(12) **United States Patent**　　　　(10) **Patent No.:** **US 8,098,599 B2**

Ho et al.　　　　　　　　　　　　　(45) **Date of Patent:** **Jan. 17, 2012**

(54) **METHOD AND SYSTEM FOR MULTIPLE PARTY TELEPHONE CALL**

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 676 days.

(21) Appl. No.: **11/353,696**

(22) Filed: **Feb. 13, 2006**

(65) **Prior Publication Data**

US 2007/0189269 A1　Aug. 16, 2007

(51) **Int. Cl.**
　　H04L 12/16　　　(2006.01)
(52) **U.S. Cl.** **370/260**
(58) **Field of Classification Search** 370/351–356, 370/401–402, 259–271
　　See application file for complete search history.

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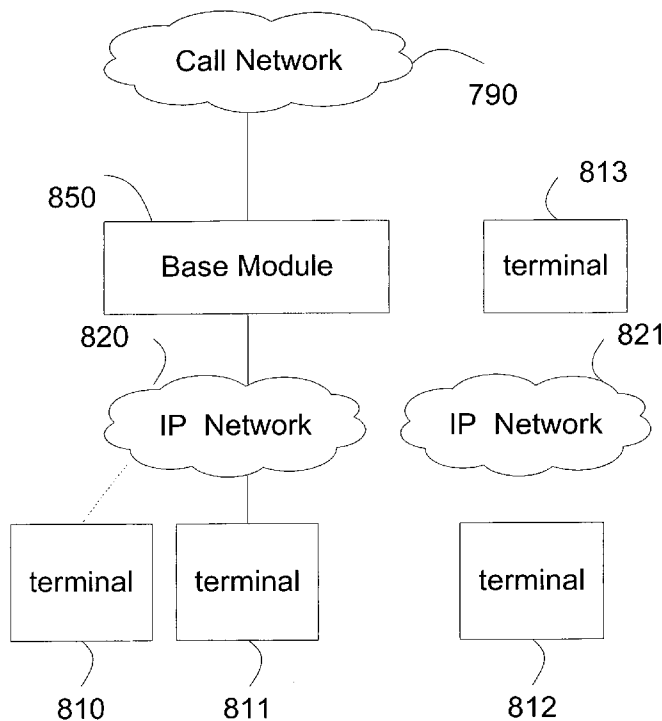* cited by examiner

