbers in a quick and easy manner. Thus, the subscriber effectively updates (i.e., modifies or adds to) the master database 130 (such as a CNAM database) by updating the private database.

[0113] As discussed above, the private database can be managed by a service provider (referred to as the "alias service provider") that resides at some location in the telecommunications network 100. There is now described a method of making a telephone call using the alias service provider. The method enables a subscriber of the alias service to dictate the information that is provided to a called party pursuant to a caller ID system. The method is described with reference to the flow diagram shown in FIG. 2. In a first operation, represented by the flow diagram box 210, a subscriber establishes a telecommunication (or network) connection with the alias service provider. The "subscriber" is an entity, such as an individual or a business entity, that has access to a telecommunication device, such as the telephone 110 shown in FIG. 1G.

[0114] As mentioned, the telephone 110 can be any type of telecommunication device. For example, in one embodiment, the telephone 110 is a conventional telephone that is connected to the SPN 120 in a conventional manner. In this case, the subscriber can connect to the alias service provider by dialing a telephone number associated with the alias service provider. Telecommunication equipment associated with and/or controlled by the alias service provider can reside, for example, at the SPN central office. Alternately, the alias service provider can reside at some other location in the network 100. In such a scenario, the subscriber can manually cause the telephone call to go through the alias service such as by dialing a special phone number associated

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with the alias service. In another embodiment, a dedicated telephone number, such as a **1010-220** type of number, is called to reach the alias service.

[0115] In another embodiment, the telephone **110** comprises a wireless telephone that operates, for example, pursuant to a cellular service or a personal communications service (PCS). In such a case, the telephone **110** can connect to the SPN and/or the alias service provider by first connecting to a base station and mobile switching station of the wireless phone service provider. The mobile phone service provider then hands control of the call over to the alias service provider. Alternately, the mobile service provider connects the call to an alias service system which the mobile service provider controls (e.g., is controlled by the mobile service provider or SPN, which provides service to the subscriber). In such a scenario, the mobile service provider implements and/or runs its own alias service. Thus, the mobile provider can build and manage its own alias service.

[0116] In one embodiment, the wireless telephone **110** is equipped With an application **310** that resides on the telephone **110**, as shown in **FIG. 3A**. The application **310** can have access to a contact directory or database **315** (also known as a phone book) that also resides on the telephone **110**. The contact directory **315** comprises a database (or list) of contacts and can include, for example, names, associated telephone numbers, address, etc., in a well-known manner. When the application **310** is executed, the application **310** causes subsequent telephone calls initiated from the telephone **110** to automatically go through the alias service provider (such as by automatically dialing a number associated with the alias service provider and then causing the alias service provider to initiate the telephone call). In another embodiment, shown in **FIG. 4**, the telephone **110** has access to a contact phone directly **184** that is remotely located at the alias service provider.

[0117] Advantageously, the application **310** interfaces with the contact directory **315** such that the subscriber can initiate calls from the contact directory by selecting an entry (or multiple entries) in the contact directory. The application **310** then automatically causes the telephone to connect to the alias service provider and provide any required information to the alias service provider without any further subscriber interaction. In this manner, the whole process is transparent to the subscriber and does not require any special actions by the subscriber other than to initially execute the application **310**. The application **310** can also be used with other telecommunications devices, such as with a VoIP enabled telephone or a hybrid cellularNoIP telephone.

[0118] In one embodiment, a backup version of the information in the contact directory **315** can reside on a computer network, such as the Internet. The telephone **110** can be used to access the contact directory **315** via the Internet, such as to view, add to, or revise the information contained in the contact directory **315**. Alternately, the user can view and edit the data in the contact directory **315** via a Web browser on a separate device (such as a PC) rather than by coupling the phone to the Internet. This can sometimes be easier than adding to or revising the information using the telephone **110**. In addition, the information contained in the backup version of the contact directory **315** can be downloaded to another phone (whether or not the other phone is compatible with the initial phone), such as if the subscriber has multiple

phones, if the subscriber switches phones, or if the subscriber loses his phone and needs to restore the information to a new phone. Thus, the information contained in the contact directory **315** is contained in the phone itself and a backup is contained at some location on the Internet or some other data network. The backup system is described in more detail below.

[0119] With reference again to the flow diagram of **FIG. 2**, after the subscriber connects to the alias service provider, the subscriber provides relevant information to the service provider, as represented by flow diagram box **215**. In an alternative embodiment of the method, shown in **FIG. 6**, the relevant information in the form of call configuration parameters are accessed from a server that stores profiles at the alias service provider, as represented by flow diagram box **220b** in **FIG. 6**.

[0120] If the subscriber manually provides the information, the subscriber provides the telephone number for the called party, which is the party that the subscriber is trying to call, such as, for example, the telephone **115** in **FIG. 1G**. In the scenario where the application **310** (**FIG. 3A**) is handling the call, the application **310** can be configured to automatically provide the called party's telephone number to the alias service provider. As mentioned, the application **310** can obtain the called party's telephone number from the contact directory **315** (**FIG. 3A**) on the telephone. Alternately, the subscriber can manually enter the called party's telephone number in response to prompts that are provided by the alias service provider.

[0121] The subscriber can select a phone number to dial via a standard (or extended) Contact Phone Directory **184** (**FIG. 4**) that resides at the alias service provider or via the service provider's Name Directory Lookup (NDL) **182** (**FIG. 4**) engine. The NDL may be a self registered (or opt-in) database of names and phone numbers or a more traditional yellow page engine, while the extended Contact Phone Directory **184** may contain phone numbers the user chooses not to enter into his phone but that reside at the alias service provider. This would allow a subscriber to have access to a private contact phone database **184** (**FIG. 4**) that would not appear within his standard contact phone directory **315** (**FIG. 3A**). The subscriber (or the application **310**) can also provide a password, login ID, an EIN or any other such information that the alias service provider uses to verify that the subscriber is authorized to make calls through the service provider.

[0122] The application itself **315** may require the subscriber to provide a user login and/or password to initiate the program itself. Additionally, specific phone numbers, contacts or dialing rules may require re-confirmation of a pass code or a different pass code chosen by the subscriber.

[0123] In another embodiment, the subscriber may select a group code to initiate a phone call, wherein more than one telephone number is associated with the group code. When the subscriber initiates a call pursuant to a group code, all numbers found within the selected group code would be called together. This would allow a subscriber to select everyone on a conference call list with a single selection without the subscriber being required to select each and every individual potential conference participant from his contact list found within his phone. These group codes

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would be located with the subscribers dialing profile or within his/her standard/extended Contact Phone Directory **184** (**FIG. 4**).

[0124] In the next operation, represented by the flow diagram box **220** in **FIG. 2**, the subscriber provides one or more call configuration parameters to the alias service provider. The configuration parameters define the configuration of the outbound call. One such configuration parameter comprises the outbound telephone number that will be used as the subscriber's telephone number for the call. The subscriber can select any of the telephone numbers that the subscriber owns (or has access to) on the private database **145**. For example, with reference to Table 2 above, the subscriber may own telephone numbers **1** and **2** in the database. For purposes of the instant call, the subscriber may select telephone number **1** (888-555-1111) as the phone number that is used as the originating number in the instant call. As described further below, when the called party receives the call, telephone number **1** will be displayed as the caller ID number and the primary indicia (888-555-1111) for telephone number **1** will also be displayed.

[0125] Another such configuration parameter relates to parameters that can be used to vary the configuration of the phone call to make it appear as if the phone call is originating from a different location or environment than the actual location or environment of origination. For example, one such parameter comprises background or environmental noise that is piped into the call, as described below.

[0126] In addition to the above, the application provides a PIM (personal information manager) functionality that is managed via user. When a user receives a call on a standard cell phone (non-PIM), a user interface screen is shown that provides pre-stored information about the calling party or the person being called. This feature works both ways. This feature provides limited PIM functionality to non-PIM-based phones.

[0127] Another configuration parameter can be whether or not caller ID is enabled for this particular call. The configuration parameter can specify that caller ID is enabled or that caller ID is blocked for this call. The configuration parameter can permit a subscriber to flag pre-selected telephone numbers (such as frequently called numbers) to have a default "unblocked" status and other numbers to have a default "blocked" status unless specified otherwise by the subscriber. Such a selective caller ID system would ensure proper dialing parameters for subscribers without the having to manually select "call block (*69) or unblock (*82)" on a call-by-call basis. Selective caller ID would allow a subscriber to flag certain frequently called phone numbers, like his home or office.

[0128] The private database can include default values for each of the configuration parameters, which default values are contained in the "other" fields of the private database. In this manner, each telephone number in the private database (or the Extended Contact Phone Directory **184**) can have a predetermined, default alias configuration that is used for outbound (or inbound) calls associated with those numbers.

[0129] As mentioned, the subscriber can add to or edit the information in the private directory that is associated with telephone numbers owned by the subscriber. The subscriber can change any of the indicia, such as to rename the primary

indicia or the alternate indicia. The subscriber can also change one of the alternate indicia to be the primary indicia. For example, in Table 2 the subscriber can change the primary indicia to read "Acme Products" rather than "John Doe".

[0130] **FIGS. 8 and 9** show a series of exemplary user interfaces that the application **310** causes to be shown on a display screen of the phone to effectuate the user provided configuration information. The user interfaces shown in **FIGS. 8 and 9** can also be used to provider configuration information pursuant to the process described in **FIGS. 1B and 1C**. A first user interface **810** includes a menu with a plurality of user-selectable features that correspond to features described herein. Selection of a menu item, such as item **2** "Directory Dial" can lead directly to a user interface **830** that displays configuration options. Alternately, a user interface **820** (**FIG. 9**) can be displayed, which includes, for example, a list of phone book entries that can be dialed.

[0131] The user can select a phone book entry (such as entry number **6** "John Adams" in **FIG. 9**) wherein the entry is associated with a particular profile. Selection of an entry can automatically change the outbound caller ID based on the number being dialed and the profile associated with that number. For example, if the number being dialed is marked as a business contact vs. a personal contact in the user's local or network database, then the outbound caller ID can be changed to a specified number. Under some circumstances, a user may flag an outgoing caller ID as "no-override" which would prevent a user from accidentally changing his outbound caller ID from his "dating number" to his "work number". This would prevent a users from making a serious mistake with dialing an individual.

[0132] The user interface enables a user to initiate a call from a first phone, such as a cell phone, and instruct the alias service to begin the call on a device other than that cell phone. This is useful for areas where there is poor cell phone coverage or when a user wishes to use a speaker phone. The user interface **410** permits the user to set up the configurations for the call, such as the number to call, the caller ID parameter(s) that should be sent (user interface **830**) and the device that will initiate the call (user interface **850**).

[0133] An authentication system which controls name/number pairs in user interface **820** verifies that a user has physical control of a number before it is added as an acceptable outbound calling ID. The user interfaces shown in **FIG. 8 and 9** and in the other figures herein are just examples of what a user interface could look like and it should be appreciated that the user interface can vary.

[0134] With reference again to **FIG. 2**, in the next operation, the alias service provider establishes a connection with the SPN (such as the SPN **120** in **FIG. 1 G**), which then establishes a connection with the called party's SPN. As discussed above, if the called party uses (or subscribes to) caller ID services, the called party's SPN will connect to the master database **130** to obtain caller ID information associated with the telephone number of the calling party. Such caller ID information is the same information contained in the private database **140** as specified by the user as of the last synchronization between the private database **140** and the master database **130**. Specifically, the SPN will access the telephone number previously specified by the subscriber during initiation of the call and the primary indicia associ-

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ated with that telephone number. Advantageously, the subscriber can easily update or change the primary indicia, or select an alternate indicia, by accessing the private database, such as through the Internet.

[0135] In a scenario where a SPN 120 allows Caller ID information to be passed on during the calling process and the receiving SPN 140 accepts the Caller ID information without requesting a phone number lookup (or ENUM lookup), then the Master Database lookup process in FIG. 1G would be replaced with a simple Send (e.g., Number and Name) from sending SPN 120 to a simple Receive (e.g., Number and Name) at the terminating SPN 125

Blocking or Filtering of Inbound Communication

[0136] A sexual predator or criminal notification feature allows a person to determine (in an automated process) if a person they are calling is a listed sexual offender or if the called person has a criminal record. In addition, the reverse can also be implemented . . . when the alias service provider recognizes that a listed sexual offender or person with a criminal record tries to call one of the virtual numbers, the called party is notified that the call they are about to receive is from a registered sexual offender or someone with a criminal record. FIG. 7 shows exemplary Sexual Predator interfaces 715 and 720 that can be displayed on the telephone 110 for defining the sexual predator features.

[0137] A stealth mode feature is designed for people that want to protect young kids or persons under care from receiving calls from unknown people. This feature blocks calls from a calling party that is not found within the user's current phone book 315 (or online CDF 184). This feature can also be used to block outbound calls to individual numbers that are not listed within the user's phone book(s). FIG. 7A shows an exemplary Stealth Mode interface 725.

[0138] An age or gender verification system allows a user to accept or deny calls based on age and/or gender as defined by an age and/or gender parameter. The age parameter may exist in the cell phone (FIG. 3A) itself or on the network (FIG. 4). The feature allows a user to restrict calls to someone: over, under, or exactly a certain age. This can be utilized by people in the dating world to verify the actual age of a caller or someone they are calling. FIG. 7A shows an exemplary Age Verification interface 730 that governs how the alias service manages such a feature.

[0139] When a call is received from someone outside of the calling parameters, the user has the ability to setup an automatic action. This action can be a pre-recorded message, an automatic hang-up, an automatic busy tone or some other action (such as the blocking actions defined above) that would deter the caller from calling again.

[0140] The age verification can be set within the phone by a carrier (or an unverified age parameter can be set by the user). If an unverified age is set, the other party can be notified. In addition, the age parameter can be set through a credit card or a social security lookup process. Other forms of guaranteed age verification may also be substituted. This feature can be used in conjunction with other rules, if desired.

[0141] In one embodiment, the alias service provider provides a system for age verification of its subscribers. As mentioned, the subscriber can set up profiles that are asso-

ciated with a single alias number or a group of alias numbers. The profiles can be associated with predefined or manually defined rules that govern how inbound and outbound communications are handled with respect to the alias numbers. Age-related features can be associated with a profile.

[0142] FIG. 7B shows an exemplary flow diagram relating to age verification features of the alias service provider. In a first operation 2410, the subscriber (user) selects accesses an age verification user interface, which can be graphical or audio. In operation 2420, the user selects whether to access a primary profile or an alternate profile. Pursuant to a first alternative operation 2490, the user associates the profile with an age that has already been stored in the telephone by an SPN associated with the telephone. In operation 2495, the user's age is stored on the telephone or on a server. The age is then readily accessible by the alias service provider such that the alias service provider can filter communications based on the age.

[0143] In another alternative operation 2430, the user manually enters his or her age. The method then proceeds to operation 2495 as described above. In yet another alternative operation 2440, the alias service provider asks for proof of the user's age. The inquiry as to proof can come from the application itself or it can be prompted from a server at the alias service provider. In operation 2450, the user provides some type of proof as to age. The proof can vary, but can include, for example, a phone number that is tied with age, a social security number, a credit card number, etc. Based on the proof provided, the alias service provider can verify the age, such as by pulling the age from a telco associated with a telephone number provided (operation 2460), an official bureau such as Social Security (operation 2470), or from a credit card issuer (operation 2480). The method then proceeds to the operation 2495 as described above.