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(57) **ABSTRACT**

Provided is a method and system for multiple party telephone calls. The method includes receiving an incoming voice telephone call from a call network at a multi-party module, and allowing, by the multi-party module, for a plurality of terminal modules to participate in the incoming voice telephone call, each terminal module comprising a telephone.
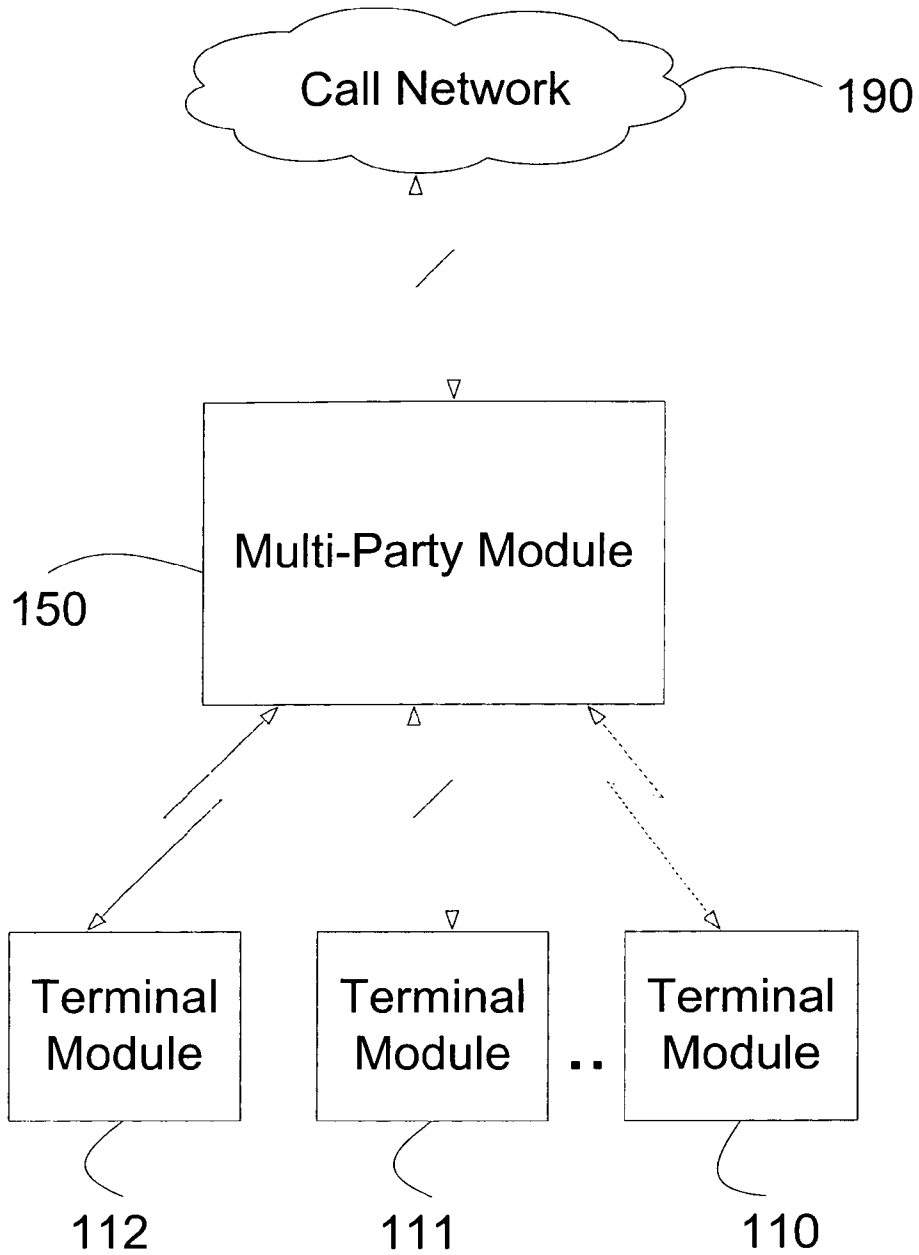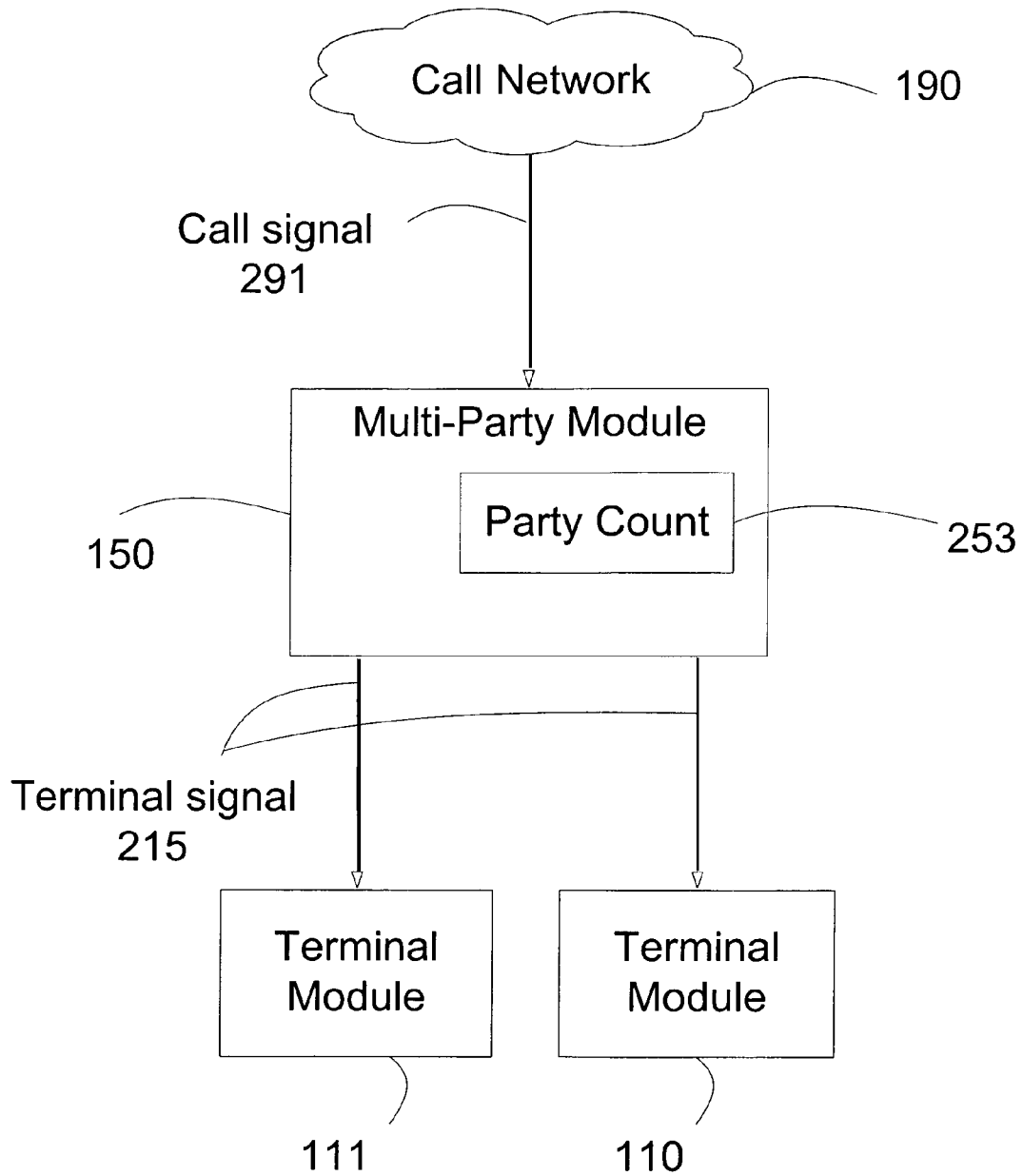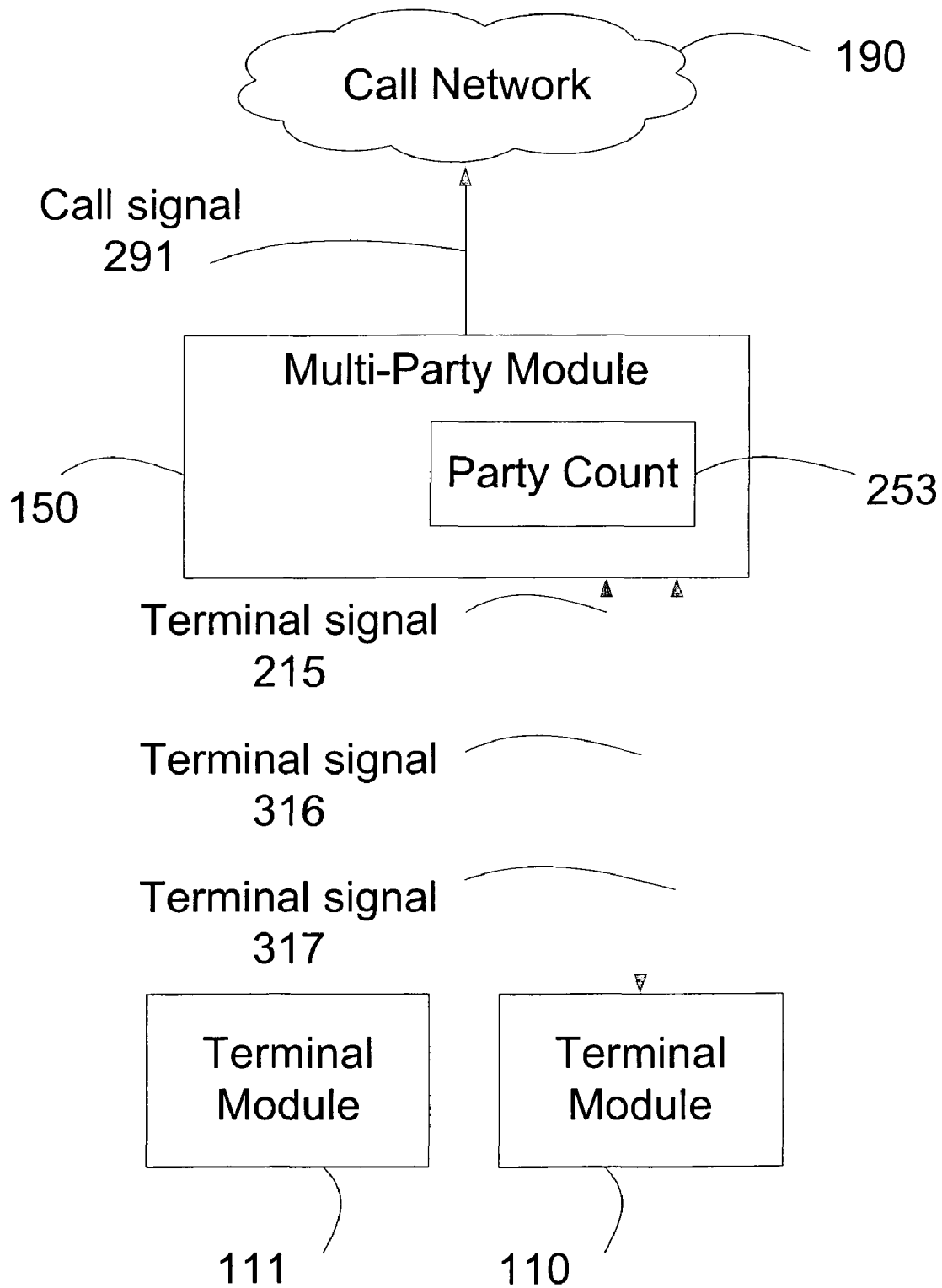
22 Claims, 8 Drawing Sheets