[0144] A popularity feature allows a user to set certain restrictions on who is allowed to call (or who they want to call) based on a popularity score. FIG. 7A shows a Popularity Ranking interface 735. In one embodiment, persons that are not subscriber's can provide information to the alias service provider relating to an individual whether or not that individual is a subscriber. Such information can be stored by the alias service provider and used in filtering of inbound and outbound calls.

[0145] A negative feedback allows a user to set certain restrictions on who is allowed to call (or who they want to call) based on a negative feedback from others. In other words, if enough people identify a person as negative, their score will reflect it. An exemplary Negative Ranking interface 1640. Similarly, a "married status" parameter may be set by individuals by simply entering the person's phone number(s) into an application or Alias Service Provider. This method of automated feedback from users would allow an Alias Service Provider to collect lots of information on users that are NOT associated with its' service and without necessarily collecting name or address information on the end user.

Backup Database Service

[0146] As described above, a backup version of the contact directory (FIG. 3A) can be stored on a data network such as on the Alias Service Provider via the Internet. The

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user can copy all the data in the contact database **315** within a particular phone to a backup database on a data network such as the internet. This allows a user to replicate his or her contact information on another phone.

[0147] In one embodiment, the backup database is configured to understand the parameter differences between different models of phones. Some phones permit a database to have a predetermined quantity of parameters for each entry, wherein the parameters are not necessarily uniform between different phone models. For example, some phones only allow one Home, Work and Cell number, while other phones allow multiple Home, Work, and Cell numbers. In addition, some phones allow fax number, email addresses and web site address for each phone contact name. The backup database is configured to map the differences between different phone models—thus allowing a user to upgrade or even downgrade data between feature rich cell phones. The user can connect his or her phone to the backup database and download data as required from the backup directory.

[0148] Because the data is stored on a data network such as the Internet, a user can recover from a damaged phone within minutes of getting his/her new phone without worries about phone handset differences or service providers. This process otherwise could have taken hours to rekey the old phone addresses or prevented a handset model change by the subscriber.

Extended Contact Phone Directories (CPD)

[0149] The contact phone directory (CPD) **184** can reside at some location on a network, such as at the alias service provider **180**. This permits the contact phone directory to store an amount of data that is not limited by the hardware of the subscriber's phone.

[0150] Conventional phone handsets typically come with limitations on the amount of phone names/numbers each handset can store. Because the data is stored locally on the phone, the privacy of each name/number contained within the phone can be compromised if physically accessed by the wrong person. The extended CPD **184** allows a user to maintain a private or extended phone directory on the Internet that is not visible or accessible to anyone but the subscriber. Moreover, because the contact items are stored remotely relative to the phone, the subscriber will not have any limitations on the amount of numbers that can be stored. The alias service provider **180** can also require that a subscriber provide security authentication in order to access the CPD **184**. Because the database is remote and requires authentication prior to access, the contact information is more secure than if stored locally as in a traditional phone **110**.

Conference Calling

[0151] As mentioned, the alias service provider can also be configured to implement an automatic conference call service. Pursuant to this aspect, a subscriber can choose multiple addresses (such as phone numbers, SIP URIs or any other type of address) within his/her contact address list contained on telephone or other endpoint client device or software. When the user hits an execute command (e.g., dial), the application **310** or **185** automatically calls all chosen numbers at once and dumps the resultant calls into a

conference room. In addition, a user may also be able to type in other numbers to be added to the conference call.

[0152] FIG. 10 shows an exemplary series of user interfaces that allows a phone user to initiate a multi-way conference call by simply typing in desired number to be called. In a first user interface **1010**, the user is provided with a list of fields that the user can populate with the numbers to be included in the conference call. The user can either manually enter the number or select the number from numbers contained in the contact database **315** or CFD **184**, such as through user interfaces **1015** and **1020**. The user can also access a user interface **1025** that permits the user to specify the identity (such as a telephone number) from which the call is initiated, as well as a user interface **1030** that permits the user to set up configuration parameters. FIG. 10 shows user interface **1040** with the conference call fields populated with the numbers to be called.

[0153] The user may also lookup people to call via a standard phone book (built into the phone **315** or residing on the aliasing system **180/184**) lookup feature.

[0154] In addition to just placing calls, the application **310** and/or user interface allows different people in one conference call to receive different caller IDs when called. Changing of the caller ID can be performed manually or by the aliasing profiles that are pre-setup on the system **180/184**.

[0155] The application **310** is also configured to implement automatic group conference calls. In this embodiment, the user (i.e. the subscriber) can add a group identifier to one or more entries in his/her contact database **315** within the phone or within his/her Private Database **145** entries or extended CPD **184**. Once group codes have been added, a subscriber can select a single (or multiple) group code(s) and hit the select (or send) key. Such a feature permits a subscriber to establish a conference call with three or more people by selecting a single entry within the phone wherein that entry is associated with a group identifier. In both Conference Calling and Group Conference Calling, the calling party may be required to enter a pass code or press a key on the phone key pad to be added to the conference call. This would prevent non-desired individuals or voice-mail systems from being added to a specific conference call.

[0156] It should be recognized that any name or number within any one of the contact phone directories (**145,184**) may have one or more group codes assigned to it. In this manner, a single entry may be assigned to Group **1**, **2** and **3**, where group **1** is for "pre-party" members, group **2** is for all party members and group **3** is for the clean-up members only, for example.

Downloadable Call Fact Database

[0157] The application **310** is also configured to implement a downloadable call fact database, which can reside at the alias service provider as part of the CFD **184**, for example. The call fact database is a database that contains downloadable data associated with telephone numbers that may be called, such as telephone numbers within the contact database **315** or the CFD **184**. The data can comprise any type of information, such as addresses, personal information, and can also include multimedia data, such as photos, ringtones, video, etc.

[0158] Once the subscriber initiates a phone call, the system may, depending on subscriber options, download

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information about the party (or parties) being called from the CFD 184. The information being downloaded may also contain pictures of each person being called. This information may be located in the Private Database 145 or the Extended CPD 184. The information contained within the download will be user selectable and will work in either a single call environment or a conference call environment mentioned above.

Automated Conference Call Reminders (ACCR)

[0159] In another embodiment, the alias service provider, via the application 310 or the application 185, provides Automated Conference Call Reminders to attendees of an upcoming conference call that has been scheduled via the alias service provider. Such reminders can be triggered by the alias service provider via a voice call, a text message, an email message, by instant message on an alternate network, or via other networks which allow asynchronous notifications. The system may include automatic calendaring notices embedded in the ACCR notifications if that is a user preference, which could be used for automatic calendar system importation. If a user declines participation in a specific call, the application 185 will temporarily modify (or ignore) the declined number during the conference call dialing process.

Automated Profile-Based Dialing (APD)

[0160] In another embodiment, the alias service provider provides Automated Profile-based Dialing (APD) capabilities. APD allows a subscriber, through the application 310 or via the Internet using a web browser or via any other type of client/server interface, to preset a method of calling a called party. The user, through a predetermined interface (such as a web interface or a phone interface), can preset a method of calling a phone number that is located within the Contact Database 315 or the Extended CFD 184. Exemplary user interfaces are described below. Pursuant to APD, the user can configure outgoing communications (such as telephone calls) so that certain profile data is used based on the address (such as a telephone number) that is to be accessed. FIG. 8 shows an exemplary series of user interface for implementing such an APD embodiment.

[0161] This is sometimes described in the context of making a telephone call using a target telephone number, although it should be appreciated that it is not limited to use within a telephone context.