Call Network   190

Multi-Party Module

150

Terminal Module

Terminal Module

.. Terminal Module

112   111   110

Figure 1

Call Network 　　～ 190

Call signal
291

Multi-Party Module

Party Count 　　～ 253

150

Terminal signal
215

Terminal
Module

Terminal
Module

111 　　　　110

Figure 2

Call Network   190

Call signal
291

Multi-Party Module

Party Count   253

150

Terminal signal
215

Terminal signal
316

Terminal signal
317

Terminal
Module

Terminal
Module

111   110

Figure 3

Figure 4

Call Network

190

Multi-Party Module

150

Party Count

253

Terminal signal

215

| Terminal Module | Terminal Module | Terminal Module |

112   111   110

Figure 5

Call Network

190

Call signal
693

Call signal
291

Multi-Party Module

150

Party Count

253

Terminal signal
617

Terminal signal
215

Terminal
Module

Terminal
Module

Terminal
Module

112

111

110

Figure 6

Call Network

790

750

Base Console

713

handset

720

IP Network

721

IP Network
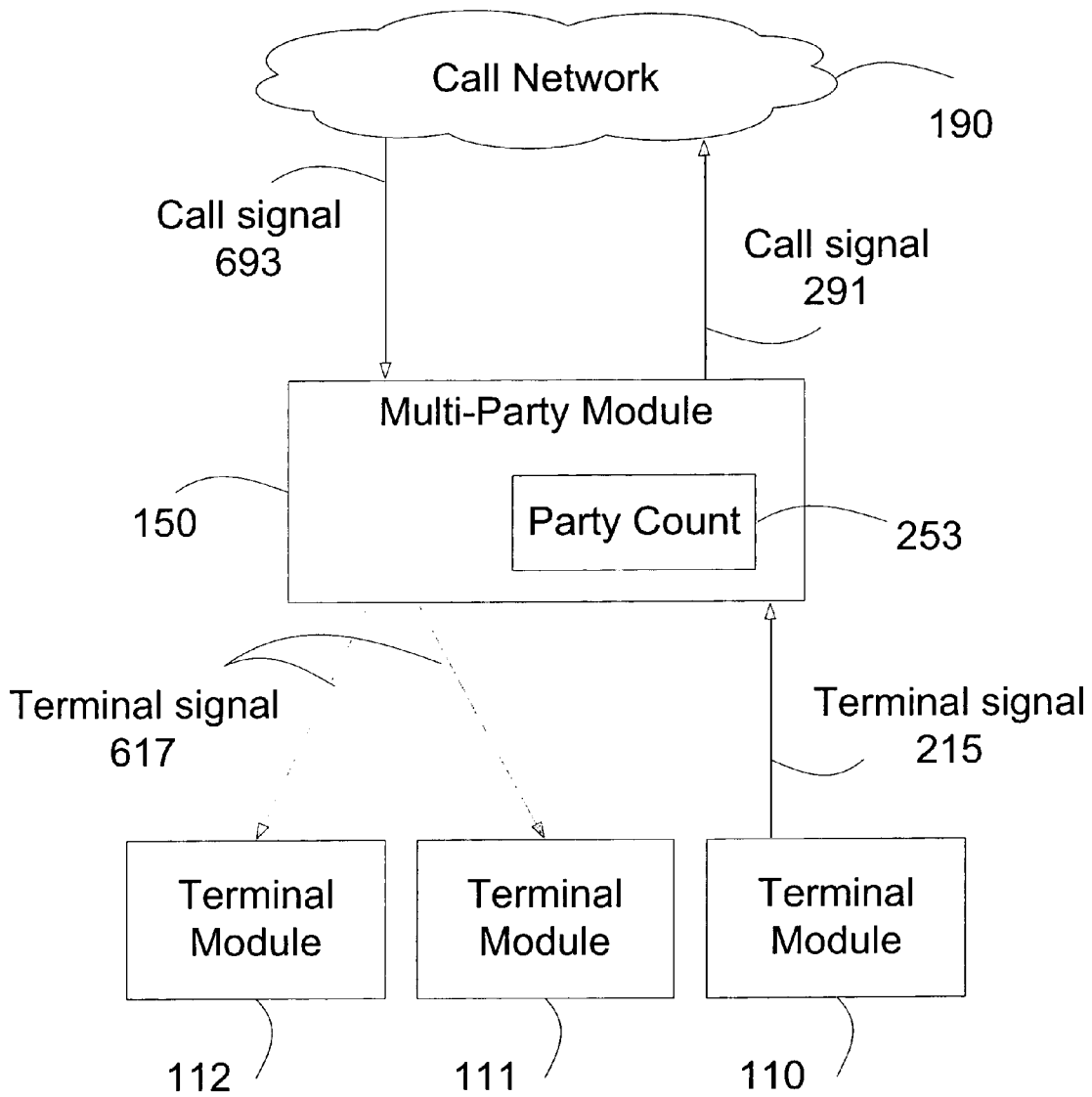
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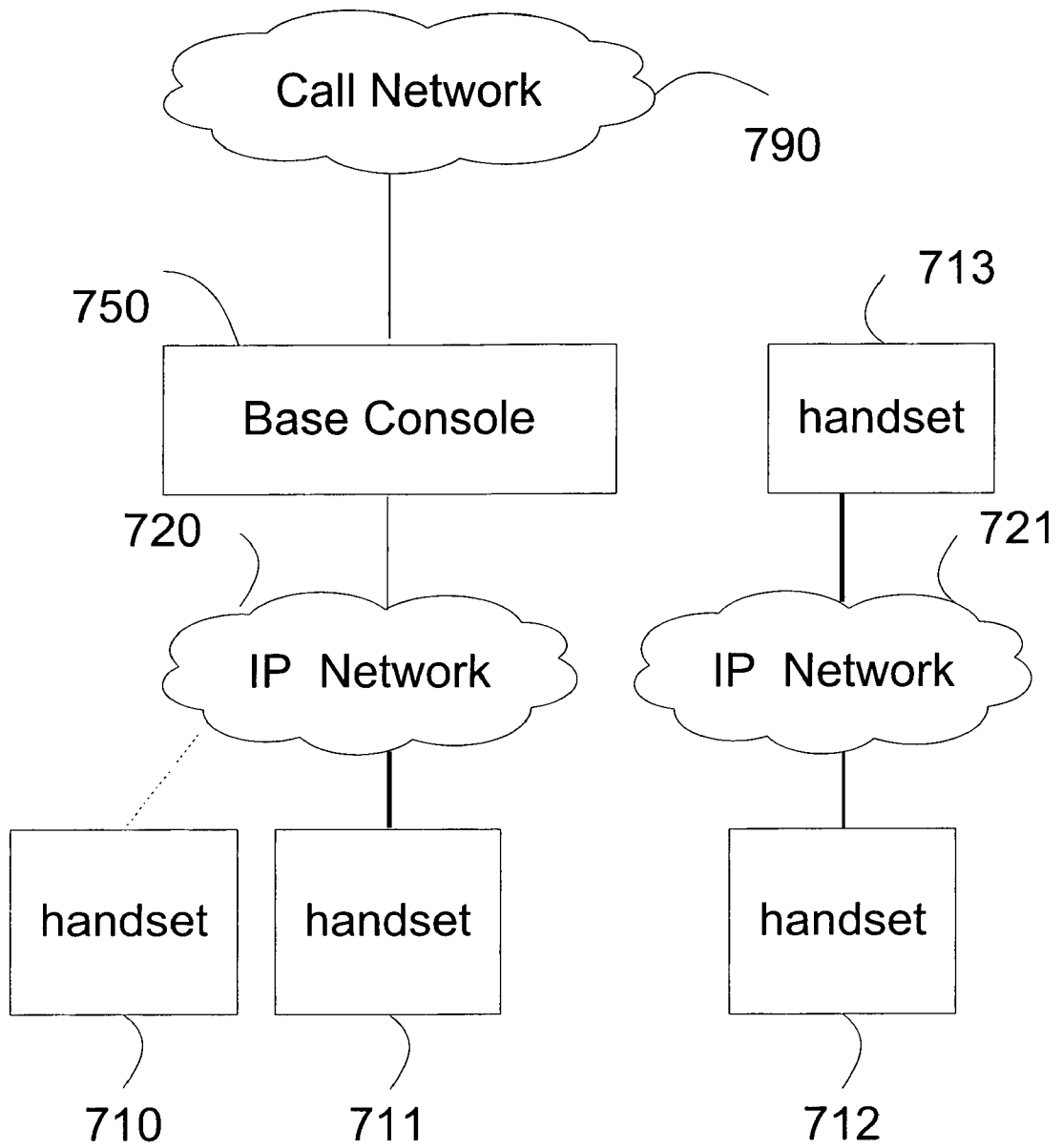handset

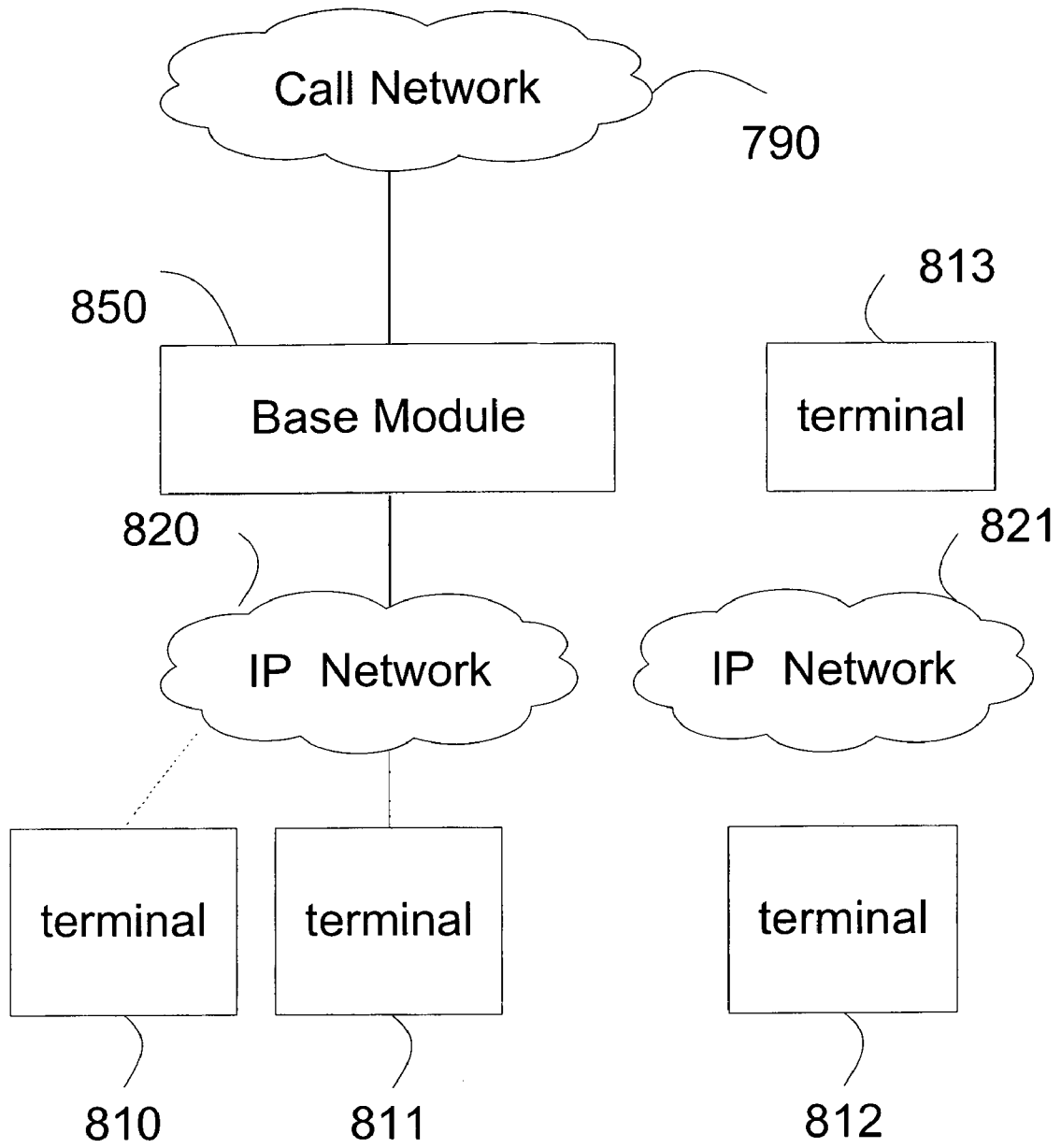710   711   712

Figure 7

Figure 8

METHOD AND SYSTEM FOR MULTIPLE PARTY TELEPHONE CALL

BACKGROUND OF THE INVENTION

The multi-party line is a time honored voice communication feature that allows users using multiple terminals to participate in a telephone call. It was widely deployed as a public telephone service when cost of telephone equipment and telephone lines was high.

Multi-party lines have been employed in key systems for small businesses, and in hotel or motel guest rooms. The deployment of cordless telephony based on wireless technologies such as Digital Enhanced Cordless Telecommunication ("DECT") and Communication Technologies 2 ("CT2") has fueled the continued use of multi-party lines.

With the advancement of Internet technologies, Internet Protocol ("IP") networks are becoming an important means for voice and data communications. Internet service providers and equipment vendors roll out successful voice communication services and products at a breathtaking pace. Many of these services and products are based on various Voice over IP ("VoIP") technologies, such as H.323 and Session Initiation Protocol ("SIP"). However, these VoIP technologies, limited by their reliance on endpoint call state management and their lack of a centralized call processing architecture, are inherently unsuitable for supporting multi-party lines. The lack of a centralized call processing architecture is problematic for supporting multi-party calls because the control of each individual telephone call is concentrated at the call's origination and destination, and once the call is established, an additional party has no centralized call processing entity to request that they be joined to the existing telephone call from. Thus, in attempting to join the existing telephone call in such an endpoint state management system, the additional party will only get a busy signal. The following examples are illustrative of various limitations VoIP technologies have with respect to multi-party lines.

In one example, a caller calls a multi-party line by alerting the multiple telephone terminals of the multi-party line. In a non-VoIP environment, when a first terminal answers the call, the centralized call processing center then removes the call alert from the other telephone terminals and establishes the multi-party call. In the case of a VoIP call, when a first terminal of the multi-party line answers the call, since there is no centralized call processing center, there is no feasible method to notify the other telephone terminals of the multi-party line to remove the call alert, thus the multi-party line call cannot be established.

In another example, a first terminal is conducting a conversation over a VoIP call. When a second terminal of the multi-party line desires to join the call, there is no feasible method to allow the second terminal to join the conversation. For example, if the second terminal attempts to call the first terminal, it receives a busy signal. Under systems using a centralized call processing center, the center is able to verify the second terminal as authorized on the multiple-party line call, and allow them to join in the already-initiated call.

In another example, a first user subscribes to a single-line residential telephone service. The first user connects the base station of her old 2.4 GHz multi-handset to the telephone line, with one handset in the kitchen, and the other handset in her daughter's bedroom. With this set up, the first user and her daughter are able to participate in a telephone call using the two handsets. Impressed with the low cost and the promise of convenience and other advanced features, the first user decides to switch to a 2-line residential VoIP telephone ser-

vice. Bundled with the service are two brand new VoIP telephones. Each VoIP telephone includes an independent VoIP application, allowing the first user and her daughter to make separate telephone calls at the same time. While enjoying the new found convenience, the first user is unaware of the limitation that the multi-party line feature no longer works. While conducting a conversation with second user, such as a relative, over the VoIP telephone in the kitchen, the first user wants her daughter to join the conversation using the VoIP telephone in her room. When the daughter picks up her phone, she is unable to join the conversation, since there is now no centralized call processor. Instead she receives a dial tone, prompting her to make a new call.

In still another example, a user owns a small business and subscribes to a single-line business telephone service. The user uses a telephone connector to connect two telephones to the telephone line. With this set up, the user can monitor the way his employee handles customer calls, and at time can join the conversation when necessary. With the expansion of his business, the user hires two new employees and switches to a 4-line VoIP business telephone service. With the new setup, the user and his employees can simultaneously handle multiple customer calls. However, the user can no longer monitor his employees' performance, and offer his help.