[0162] This might be better understood in the context of an example using the exemplary subscriber profile data shown below. In the example below, the subscriber has 1 or more aliases that can be chosen by the subscriber when calling another subscriber. The APD system, via the application 310 or 180, automatically looks up the profile data parameter for any number that has been dialed through the system and applies an APD method to the outgoing call based on the profile associated with that number. For example, if a subscriber called 415-555-1111, then the APD system automatically assigns profile #1 to the outbound phone call based on previous history of calls. Thus, the phone number 415-555-1111 could be used to populate the caller ID parameters.

[0163] FIG. 11 shows an exemplary user interface for a call history manager that displays call history information that can be grouped pursuant to received calls that pertain to a call profile or group. Instead of just displaying the total

number of missed calls as in conventional phones, the user interface shows a list of calls based on profiles to which each call was associated. This way, a user can jump directly to those critical messages associated with a particular profile or group, while leaving the less critical calls to return at a later time. Similar to a normal cell phone history listing, a user can click on any virtual listing and see all of the information for each caller (e.g., name, number and time of date for the call).

[0164] With reference to FIG. 11, a first user interface 1110 provides a general menu of items, such as item number 4 that leads to a "Call History" user interface 1015. The user interface 1110 leads to other user interfaces such as a Missed Calls interface 1120, a Dialed Calls interface 1125 and a Received Calls interface 1130. The calls can be grouped according to how they relate to predetermined user profiles.

[0165] In another example, if a subscriber called 650-555-5222, then profile 2 would be selected as profile 2 is associated with that number. Other embodiments can include more sophisticated profile dialing, such as the application appending client IDs to calling records and/or automatically recording a phone call and then later forwarding that call to someone's email account.

[0166] Such features can be advantageous in certain professions, such as for the legal profession, where adding the client ID to the dialing profile (and later to a generated call log) would allow an attorney to more effectively bill a client for calls made away from his/her office. In another example, a hospital may want to establish a three way conference call between a doctor, a translator and a patient. In this call, the patient ID would be added to the record and the call can be automatically recorded for a number of reasons. Once the call is completed, the application sends the recording (along with the client ID) to the doctor and the translator which may be located in different locations. The APD system can also be configured to act in a predetermined manner for certain phone number prefixes or number match criteria within an endpoint address. For example, if the subscriber were to call any number that began with 650-600 prefix, then the "Block caller ID" flag would be set. This can similarly work on other endpoint addresses, such as SIP URIs, instant messaging identifiers, or other endpoint identification methods.

[0167] The APD feature can be overridden on any call by changing a flag or value on the handset or cellular phone.

Exemplary Subscriber Profile Data

Profile #	Profile Name	Number	Name
1	Home	415-555-1111	Jeff Black
2	Work	650-555-2222	PartnerVision
3	Cell	408-555-3333	J. Black
4	Dating	926-555-4444	Jeff
6	Parents	210-555-5555	Mr. & Ms. Black
7	Call Blocking	415-555-1111	Restricted/Blocked
8	Web Phone	jeff@test.net	My SIP identity

[0168] FIG. 12 shows series of exemplary user interfaces that can be used to set-up and manage profiles. From an initial user interface 1210, a Profile Manager interface 1215 can be accessed that presents a menu of profile-related management features. The Profile Manager interface 1215

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allows users to setup one or more profiles on the system via user interfaces **1220-1235**. For example, each profile can be setup to: 1) ring one or more number when called—in any fashion, like serial or parallel dialing; 2) automatically preset (or modify) an outbound caller ID to a specified number based on a specific profile or group. It should be appreciated that the profile setup can vary as well as the user interfaces associated with the setup.

Selection of Long Distance Service Providers

[0169] The alias service provider can act on outgoing communications such that outgoing communications to end-point addresses are automatically connected over a predetermined network.

[0170] For example, telephone numbers dialed out of the telephone are automatically dialed via a predetermined long distance provider. For example, the application **310** can cause the telephone to use a standard Long Distance Provider or to dial a number associated with a long distance service (e.g. such as a “1010” type of long distance service). The application **310** can be configured such that the use of a predetermined long distance provider occurs automatically so that the user is not required to modify his normal phone or cell phone entries. In other words, the user simply selects an entry from the contact database and the application **310** automatically initiates the call through a predetermined long distance provider.

[0171] The application **310** can also be configured to dial 4 digit extensions, non-valid phone numbers or alphanumeric words to connect to remote telephone or VoIP users via a standard cell phone. In the case of 4 digit extension dialing, a profile engine in the application maps the 4 digit number found within the phone contact directory **315** or the extended CPD **184** to a full phone number or a VoIP number for dialing purposes. Internet-based address dialing such as by using a VoIP phone with a SIP URI (i.e.: jane@company.com) or instant messaging identity would also be handled via the profile engine since this is a non-standard telephone number for traditional telephone networks. Use of external LDAP directories may also be used to map an short number to an endpoint destination.

[0172] It may be possible for the intermediate carrier to be a media type which is not directly corresponding to the originating media type, and furthermore it may be possible for there to be several translations between media types in the path. As an example, a user may purchase a long distance service plan via a VoIP carrier which uses SIP. That long distance carrier would provide a SIP username and password to allow the user to access their account. The user would provide this username/password pair to the alias service provider, who would in turn use that account on behalf of the end user at such time as that end user desired to make calls from their cell phone that utilized the VoIP account. Example rules are least-cost routing, time-of-day routing, destination or origin routing. Other rules are also possible.

[0173] Thus, dis-similar endpoint identification methods and media methods can be interconnected via the media proxy. As an example, the user may select via the interface on the device (in this example, a mobile phone) an endpoint identifier such as “user@domain.com”, which is a SIP URI. The media proxy will instruct the mobile phone application to dial a specific telephone number associated with a trunk

on the media proxy. The media proxy will then accept that “leg” of the call, and will attempt to create a SIP call “leg” to the given SIP URI. If the SIP RTP session is established with “user@domain.com” then the media proxy will relay the media streams between the two call legs, thus completing the call.

Automated Caller-ID Profile Routing System (APRS):

[0174] The application **310** can also be configured to implement profiles that allow a subscriber to automatically route incoming callers to different voicemail boxes (or a single voicemail box that contains different voicemail greetings). This is referred to herein as an Automated Caller-ID Profile Routing System (APRS).

[0175] An exemplary profile is shown below for a subscriber Michele. The profile includes a name that is associated with a voicemail box and a comments that generally describes the type of calls that are routed to that voicemail box. For example, voice mail box **1** is for incoming calls from professional clients, while voice mail box **6** is for unwanted calls (i.e., annoying people.) The voicemail greeting for each voicemail box can be particularly tailored for the type of call associated with that box.

Exemplary Profile		
#	Name	Comment
1:	Dr. Michele Alison	(Professional Clients)
2:	Michele Alison	(Colleagues)
3:	Michele	(Close Friends)
4:	MJ	(Alias for dating)
5:	“number only”	(All unknown people)
6:	“No Service”	(Annoying People)

[0176] In this exemplary scenario, the user Michele established five different voicemail greetings for a single inbound phone number (e.g., 888-555-1119). It should be appreciated that this scenario is exemplary and that any number of voice mail greetings can be established for any number of inbound phone numbers.

[0177] When a call is received by the APRS, a number that is recognized via caller ID would be processed against the user’s routing profile. The following is an example of routing rules that can be used.

Number:	Profile	Pass-through	Pass call to:
650-555-1111	1	Yes	Cell, Work, Home
212-555-3001	2	Yes	Cell, Home
301-555-7171	3	Yes	Cell
415-555-6969	4	No	
--****	5	No	
327-555-7734	6	No	
jane@company.com	1	Yes	Work

[0178] In the example above, if Michele receives a call from 650-555-1111, then the call is passed to her cell phone, work phone and home phone all at the same time. The alias service provider has records that define the subscriber’s cell, work, and home phone numbers. If the call is answered by

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Michele (with the correct system response), then that device would get the call. For example: She answered the call on her cell.