In another example, a hotel has upgraded its guest rooms to use VoIP technology. Each telephone runs a separate VoIP application, and each connects to the hotel's IP-Public Branch Exchange (IP-PBX) to receive telephone service. A user is a frequent patron of the hotel. Before the upgrade, she was able to use any telephone in the guest room to engage in a call. Unaware of the recent upgrade, she checks into the hotel the night before an important strategic presentation to her customers. She checks into an executive suite that features multiple VoIP telephones, one at the night stand in the bedroom, one on a desk in the adjacent living quarter, and one in the bathroom. At two o'clock a.m., the telephone rings. She wakes up and picks up the call with the VoIP telephone at the night stand and speaks to her boss regarding critical changes in the customer presentation. After talking for a minute, she decides to move to the living quarter in order to review the presentation in her laptop while expecting to resume the conversion with her boss with the VoIP telephone at the desk. But when she picks up the VoIP telephone, she is unable to speak to or hear from her boss, because there is no centralized call processor tying the telephone lines in the room together into a multi-party line.

The foregoing examples demonstrate the need to provide multi-party line features over an IP network.

SUMMARY OF THE INVENTION

An aspect of the present invention provides a multiple party telephone call method. The method includes receiving an incoming voice telephone call at a multi-party module, and allowing, by the multi-party module, for a plurality of terminal modules to participate in the incoming voice telephone call. Each terminal module is associated with a telephonic end-user device, such as a telephone handset, a video telephone, or a facsimile machine.

In another aspect of the invention, the voice telephone call in the above method is a Plain Old Telephone Service ("POTS") telephone call.

In another aspect of the invention, the voice telephone call in the above method is a cellular telephone call.

In another aspect of the invention, the voice telephone call in the above method is a VoIP telephone call.

In another aspect of the invention, the voice telephone call in the above method is offered by an Instant Messaging ("IM") service.

In another aspect of the invention, the voice telephone call in the above method is offered by a Private Branch Exchange ("PBX").

In another aspect of the invention, the voice telephone call in the above method is offered by an Internet Protocol Private Branch Exchange ("IP-PBX").

In another aspect of the invention, the method further includes incrementing a zero-initialized counter each time a terminal module begins participation in the voice telephone call.

Another aspect of the invention includes the above method, further decrementing the counter each time a terminal module ends participation in the voice telephone call, and terminating the voice telephone call by the multi-party module when the counter returns to zero.

In another aspect of the invention, the multi-party module may terminate the incoming voice telephone call.

Another aspect of the present invention provides a multiple party telephone call system. The system includes a multi-party module in communication with a call interface, such as a call network, and also in communication with a plurality of terminal modules, each terminal module comprising a telephone, and the multi-party module comprising a processor and associated memory. The processor and memory are configured to receive an incoming voice telephone call from the call network at the multi-party module, and allow for a plurality of terminal modules to participate in the incoming voice telephone call.

Another aspect of the invention includes the system above, in which the processor and associated memory of the multi-party module are further configured to increment a zero-initialized counter when a terminal module begins participation in a voice telephone call, and to decrement the counter when a terminal module ends participation in a telephone call.

In another aspect of the invention, the processor and associated memory of the multi-party module are further configured to disconnect the voice telephone call when the counter returns to zero.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a schematic diagram illustrating a multi-party module communicating to a call network and a plurality of terminal modules, in accordance with an embodiment of the invention;

FIG. 2 is a schematic diagram illustrating the processing of an incoming voice telephone call by an exemplary multi-party module;

FIG. 3 is a schematic diagram illustrating the processing of an outgoing voice telephone call by the multi-party module in accordance with an embodiment of the invention;

FIG. 4 is a schematic diagram illustrating a process for a terminal module to join an ongoing voice telephone call in accordance with an embodiment of the invention;

FIG. 5 is a schematic diagram illustrating a process for a terminal module to leave an ongoing voice telephone call, in accordance with an embodiment of the invention;

FIG. 6 is a schematic diagram illustrating a process for a terminal module to leave an ongoing voice telephone call, in accordance with an embodiment of the invention;

FIG. 7 is a schematic diagram illustrating a multi-handset system that allows multiple handsets to participate in a voice call provided by a public voice communication service, in accordance with an embodiment of the invention; and

FIG. 8 is a schematic diagram illustrating a multi-terminal system that allows multiple terminals to participate in a voice call provided by a private voice communication service, in accordance with an embodiment of the invention.

DETAILED DESCRIPTION

In the following description, for purposes of explanation, specific numbers, materials and configurations are set forth in order to provide a thorough understanding of the invention. It will be apparent, however, to one having ordinary skill in the art, that the invention may be practiced without these specific details. In some instances, well-known features may be omitted or simplified so as not to obscure the present invention. Furthermore, reference in the specification to "one embodiment" or "an embodiment" means that a particular feature, structure or characteristic described in connection with the embodiment is included in at least one embodiment of the invention. The appearances of the phrase "in an embodiment" in various places in the specification are not necessarily all referring to the same embodiment.

FIG. 1 schematically illustrates a multi-party module 150 communicating to a call network 190 and a plurality of terminal modules 110, 111, 112, in accordance with an embodiment of the invention. The plurality of terminal modules 110, 111, and 112, as used in the foregoing description of embodiments of the invention, are also intended to include the cases of either a single, only two, or more than three terminal modules, where appropriate in context.

In an embodiment of the invention, a multi-party call system includes a call network 190, which includes a network that supports a call such as but not limited to a voice telephone call. A multi-party module 150 communicates with the call network 190, and with one or more terminal modules 110, 111, 112 with which it is connected. Terminal modules 110, 111, 112 each are in turn interfaced to a telephone end-user device, such as a handset or speakerphone (not depicted). Terminal modules 110, 111, 112 and multi-party module 150 each include a microprocessor and associated memory configured to perform the methods described herein. For example, the multi-party module 150 and terminal modules 110, 111, 112 may each be personal computers in communication with each other via a network connection, and the multi-party module may also communicate with the call network 190 over a network, such as but not limited to the Internet. In other examples, the terminal modules 110, 111, 112 can each be a customized stand-alone unit having the requisite processor, memory and programming to perform the methods required of terminal modules, or the terminal modules may be integrated with telephone end-user device, such as but not limited to telephone handsets. Likewise, the multi-party module 150 can be a stand-alone unit or integrated with another unit which provides access to the call network 190 and terminal modules 110, 111, 112.

While it is preferred in an embodiment for the multi-party module 150 to be co-located with the group of terminal modules 110, 111, 112 with which it is in communication, this is not required. The multi-party module 150, and even the terminal modules 110, 111, 112 themselves, can be located in a different physical location from their connected end-user devices and from each other, although co-location does provide certain advantages well known to those of ordinary skill in electronics, such as, but not limited to, reduced signal degradation with shorter communication pathways. It is preferable to use a chipset that supports multi-mode voice applications, for example, a chipset that includes Global System for Mobile Communications ("GSM") and Voice over Wi-Fi

("VoWi-Fi") capability, such as Intel's PXA™ chipset; or a chipset that supports Voice over IP and cellular applications, such as those based on Texas Instrument's Wireless Any-Network Digital Assistant ("WANDA") architecture.

The call network **190** can include but is not limited to a public switched telephone network ("PSTN"), a cellular network, a corporate telephone network, an Internet Protocol ("IP") network, the Internet, a corporate Virtual Private Network ("VPN"), a Voice over IP ("VOIP") network, or any combination of these.

In an embodiment, multi-party module **150** functions to process a voice telephone call between the multi-party module **150** and call network **190**. The voice telephone call can be a Plain Old Telephone Service ("POTS") telephone call offered by a residential telephone service provider, a cellular telephone call offered by a mobile telephone service provider, a VOIP call offered by an Internet telephony service provider, such as Vonage, or Skype, a call offered by an Instant Messaging ("IM") service, such as Yahoo Messenger, MSN Messenger, AOL AIM or GoogleTalk service, a call offered by a Private Branch Exchange ("PBX"), an Internet Protocol Private Branch Exchange ("IP-PBX"), or a softphone call.

A softphone is software that simulates a telephone and runs on a general purpose computer. Typically, a softphone supports VoIP communications. Examples of a softphone are CounterPath's eyeBeam™, Pingtel's sipXphone™ and Nortel's i2050 Software Phone™.

Examples of the multi-party module **150** processing a voice telephone call include the multi-party module **150** receiving an incoming voice telephone call from the call network **190**, making an outgoing voice telephone call to the call network **190**, disconnecting an incoming voice telephone call, or disconnecting an outgoing voice telephone call. The multi-party module **150** can process a voice telephone call by sending call signals to, and receiving call signals from the call network **190**. These call signals together represent the actual voice telephone call itself, usually in a compressed format, plus any information needed at the terminal end-user devices to restore the voice telephone call to its original acoustic sound.

Call signals used by the invention can be based on a standard protocol or an extension of a standard protocol, such as POTS protocol, Global System for Mobile Communications ("GSM") protocol, Session Initiation Protocol ("SIP"), H.323 protocol, Megaco protocol, a SIP extension protocol, an H.323 extension protocol, a Session Initiation Protocol for Instant Messaging and Presence Leveraging Extensions ("SIMPLE") based protocol, an Extensible Message and Presence ("XMPP") based protocol, or may be based on a proprietary protocol, such as Skype protocol, or a protocol used in a PBX or a key system.

Communication between the multi-party module **150** and the terminal modules **110, 111, 112** can be over a packet data network. The packet network can be an IP network, and may include the Internet, or a corporate VPN, or a wired IP network, such as an Ethernet Network, or a wireless IP network, such as a Wireless Fidelity ("WiFi") network, or a Bluetooth network, or a hotspot wireless network.

The multi-party module **150** includes the functionality of allowing the terminal modules **110, 111, 112** to participate in a voice telephone call that is already present between the multi-party module **150** and the call network **190**. This functionality may be provided in several ways. For example, once a voice telephone call is present between the multi-party module **150** and the call network **190**, the multi-party module can send a signal to the terminal modules **110, 111, 112** indicating there is a call in progress. In turn, one or more of the

terminal modules may cause their associated telephone end-user device, such as a handset (not depicted), to signal an end user that the call is in progress. In response, the end user may next use their telephone end-user device to attempt to join the voice telephone call, and the telephone end-user device can then communicate with its associated terminal module **110**, which, in turn, may send a terminal signal to the multi-party module **150**, which may then allow the user to join the voice telephone call in progress.

In another embodiment, a terminal module **110, 111, 112** can be associated with other telephone end-user devices, such as, but not limited to fax machines, answering machines, cordless handsets or personal computers.

In a preferred embodiment, the multi-party module **150** mediates the voice signals from the call network **190** and from the terminal modules **110, 111, 112**. The voice signals from the call network **190** represent the voice of a remote user of the voice telephone call. The voice signals from terminal modules **110, 111, 112** represent the voice of a user using the telephone end-user devices associated with their respective terminal modules **110, 111, 112**. For example, the multi-party module **150** can combine the voice signals from the terminal modules **110, 111, 112** and send the combined voice signals to the call network **190**. In another example, the multi-party module **150** sends the voice signals from the call network **190** to the each of the connected terminal modules **110, 111, 112**.

As previously mentioned, the multi-party module **150** may convert voice signals to a standard codec format before sending them to either the call network **190** or to connected terminal modules **110, 111, 112**. For example, in one embodiment, the multi-party module can transmit and receive voice signals over a packet data network, such as an IP network, to connected terminal modules **110, 111, 112**. In other embodiments, the voice signals can be sent in Real-time Transport Protocol ("RTP") packets, or may be represented by a speech codec, such as G.711, G.723.1 and G.729, or may be represented by a wide bandwidth codec that encodes high fidelity voice signals, such as Moving Picture Experts Group Layer 3 ("MP3") or Windows Media Audio ("WMA"), or by a wide bandwidth codec that encodes multiple channels of high fidelity voice signals. In another embodiment, the multi-party module **150** may transmit and receive voice signals over a second network, such as, but not limited to, a network based on CT2 or DECT technology.

The multi-party module **150** can connect to one or more terminal modules **110, 111, 112** in order to allow the users of the telephone end-user device associated with each of the terminal modules **110, 111, 112** to participate in a voice telephone call. The multi-party module **150** can communicate with one or more terminal modules **110, 111, 112** by establishing a communication session between the multi-party module **150** and each of the connected terminal modules **110, 111, 112**. For example, the communication session may include a TCP session between the multi-party module **150** and the connected terminal modules **110, 111, 112**. Such a communication session may be initiated by either the terminal modules **110, 111, 112** or the multi-party module **150**. In another example, the communication session includes an UDP session between multi-party module **150** and each of the connected terminal modules **110, 111, 112**, which may be initiated by either the multi-party module **150** or the connected terminal modules **110, 111, 112**.

In an exemplary embodiment of the invention, the multi-party module **150** receives a request from one or more of the terminal modules **110, 111, 112** to register for participation in an ongoing voice telephone call. The request may include the IP address of the one or more terminal modules **110, 111, 112**,

as well as information relating to the terminal module's voice telephone call participation capabilities, such as supported media codecs, authentication credentials or security keys, etc.

After multi-party module **150** processes the request, the multi-party module **150** may then connect to the one or more terminal modules **110, 111, 112**. This connection, as previously described, may entail the establishment of a communication session with the terminal modules **110, 111, 112**, in which the multi-party module **150** can send signals to, and receives signals from the connected terminal modules **110, 111, 112** in subsequent communication for voice telephone call participation.

For example, in an embodiment, the communication session can include an IP multicast session between the multi-party module **150** and the connected terminal modules **110, 111, 112**, in which the multi-party module **150** may include a multicast IP address, and the multi-party module **150** may associate the IP address of each of the terminal modules **110, 111, 112** with this multicast IP address.

FIG. **2** is an exemplary illustration of processing an incoming voice telephone call, represented by the call signal **291**, by a multi-party module **150**. The multi-party module **150** connects to the terminal modules **110, 111**, and also maintains an internal counter, party count **253**, indicating the number of terminal modules **110, 111** that are participating in the incoming voice telephone call. When the multi-party module **150** receives the call signal **291** from the call network **190** indicating the arrival of the incoming voice telephone call, the multi-party module **150** sets its party count **253** to "0". Preferably, the multi-party module **150** then composes a terminal signal **215** for alerting for the terminal modules **110, 111** of the incoming voice telephone call. The multi-party module **150** then sends the terminal signal **215** to the terminal modules **110** and **111**. The terminal signal **215** may also include an instruction for the terminal modules **110, 111** to generate an alerting signal, such as a ring tone, to their associated telephone end-user device (not depicted).

FIG. **3** illustrates the processing of an outgoing voice telephone call by the multi-party module **150** in accordance with an embodiment of the invention. The multi-party module **150** is connected to terminal modules **110** and **111**, as described above. As previously described, the multi-party module **150** can include a counter, party count **253**, to track the number of terminal modules **110, 111** that are involved in the outgoing voice telephone call. For example, prior to the outgoing voice telephone call being present, the multi-party module's party count **253** has an initial value of "0".