[0179] On the other hand, if Michele does not properly answer the phone pursuant to the correct response in the rules, then the call will be processed according to the user's APRS profile and routed to a particular voice mail box associated with that number. In this case, the user would receive a voice prompt, for example that "Dr. Michele Alison" is not available and please leave a message. Accordingly, if a call from 301-555-7171 is received, then the call is routed to Michele's cell phone. If there was no answer, then the user would receive a message that "Michele" is not available and please leave a message. Additionally, if Michele receives a call from 415-555-6969 then no attempt would be made to reach Michele and the user would be immediately dropped into voicemail with a message that stated that "MJ" is not available and please leave a message.

[0180] FIG. 13 shows a series of user interfaces that permit a user to access and manage voicemail messages. The voicemail interfaces allow can display a list of voicemails grouped by given profiles or groups. When a user clicks on a profile or group displayed on a user interface (such as user interface 1315, a complete listing of missed calls and/or voice mails for that group is returned (such as in user interface 1320). If a user desires to listen to the 7th message, for example (from Hanna in the UI 1320 in FIG. 13) without listening to the previous voicemails, the user can scroll down the list and select #7 and press a button on the phone associated with play. In addition, if the user desires, a copy of all virtual voicemails is forwarded to a predetermined location, such as an email account for later listening (e.g., from an airplane at 30,000 feet) or simply archived for a user. In addition, a user can forward any voicemail by simply entering in an email address or another telephone number, as represented by diagram boxes 1325 and 1330 in FIG. 13.

[0181] If the user specifies, then specific numbers (e.g., 968-555-7734) can be placed on a (BLOCKED) list that will return a message similar to "The number you have reached is no longer in service. No additional information is available at this time." Lastly, for all other calls (wild-carded) that are not found within the profile list, the users (or callers) would receive a message that stated "The person you are calling" or "the caller you are trying to reach" is not available; Please leave a message after the tone.

[0182] The APRS system can also accommodate direct IP (or Internet Protocol) numbers on both an inbound and an outbound basis. In the example above, if Michele receives a call from SIP URI "jane@company.com" the caller would be transferred to Michele's work number only. Barring any answer by Michele's work number, the caller would receive a voice prompt that is particularly suited for work-related calls. The prompt could state, for example, "Dr. Michele Alison is not available, please leave your name and number and she will call you back during normal business hours. If this is an emergency, then please call 911."

[0183] The APRS is configured to notify the subscriber (e.g., Michele) who is attempting to call her and the subscriber then has the ability to take the call or simply reject it. If the user rejects the call, then the call is pushed back into the voicemail system. If the user decides to take the call, then the device the user is currently on gets the call. In

addition, if the user decides to take the call, he/she is allowed to add background noise (or muxed noises) onto the phone line when he/she takes the call, as described above. An example might be that Michele decides to take the call from home, but she wants to add her default hospital background noise to the call to make the caller believe that she is still at the hospital or another user may add conference center background noise when he/she is really on the golf course. The system also allows the user to take the call and have the entire conversation (or a portion thereof) recorded. Any recording that is made can be emailed back to the subscriber and/or caller (if their email address is known to the system).

[0184] It should be appreciated that the above rules are exemplary and that the APRS can be configured to implement any of a variety of rules for routing incoming calls. For example, the APRS profiles can also route calls based on other parameters, such as, time of day, user status physical location, and other parameters like Account Number (for professionals such as attorneys). This last scenario would allow any valid client to contact the attorney at home at any time of day or night in case of an emergency. The trigger for routing, for example, can be the client account number.

[0185] In addition to conditionally routing multiple calls to the same number(s) into one or more voicemail boxes with different voice prompts, the system can also route multiple phone numbers into one voicemail box with one or more voicemail prompts. This allows the user above (such as Michele) to have her Home, Cell and Work numbers screened by the profiling system above. The APRS allows the same in-bound caller ID numbers to be profiled the same way and all messages could be put in the same voicemail box regardless of the number dialed by the same inbound user.

Background Noise

[0186] As mentioned, the alias service provider is configured to insert background noise into the call such that the called party will hear the background noise as a background to the calling party's voice during the call. The background noise can vary. For example, the background noise can make it appear that the caller is calling from a cell phone, can provide static, can provide "outside" noises to make it appear that the caller is outside, etc.

[0187] Any type of background noise can be used to make it appear that the caller is calling from a certain location or from a certain environment. For example, the alias service provider can insert background noises typically associated with a hospital, airport, or convention center. The alias service provider can maintain a background noise database 183 (FIG. 4) comprised of pre-canned selections. Alternately, the subscriber may upload his/her own background noises to the Background Noise Database 183 (as seen in FIG. 4) from his/her PC or phone 110.

[0188] FIG. 14 shows a series of exemplary user interfaces for configuring background noise for calls. A Background Sound Manager interface 1415 is accessed from an Identity Manager interface 1410. The interface 1415 presents the user with a menu of features that can be selected to configure the background noise, if any, for a call. For example, an interface 1420 can be used to select the type of background noise, while an interface 1425 configures the background noise. Background noises are stored within the Alias Service Provider and may be used for both inbound and outbound calling.

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Advanced Message Notification (AMN):

[0189] The application **310** or **185** is also configured to implement a feature referred to as Advanced Message Notification (AMN). Pursuant to AMN, the application notifies the subscriber when a call or voicemail arrives from a predetermined caller. This permits the subscriber to know when a new voicemail has arrived and to be notified (via the Caller ID feature) what caller is being transferred to the subscriber.

[0190] The AMN system can also notify a subscriber that a priority message has arrived based on the APRS system default profiles, described above. In other words, Michele could be notified via an automated response system or a text messaging or email system that she had just received a call from someone on her dating profile even though all such calls always go directly into her virtual voicemail system. Secondly, this same feature can be used to clarify the name or other attributes of a live transferred call. In this case, the APRS system can append or prefix additional information to (or replace entirely) the Caller ID information that belongs to the originating caller. Lastly, the AMN system may replace the caller ID information with predetermined indicia. For example, the AMN system can notify the subscriber Michele that she has just received a voice message from “boyfriend” instead of the person’s real name.

Voicemail and Call Tracking (CT)

[0191] The application **310** or **180** includes a Voicemail and Call Tracking (CT) engine that enables the application to track remote calls to other assigned numbers to a subscriber in a similar way to how conventional cell phones track in-bound, out-bound, missed calls, and voicemails today. One exemplary difference is, the application **310** allows the subscriber to see this information for one (or more) remote phone numbers that are not directly assigned to that cell phone. This would allow Michele to see that she had 69 voicemails in her dating mailbox and 6 missed calls from phone number “926-555-4444”. The CT engine communicates with the application **310** via a wireless network connection on a when-available basis.

Presence Dialing and Proximity Based Predictive Dialing for Dial-In Services:

[0192] The application is configured such that a subscriber can set his or her profile to ignore calls based on the physical location of the subscriber and/or time of day (or day of the week). The criteria for ignoring particular calls can be defined in a database associated with the subscriber wherein the database resides at the alias service provider. Alternately, the application is configured to user different versions of subscriber profile databases for the subscriber based on various criteria, such as the time of day, the subscriber’s location, time of year, etc.

[0193] For example, exemplary subscriber Jeff could restrict all calls (or selective calls) within his profile from reaching him via one or more of his phone numbers whenever he approaches his physical work or home location. As represented in **FIG. 5**, the application within the phone tracks the physical location of the subscriber using well-known location tracking methods and devices. When the subscriber approaches a known location, the dialing profiles within the subscriber’s control may change.