In an embodiment, in order to establish the outgoing voice telephone call, the multi-party module **150** receives a terminal signal **215** from terminal module **110**, indicating a request to start a voice telephone call. Next, the multi-party module **150** sets Party Count **253** to "1", composes a terminal signal **316** for indicating the readiness for initiating an outgoing voice telephone call, and sends the terminal signal **316** to terminal module **110**. The terminal signal **316** may include an instruction for the terminal module **110** to generate a dial tone to its associated telephone end-user device (not depicted). Next, the multi-party module **150** receives from terminal module **110** a terminal signal **317**, which preferably includes calling information necessary for establishing the telephone call, such as a telephone number. In various other embodiments, the calling information may include a name or a Universal Resource Identifier ("URI"). Next, multi-party module **150** composes a call signal **291** for initiation of an outgoing voice telephone call based on the calling information, and sends the call signal **291** to the call network **190** to initiate the outgoing voice telephone call.

FIG. **4** illustrates a process for a terminal module **111** to join an ongoing voice telephone call in accordance with an embodiment of the invention. The multi-party module **150** is connected with terminal modules **110** and **111** in a manner such as described above, and an ongoing voice telephone call (not depicted) is present between the call network **190** and the multi-party module **150**, after multi-party module **150** receives a call signal **491** indicating the arrival of an incoming voice telephone call. The ongoing voice telephone call may alternatively be initiated after the multi-party module **150** initiates it. In either case, the multi-party module can maintain a counter indicating the number of terminal modules participating in the voice telephone call, party count **253**, which is initialized to

Next, in an embodiment, the multi-party module **150** receives from a terminal module **111** a terminal signal **215** indicating a request to join the voice telephone call, and, in response, the multi-party module **150** can increase the party count **253** by "1".

In another embodiment, the ongoing voice telephone call is an incoming voice telephone call that has not yet been answered, and multi-party module **150** composes a call signal **291** for answering the incoming voice telephone call, and sends the call signal **291** to the call network **190** to answer the incoming voice telephone call.

FIG. **5** illustrates a process for a terminal module **110** to leave an ongoing voice telephone call, in accordance with an embodiment of the invention. Multi-party module **150** is connected to one or more terminal modules **110, 111, 112**, and an ongoing voice telephone call is present between the call network **190** and the multi-party module **150**. In an exemplary embodiment, terminal module **110** and **111** are involved in the ongoing voice telephone call, and the multi-party module has set its party count **253** to a value of "2". Next, the multi-party module **150** receives a terminal signal **215** from terminal module **110** indicating that user of the telephone end-user device associated with terminal module **110** wants to leave the voice telephone call. In response, multi-party module **150** decreases party count **253** by 1, leaving party count **253** with an after-decrease value of "1".

FIG. **6** illustrates a process whereby an ongoing voice telephone call is terminated by the leaving of a terminal module **110** from participating in the call, in accordance with an embodiment of the invention. The multi-party module **150** is connected to terminal modules **110, 111** and **112**, and terminal module **110** is involved in an ongoing voice telephone call. The multi-party module **150** thus has set its party count **253** to a value of "1". Next, the multi-party module **150** receives from terminal module **110** a terminal signal **215** indicating a request to leave the voice telephone call. The multi-party module **150** decreases party count **253** by 1, leaving its party count **253** with an after decrease value of "0", indicating that no terminal modules are involved in the ongoing voice telephone call. In an embodiment, the multi-party module **150** then composes a call signal **291** for disconnecting the voice telephone call, and sends it to the call network **190** to disconnect the ongoing voice telephone call.

In another embodiment, terminal modules **111** and **112** are both involves in an ongoing voice telephone call, and party count **253** has a value of "2". When the multi-party module **150** then receives a call signal **693** from the call network **190** indicating the disconnecting of the ongoing voice telephone call, the multi-party module **150** sets party count **253** to "0". The multi-party module **150** may then compose a terminal signal **617** indicating the disconnection of the ongoing voice telephone call, and sends terminal signal **617** to terminal modules **111** and **112**.

FIG. 7 is a schematic diagram illustrating a multi-handset system that allows multiple handsets to participate in a voice call provided by a public voice communication service. In an embodiment of the invention, a multi-handset system allows multiple handsets to participate in a voice telephone call provided by a public voice communication service. For example, a multi-handset system may include a base console 750, which includes a multi-party module. The base console connects to a plurality of handsets 710, 711, 712, and a handset may include a terminal module.

Further, the base console of a multi-handset system may connect to a call network 790 to receive a voice communication service. A multi-handset system may be used in a home, and the voice communication service is a residential telephone service. Alternatively, the multi-handset system is used in a small business environment, and the voice communication service is a business telephone service or a collaboration service that includes voice communication.

A base console 750 may connect to a cellular base station, such as a Global System for Mobile Communications ("GSM") base station or a Code Division Multiple Access ("CDMA") base station, and the base console can receive cellular telephone service over the base station. Alternatively, the base console of a multi-handset system can connect to a broadband gateway, and the base console may receive an IP-based voice communication service over the broadband gateway. Examples of the IP-based voice communication service are a Voice Over IP ("VoIP") service such as Vonage™, or Skype™, an IP centrex service, or a service based on Peer-to-Peer Instance Messaging ("IM") based technology, such as Yahoo Messenger™, MSN Messenger™, or GoogleTalk™.

Base console 750 may communicate with a plurality of handsets 710, 711, 712, 713 by sending terminal signals to and receiving terminal signals from the plurality of handsets 710, 711, 712, 713, allowing the plurality of handsets to participate in a voice telephone call.

In an embodiment, IP network 720 and IP network 721 may be the same home network, base console 750 is in the kitchen, handset 710 is in the living room; handset 711 is in the bedroom, and handset 712 is in the family room. In another embodiment, handset 713 attaches directly to base console 750 via a data interface, such as Universal Serial Bus (USB) or Bluetooth interface.

In another embodiment, IP network 720 is a home network and IP network 721 is a corporate IP network, such as a corporate VPN. IP network 720 and IP network 721 may be connected over the Internet. Base console 750, handset 710 and handset 711 may be in the home, handset 712 may be in an office. Handsets 710, 711, 712 and 713 can participate in a voice call between base console 750 and the public voice communication service.

In yet another embodiment, IP network 720 is an office IP network in a corporate headquarters and IP network 721 is an office IP network in a branch office. IP network 720 and IP network 721 may be are connected over the Internet or a corporate VPN. Base Console 750, handset 710 and handset 711 may be in the corporate head quarter and handset 712 may be in the branch office. Handset 710, 711, 712 and 713 can participate in a voice call between base console 750 and the public voice communication service.

A multi-handset may be used in a small business, such as a restaurant, a travel agency or a small retail store.

FIG. 8 is a schematic diagram illustrating a multi-terminal system that allows multiple terminals to participate in a voice call provided by a private voice communication service.

A multi-terminal system allows multiple terminals to participate in a voice telephone call provided by a private voice communication service. A multi-terminal system can include a base module.

A base module 850 of a multi-terminal system preferably includes a multi-party module, and connects to a plurality of terminals 810, 811, 812, 813. Further, the base console of a multi-terminal system may connect to a call network 790 to receive a voice communication service.

In at least one embodiment a base module receives private voice communication service from a Public Branch Exchange ("PBX"), an IP-PBX, a key system, or a corporate collaboration service based on IM technologies.