[0194] In addition, the ability to control calls via presence (or location) can be tied into standard databases, day-timers or related applications. Lastly, the “presence dialing” feature enables exemplary subscriber Jeff to call or text message anyone (or everyone) that was within his contact database which was currently within a one mile radius (or some other distance) of his current position. This form of presence dialing is envisioned as being useful for people that are dating or just ad hoc groups that wish to communicate based on their current proximity to each other.

[0195] In order to access the alias service provider, the subscriber’s phone can call a service number associated with the service provider. In one aspect, the application **310** is configured to locate the most efficient (either from a physical or monetary standpoint, for example) dial-in service number before it dials an alias service provider. The application **310** performs such a task via conventional cellular tower triangulation or though a common method of GPS (or assisted GPS) positioning (as represented in **FIG. 5**). Once the cell phone’s location is determined, a lookup table is accessed by the application and used to identify the closest dial-in locations for the phone to call, thus reducing local and long distance charges to the user.

[0196] It should be understood that the system may allow a user to select a dial-in number from a list of numbers. In the example in **FIG. 5**, Jeff’s cell phone correctly locates the closest dial-in service in Menlo Park. This ability to automatically locate the closest dial-in facility saved exemplary subscriber Jeff from guessing which facility is closest to his location. In addition, this automated location service also saved Jeff from potentially dialing a long distance number to reach a service that actually had a local dial-in number. This service would save Jeff in long distance costs.

[0197] As in Group Conference Calling, subscriber may dial a group call, but limit or restrict all numbers that are located more than a certain distance from the subscriber. In **FIG. 5**, for example, Jeff (the subscriber) initiates a conference call with group **1** which contains Marcy, Michele and Alison and Jeff specified that a 1 mile radius limit should be associated with the call. Since Jeff selected a 1 mile radius, only Michele and Alison were connected to the call. Marcy is outside the 1 mile radius and so the application **310** automatically excludes Marcy from the call.

[0198] It should be noted that the application **310** within the phone will selectively pull local dial-in phone numbers on an ad hoc basis over a wireless network connection. Depending on the available memory and user selections, the application **310** may download more, or fewer, numbers.

[0199] Lastly, the application may also contain alternative default dial-in numbers like toll free numbers or default number in specific countries where local service may, or may not, be available.

[0200] **FIG. 15** shows an exemplary series of user interfaces for setting up proximity based dialing parameters and other configuration parameters. An initial user interface **1510** includes selectable menu items that lead to additional user interfaces, such as a password manager interface, a scheduled backup interface **1520**, and connection type interfaces **1525** and **1530**.

[0201] To connect a user to the closest dial-in point, the system can allow a user to select the closest dial-in location

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or to specify that the AGPS (or GPS) within the phone may be activated to automatically find the closest dial-in point. In the situation where a simple tower number is available, the system can also use this parameter to find the closest dial-in point. Although this is not as accurate as using the GPS positioning in the phone (or cellular network), it is close enough in most cases to find a location number for the phone to call.

[0202] In another embodiment, the alias service provider system enables subscribers to be called back anywhere in the world and then turn control of the session over to the called party. The system can cause a pre-recorded message to be played when the call is answered. This message may ask for a specific person (e.g., when calling your hotel room through a switchboard) or it may ask the called party for a security pass-code to relinquish control over to the called party. At this point in time, the user is connected back into the main system as if he/she had called in directly, but without the cost associated with it.

Emergency Out-of-Service (EOS) Option

[0203] A subscriber may, through the application 310 within his phone or through dialing into the alias service provider, turn on an Emergency Out-of-Service (EOS) option. The EOS option causes the subscriber's phone to immediately give predetermined callers (such as all callers, or just selected groups of callers, or just unknown callers) the impression that this phone number had been discontinued or pulled out of service. This feature would allow a subscriber to quickly remove all evidence of his use of a particular phone number on a temporary basis.

Sponsor-A-Call Advertising:

[0204] Prior to a subscriber actually completing a phone call, he/she may have an option to hear a downloaded advertisement wherein the alias service provider downloads the advertisement to the subscriber's phone. If the subscriber chooses to listen to the entire advertisement, then the advertiser may pick up the cost of the subscriber's call or a portion of the subscriber's call. To verify the ad was heard, the subscriber may have to type in a code that was broadcast during the advertisement or other verification methods which relate the advertisement to the product or the content of the advertisement itself. This type of advertisement may be also present on outbound calls, which would require called destinations to listen to an advertisement before completely connecting a call from a subscriber.

Banner Advertisements:

[0205] The application 310 may employ some form of banner advertisement while the application is being utilized. These banner ads would be downloaded over a wireless network connection to the phone on a regular basis. If the user's location is known via GPS or other means, then the advertisements may be localized to the user's current position. The application 310 may also act as a background screen manager (e.g. a wall paper application) where downloadable ads can be displayed as the default screen background or ads (which may be icons) may also be displayed the outside window of a person's cell phone while the phone is now being used. Once the user agrees to effectively turn his/her phone into a walking billboard, the advertiser (e.g., Nike) may be willing to pay for a portion of the users phone bill. (Note: On a flip phone, the would allow an advertiser to

place an ad on smaller outside window while the person is walking around and a larger ad on the main screen when the phone is open.

Verification System (VS):

[0206] The alias service provider may employ a Verification System (VS) feature that would allow suspecting victims of a harassing or otherwise undesired recent phone call to verify if they have received any calls from an alias service provider system. For example, if phone 115 received several phone calls from a person pretending to be an insurance person, the operator of phone 115 would be able to call into one the alias service provider systems to check if any alias service provider subscribers had called the phone 115 within a certain period of time. It should be noted that the operator of phone 115 would not have to be a subscriber to the alias service provider. This features would simply inform individuals if they have be contacted by anyone using an alternative identity. To verify that the requesting party is truly located at the phone number which they wish to check, the system would dial back the requestor immediately following the request. The alias service provider may associate a fee with this lookup request. There would be an automated method of inquiry to the same results made available to law enforcement agencies. The alias service provider may create a "blacklist" which forbids identity-modified calls from reaching certain numbers on a global basis (across all users.) This blacklist may be managed by a centralized arbiter, or may be modified by individuals.

Alias Phone Number Verification (APNV):

[0207] Similar to the SVS system, the APNV system is designed to verify that a subscriber has authorization over a particular phone number before it allows a number to be added to the subscriber's available outbound calling directory. For instance, if a subscriber 110 wants to add another phone number 115 to his/her alias list for out-bound dialing, the system (or application 185) would issue the subscriber 110 a pass code to use to activate that number within the database. One the pass code has been assigned to the subscriber, the system would terminate the call and redial the requested (115) number. Upon answering the phone, the subscriber would have to enter the correct pass code within a certain number of tries or the number would be not be added to the subscriber's calling list. In some cases, the subscriber may request a call back time or a call back delay to allow him/her to be physically at the phone when the verification call is made.

Name Directory Lookup (NDL):

[0208] The alias service provider also can implement a Name Directory Lookup (NDL) feature. Pursuant to the NDL feature, the user looks up a name within the cell phone application 310, which could be connected to a traditional yellow page engine or a custom or private phone directory. In one instance, the NDL 182 engine may contain information on residents in a small community. Once the end-name is chosen, the phone would read the phone number and automatically dial the number. In this case, the phone could either dial the number directly through the phone or through the alias service provider or another local long distance provider.

[0209] In another embodiment of the NDL 182 engine, people or businesses may be able to self-register themselves.

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This would be similar to a standard yellow page application, but users and businesses will be self registering themselves. This system may contain more than one NDL engine or database **182**. Various NDL databases may be used for specialty applications like dating. Adds, Deletes and Changes would also be controlled by the users or businesses.

Automated ENUM

[0210] The alias service provider is also configured to implement an Automated ENUM feature. The automated ENUM system enables a subscriber to automatically verify a number being called against an ENUM database to identify if this number has a corresponding VoIP service or phone number. In the case where a VoIP address exists, the call would be processed via VoIP protocols. If the ENUM test failed, then the phone call would be processed as a normal phone call through the alias service provider. ENUM is specified in the IETF document RFC #2916.

Selective Do-Not-Disturb (DND):

[0211] Pursuant to a Selective Do-Not-Disturb (DND) feature, the application **310** configures the phone to be set to a do-not-disturb status based on data associated with an incoming call. Thus, DND can be set on an individual basis. Conventional cell phones and VoIP phones that support Do-Not-Disturb features apply this feature to all calls during a certain period. The alias service provider enables a user to set selectively (via the subscriber's profile) the DND feature. In an example, Michele could selectively turn on the DND feature for certain classes of calls (or calls from a selected caller) during certain times of the day or whenever her presence/location changes.

[0212] FIG. 16 shows a series of user interfaces that can be used to manage a Selective Do-Not-Disturb feature. An initial user interface **1610** includes menu items including menu item "6" that leads to a Do Not Disturb Manager interface **1215**. The Do-Not-Disturb Manager interface **1615** displays a list of profiles that can be set to Do-Not-Disturb mode either on an entire profile basis or for individual numbers within the profile. Thus, the user can selectively put one or more profiles into do-not-disturb mode. This can be done manually or automatically by time of date, or other preset conditions.

[0213] When any profile is activated, the users has the ability to setup an override setting that allows individuals in blocked (or temporally blocked) profile or groups to bypass this mode and ring on the desired phone/number as if the user was not part of a block group.

[0214] In addition, the override flag may also cause a warning (or prompt) to be played to any user within an override status (while currently enabled) to tell the user that the cell phone user is currently in do-not-disturb mode. The system may simply warn the caller or ask the caller if he/she wishes to still be connected to the called party.

[0215] There may exist also a global setting to allow a user to block all of a calling profile at once or a user may block each individual profile (or specified caller ID) on an individual basis.

Alias Communication Protocol (ACP):

[0216] It is understood that the applications **310** (FIG. 3A) may establish a connection with the alias service provider

180 through either a tone-based protocol over traditional phone lines **161** (including cellular connections) or via a wireless network connection **162** using a protocol like TCP/IP or similar.

[0217] In any event, all forms of communications referenced above between a subscriber **115** and an end user **110**, including all applications (e.g., **310**, **325**, **145**, **181**, **182**, **183**, **184**, **185**, etc.) in-between, may use either phone lines **161** or wireless network connections **162** to effectuate one or more calls. In some cases, both forms of connections may be used simultaneously.

[0218] An example of the latter could be a subscriber **115** generating a ten-way conference call where the application **310** connects to the alias service provider **180** via a wireless network connection **162** to establish the call setup parameters and then the alias service provider **180** communicates back with the subscriber **115** via a phone line (or wireless phone connection) **161** to complete the call.

[0219] In another embodiment, where the call setup is less demanding, the application **310** may complete the entire call setup and dialing process over a traditional phone line **161** (or cellular connection). In this case, a tone-based hand-shaking protocol would be used.

[0220] Lastly, it should be understood that the majority of the steps found within FIG. 2B may, in some cases, simply reside within the alias service provider's systems, instead of the subscriber's application **310** or contact database **315**. For example, if a subscriber **110** calls another subscriber **115** from a pay phone, the subscriber may only be required to log into the service and then dial the subscriber **115** he/she wishes to call. In this embodiment, the system simply uses the call setup defaults found within the subscriber's profile for the called party **115**.

Call Cost Manager

[0221] The alias service provider provides a Call Cost Manager feature that allows a user to enter a phone number into a phone and receive a proposed charge for the alias service provider to implement the call. Instead of actually dialing the number, the system automatically prices the phone call based on what the call might cost at that moment. In addition, the system prices out a per minute charge and an estimated (or exact) price for a call over an exact duration. The start/end time of the calls, the time of day, the underlying network, quality preferences, and any other characteristic of the call (including services offered by the service provider) may be included in the price quote.

[0222] A number of embodiments of the invention have been described. Nevertheless, it will be understood that various modifications may be made without departing from the spirit and scope of the invention. Accordingly, other embodiments are within the scope of the disclosure.

What is claimed:

1. A method of controlling communications of an end user, comprising:

receiving a communication signal at an intermediate media proxy, the communication signal inbound or outbound relative to at least one telecommunications network that is controlled by an entity separate from the intermediate media proxy;

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controlling the communication signal, wherein the control is independent of limitations enacted by the entity that controls the at least one telecommunications network.

2. A method as in claim 1, further comprising administering a set of filters that define how the communication signal is controlled.

3. A method as in claim 1, wherein controlling the communication signal comprises assigning an identification to an inbound or outbound communication signal, the identification being different from an identification assigned by the entity that controls the at least one telecommunications network.

4. A method as in claim 1, wherein the communication signal is a telephone call and wherein the identification is a caller identification.

5. A method as in claim 3, wherein the identification is assigned based on predetermined rules contained in a profile.

6. A method as in claim 1, wherein controlling the communication signal comprises associating background noise with the communication signal.

7. A method as in claim 1, further comprising assigning at least one alias address to the end user, wherein the at least one alias address points to a network location where a communication link can be established and wherein the intermediate media proxy controls communications relating to the at least one alias address.

8. A method as in claim 7, wherein the at least one alias address comprises an alias telephone number.

9. A method as in claim 7, wherein a plurality of alias addresses is assigned to an end user, and further comprising providing a notification to the end user when a communication is received on an alias address, the notification identifying the alias address on which the communication is received.

10. A method as in claim 7, further comprising managing a profile relating to at least one alias address, the profile defining one or more actions that are performed when a communication is received with respect to an alias address in the profile.

11. A method as in claim 1, wherein the communication signal is an outbound communication signal of a first communication protocol and is directed to a target address, and further comprising establishing a communication link with the target address, wherein the communication link is of a second communication protocol different than the first communication protocol.

12. A method as in claim 7, wherein the alias address expires after a predetermined time period.

13. A method as in claim 1, further comprising varying the control of the communication signal based on parameters determined to be associated with the communication signal.

14. A method as in claim 13, wherein the parameters relate to an originator of the communication signal, the parameters including age and marital status and negative feedback data.

15. A method as in claim 1, wherein an inbound communication signal is received, the inbound communication signal being originally directed to a first address, and further comprising:

redirecting the inbound communication signal to a second address; and

notifying an end-user that the inbound signal has been redirected to the second address.

16. A method as in claim 1, wherein an outbound communication signal is received, the outbound communication signal associated with a destination address specified on a handset that originated the outbound communication signal, and further comprising:

directing the outbound communication signal to a destination address different from the destination address specified on the handset.

17. A method as in claim 1, wherein an end-user uploads data to the to the intermediate media proxy, the data defining how the intermediate media proxy controls the communications signal.

18. A method as in claim 1, wherein the communication signal is first received from a first communication device that sent the signal for receipt by a second communications device controlled by the end user, and further comprising:

establishing a communication link between the first communication device and the second communication device pursuant to information associated with and selected by the user of the first communications device.

19. A method as in claim 1, wherein an inbound communication signal is received, the inbound communication signal originating from an actual originating address, and further comprising:

directing the inbound communication signal to the end user; and

providing the end user with an originating address that differs from the actual originating address.

20. A computer-readable medium having computer-readable data for performing method of controlling communications of an end user, the method comprising:

receiving a communication signal at an intermediate media proxy, the communication signal inbound or outbound relative to at least one telecommunications network that is controlled by an entity separate from the intermediate media proxy;

controlling the communication signal, wherein the control is independent of limitations enacted by the entity that controls the at least one telecommunications network.