In an embodiment, multiple terminals 810, 811, 812, 813 may connect to the base module 850 over an IP network to participate in a voice telephone call between base module 850 and a private voice communication service. For example, base module 850 may communicate with the terminals 810, 811, 812, 813 by sending terminal signals to and receiving terminal signals from the terminals 810, 811, 812, 813 over an IP network, allowing the terminals 810, 811, 812, 813 to participate in a voice telephone call.

In an embodiment, IP network 820 and IP network 821 may be the same network inside a hotel room. Base module 850 may be located at a night stand next to a bed, terminal 810 may be located in the living room next to a television, terminal 811 may be located in the bathroom, terminal 812 may be located on a desk, and terminal 813 may attach directly to base module 850 via a data interface, such as a Universal Serial Bus ("USB") or Bluetooth™ interface. Terminals 810, 811, 812, 813 can participate in a voice telephone call between base module 850 and a private communication service.

Alternatively, IP network 820 may be an office network in a corporate headquarters, and IP network 821 may be an office network in a branch office. IP network 820 and IP network 821 may be connected over the Internet or a corporate VPN. Base module 850, terminal 810 and terminal 811 may be in the corporate headquarters and terminal 812 may be in the branch office. The terminals 810, 811, 812, 813 can participate in a voice telephone call between base module 850 and a private voice communication service.

In another embodiment, IP network 820 may be an office network in an office area, and IP network 821 may be a laboratory network in a laboratory. IP network 820 and IP network 821 may be connected over a corporate IP network. The base module 850, terminal 810 and terminal 811 may be in the office area, and terminals 812 and 813 may be in a laboratory area. Terminals 810, 811, 812 and 813 can participate in a voice call between the base module 850 and a private voice communication service.

In exemplary systems, a multi-terminal system is used in a hotel or motel guest room, an office inside a campus, such as a laboratory, a warehouse, or a guard position, or during an event, for example, in a ticket office, a concession booth, a first aids station.

Although the invention herein has been described with reference to particular embodiments, it is to be understood that these embodiments are merely illustrative of the principles and applications of the present invention. It is therefore to be understood that numerous modifications may be made to the illustrative embodiments and that other arrangements may be devised without departing from the spirit and scope of the present invention as defined by the appended claims.

The invention claimed is:

1. A multiple party telephone call system in communication with a first Internet Protocol ("IP") data network, comprising:

a multi-party module in communication with a call interface associated with the first IP data network, and facilitating communications for a plurality of terminal modules that are accessible only via a single multi-party phone line that is separate from the first IP data network, each terminal module associated with a telephonic end-user device; and

the multi-party module comprising a processor and associated memory, the processor and memory configured to:

receive an incoming voice telephone call from the call interface, the call interface receiving the incoming voice telephone call over the first IP data network; and

allow for the plurality of terminal modules within the single multi-party phone line to participate in the voice telephone call simultaneously wherein the multi-party module communicates with one or more additional terminal modules of the plurality of terminal modules over at least one additional IP data network and with at least another terminal module of the plurality of terminal modules over a second IP data network disparate from the first IP data network and the at least one additional IP data network.

2. The system according to claim 1, the processor and associated memory of the multi-party module further configured to increment a counter when a terminal module begins participation in a voice telephone call, and to decrement the counter when a terminal module ends participation in a telephone call.

3. The system according to claim 2, the counter being initialized to zero.

4. The system according to claim 3, the processor and associated memory of the multi-party module further configured to disconnect the voice telephone call when the counter returns to zero.

5. The system according to claim 1, the telephonic end-user device being selected from the group comprising: a telephone handset, a video telephone, a facsimile machine, a Voice over Internet Protocol ("VOIP") telephone, or a softphone.

6. The system according to claim 1, wherein each of the plurality of terminal modules within the single multi-party phone line is operable to transmit a request to the multi-party module to participate in the voice telephone call.

7. The system according to claim 1, wherein each of the terminal modules is operable to receive a request from another terminal module at the multiparty module, the request including an IP address of at least one of the plurality of the terminal modules participating in the ongoing voice telephone call.

8. The system according to claim 1 wherein the multiparty module facilitates communication for the plurality of terminal modules such that all the terminal modules are alerted upon the multiparty module receiving a call signal for the incoming voice telephone call.

9. A method for permitting multiple party telephone calls among terminal modules within a single multi-party phone line wherein the single multi-party phone line is in connection with a first Internet Protocol ("IP") data network and is separate from the first IP data network comprising:

receiving an incoming voice telephone call at a multi-party module that facilitates communications for the terminal

modules that are accessible only via the single multi-party phone line, from a call interface associated with the first IP data network;

sending a signal from the multi-party module to at least one of the terminal modules, connected to the multiparty module via another IP data network, that there is a call in progress;

sending a signal from the at least one terminal module to an end user that there is a call in progress;

sending a signal from the at least one terminal module to the multi-party module; and

allowing, by the multi-party module, for the at least one terminal module and a second one of the terminal modules within the single multi-party phone line connected to the multi-party module via a second IP network, disparate from the first IP data network and the other IP data network, to participate in the incoming voice telephone call simultaneously such that the at least one and the second terminal modules are allowed to join and participate in an ongoing voice telephone call upon receiving the incoming call, each terminal module associated with a telephonic end-user device.

10. The method according to claim 9, in which the incoming voice telephone call is a VoIP telephone call.

11. The method according to claim 9, in which the incoming voice telephone call is offered by an Instant Messaging ("IM") service.

12. The method according to claim 9, in which the incoming voice telephone call is offered by an Internet Protocol Private Branch Exchange ("IP-PBX").

13. The method according to claim 9, further comprising incrementing counter each time a terminal module begins participation in the call.

14. The method according to claim 13, the counter being initialized to zero.

15. The method according to claim 14, further comprising:

decrementing the counter each time a terminal module ends participation in the call; and

terminating the call by the multi-party module when the counter returns to zero.

16. The method according to claim 9, further comprising terminating the incoming voice telephone call by the multi-party module.

17. The method according to claim 9, wherein the multi-party module communicates with one or more of the plurality of terminal modules over one or more additional IP data networks.

18. A method for permitting multiple party telephone calls among a plurality of terminal modules within a single multi-party phone line wherein the single multi-party phone line is in connection with a first Internet Protocol ("IP") data network and is separate from the first IP data network comprising:

transmitting a signal from one of the terminal modules to a multi-party module facilitating communication for the terminal modules requesting to start a voice telephone call, wherein the plurality of terminal modules are accessible only via the single multi-party phone line;

composing a signal indicating readiness for initiating an outgoing voice telephone call, and sending the signal to a terminal module connected to multi-party module via another IP data network that is disparate from the first IP data network ;

receiving from the terminal module connected via the other IP data network at the multi-party module a signal including calling information necessary for establishing the voice telephone call;

composing a call signal for initiation of an outgoing voice telephone call based on the calling information,

sending the call signal to the first IP data network to initiate an outgoing voice telephone call by the multi-party module;

sending the outgoing voice telephone call from the multi-party module over the first IP data network; and

allowing, by the multi-party module, for at least a second one of the plurality of terminal modules within a single multi-party phone line connected to the multi-party module via a second IP data network, disparate from the first IP data network and the other IP data network, to participate in the outgoing voice telephone call simultaneously such that at least second one of the terminal

modules is allowed to join and participate in an ongoing voice telephone call upon reception of the outgoing telephone call by a callee wherein each terminal module is associated with a telephonic end-user device.

19. The method according to claim **18**, the telephonic end-user device being a telephone handset.

20. The method according to claim **18**, the telephonic end-user device being a video telephone or a facsimile machine.

21. The method according to claim **18**, the voice telephone call being a VOIP telephone call.

22. The method according to claim **18**, wherein the multi-party module communicates with one or more of the plurality of terminal modules over one or more additional IP data networks.

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