21. An intermediate media proxy, comprising:

telecommunications equipment communicatively coupled to a telecommunications network, wherein the telecommunication equipment is configured to receive network communication signals over the network, wherein the intermediate media proxy enables an end user to control parameters, behavior, and identity of the network communication signals regardless of limitations enacted by an end user's communications device or communications service provider.

22. A method of establishing a conference call, comprising:

identifying two or more participants of a conference call;

providing a conference call notification to each of the two or more participants, wherein the conference call notification includes an identifier descriptive of an originator of the conference call; and

varying the identifier for each participant that receives the notification.

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23. A method as in claim 22, wherein the identifier is a caller ID.

24. A method as in claim 22, wherein the conference call notification is automatically provided based on a profile.

25. A method as in claim 24, wherein the profile includes rules that vary based on the participant identified for the conference call.

26. A method as in claim 22, further comprising displaying a graphical user interface that permits a user to identify the two or more participants and to identify the identifier for each participant.

27. A method as in claim 22, wherein the two or more participants are identified using an address data base.

28. A method of initiating a conference call, comprising:

presenting a graphical user interface (GUI) on handset, wherein the GUI includes at least one field for providing a conference call participant and at least one field for providing an originating identification to be provided in a notification to the conference call participant, wherein the originating identification is associated with an originator of the conference call.

29. A method as in claim 29, wherein the graphical user interface permits the conference call participant to be selected from digital phone book.

30. A method as in claim 29, wherein the originating identification is a caller ID.

* * * * *

Exhibit S

**IN THE UNITED STATES DISTRICT COURT
FOR THE WESTERN DISTRICT OF TEXAS
WACO DIVISION**

FLYPSI, INC. (D/B/A FLYP),

Plaintiff,

vs.

DIALPAD, INC.,

Defendant.

Civil Action No. 6:21-cv-642-ADA

JURY TRIAL DEMANDED

**PLAINTIFF FLYPSI, INC.'S DISCLOSURE OF
CLAIM CONSTRUCTION EXTRINSIC EVIDENCE**

Pursuant to the Court's Order Governing Proceedings - Patent Cases ("OGP") and Scheduling Order (Dkt. No. 23), Plaintiff Flypsi, Inc. ("Flyp") identifies its extrinsic evidence in support of the plain and ordinary meaning of the identified term and in rebuttal to Defendants' constructions.

Extrinsic evidence that may be relied upon by Flyp can be found at Bates No. FLYP0004292 – FLYP0004344 in Flyp's contemporaneous document production and are as follows:

- Oxford University Press, A Dictionary of Computing, Sixth Edition (2008), pg. 501-502;
- Barron's, Dictionary of Computer and Internet Terms, Eleventh Edition (2013), pg. 482-483;
- Harry Newton, Newton's Telecom Dictionary, Twenty-Sixth Edition (2011), pg. 1108;
- Ray Horak, Webster's New World Telecom Dictionary (2008), pg. 466-467;
- Telephone Switch, PC Mag Encyclopedia (available at <https://www.pcmag.com/encyclopedia/term/telephone-switch>);

- Telco Switch, PC Mag Encyclopedia (available at <https://www.pcmag.com/encyclopedia/term/telco-switch>);
- Switch.co Renames as Dialpad, Transforms Workplace Communications (available at <https://www.dialpad.com/press/switchco-renames-as-dialpad/>);
- Duvid Rottenberg, FreeSWITCH Explained (2021) (<https://freeswitch.org/confluence/display/FREESWITCH/Introduction>);
- Duvid Rottenberg, FreeSWITCH Explained (2017) (<https://freeswitch.org/confluence/pages/viewpage.action?pageId=15696289>);
- "FreeSWITCH 1.0.1 "Phoenix" Released!" July 2008 (available at <https://web.archive.org/web/20080819123414/http://www.freeswitch.org/node/130>);
- "It's Official! FreeSWITCH 1.2 Has Been Released," August 2012 (available at <https://web.archive.org/web/20120920033724/http://freeswitch.org/node/410>);
- The 5ESS Switching System, The AT&T Technical Journal, July–August 1985, Vol. 64, No. 6, Part 2;
- Houghton Mifflin, Webster's New College Dictionary, Third Edition (2005), pg. 70.

In addition, Flyp reserves the right to rely on any of the references cited on the face of the Asserted Patents, which are all publicly available.

Flyp may also rely on previously produced documents by Defendant as extrinsic evidence in this matter including:

- DAP000000115-119
- DAP000000133-138
- DAP000000156-159
- DAP000000186-193
- DAP000000266-275
- DAP000000315-329
- DAP000000339-341
- DAP000000360-365
- DAP000000382-387

- DAP000000401-409
- DAP000001337
- DAP000001338
- DAP000001339-1392
- DAP000001449-1477
- DAP000001510-1512

Flyp also discloses that it may rely on Dr. Scott Nettles for rebuttal testimony, which Flyp anticipates will include at least testimony regarding the plain and ordinary meaning of the identified terms, Defendant's anticipated indefiniteness arguments, and evidence of on one of ordinary skill in the art at the time of the invention.

More specifically, Flyp may rely on upon the opinions of Dr. Nettles in support of its position that the identified term be given their plain and ordinary meaning and in rebuttal of Defendant's constructions, which are improper in view of the intrinsic evidence. In particular, Dr. Nettles may testify regarding the plain and ordinary meaning of the claims and claim terms as understood by a person of ordinary skill in the art who has read the specification of the Asserted Patent, their related applications, their file histories of Asserted Patents and their related applications, and any other relevant intrinsic or extrinsic evidence. Further, Dr. Nettles may rebut the appropriateness of Defendant's proposed constructions, or any expert declaration submitted by Defendant in view of the same evidence.

In addition, Defendant has alleged that the terms "switch" and "associated with" are indefinite. In response to these allegations, Flyp may rely upon the opinions of Dr. Nettles that these elements are not indefinite. Such opinions may include the knowledge of one of skill in the art at the time of the relevant inventions as well as opinions regarding support for the same in the

specification and intrinsic evidence. It is expected that to the extent Dr. Nettles is used for this rebuttal evidence it will be limited in scope to opining on the validity of the above identified elements in light of Defendant's allegations of indefiniteness or plain and ordinary meaning. This testimony may be required to point out portions of the specification of the Asserted Patents and any other relevant intrinsic or extrinsic evidence that provide support for the allegedly indefinite claim elements in light of the general knowledge of one skilled in the art at that time.

Dr. Nettles is qualified to opine on such matters based on his industry and academic experience. Such experience includes over 40 years of industry and academic experience as a professor at the University of Texas and various other universities, as a researcher in fields of wireless and mobile networking and in the design and implementation of router, switch, firewall, and wireless node prototypes, as well as being a testifying expert in numerous patent cases.

Finally, Flyp reserves the right to rely on any extrinsic evidence identified by Defendant.

Flyp identifies the preceding extrinsic evidence in anticipation of the meet-and-confer process set forth the OGP and the Scheduling Order. Thus, Flyp's identification is based upon its current knowledge, its current understanding of Defendant's invalidity contentions and corresponding implied claim constructions, and its investigations to date. Accordingly, Flyp reserves the right to amend its disclosures of extrinsic evidence based on, without limitation, the Defendant's identification of extrinsic evidence, information learned while meeting and conferring with Defendant's counsel pursuant to the OGP, any changes to Defendant's construction or contentions, or for any other appropriate reason.

DATED: December 1, 2021

Respectfully submitted,

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CERTIFICATE OF SERVICE

I certify that a true and correct copy of the foregoing document was served upon the following counsel, via email, on December 1, 2021.

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