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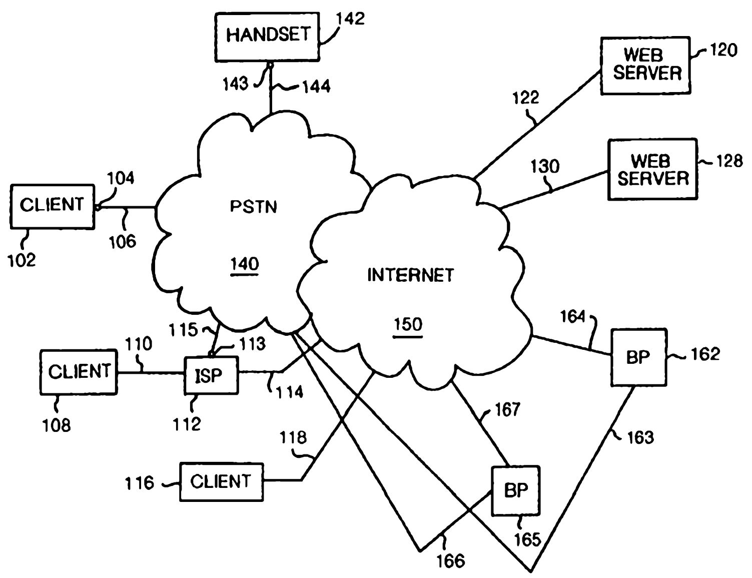

(57) Abstract

A method and apparatus for establishing a voice call to a PSTN extension on behalf of a networked client computer, and routing the voice call off of the network, is provided. In a first embodiment, the apparatus comprises a storage medium having stored therein a plurality of programming instructions for implementing a set of communication services for facilitating establishment of the voice call to the PSTN extension, and an execution unit, coupled to the storage medium, for executing the plurality of programming instructions. The set of communication services include services for soliciting inputs from one or more other apparatuses on one or more operating characteristics associated with the establishment and support of the voice call to the PSTN extension, and selecting either the apparatus itself or one of the other apparatuses to place the voice call to the PSTN extension.

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METHOD AND APPARATUS FOR SELECTING AN INTERNET/PSTN CHANGEOVER SERVER FOR A PACKET BASED PHONE CALL

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to the field of telecommunications and, in particular, to a method and apparatus for selecting an internet/PSTN changeover server for a packet based phone call.

2. Background Information

Numerous advances have been made in recent years in the field of telecommunications. In particular, the field of internet telephony has emerged as a viable technology that is evolving at an ever increasing rate. Evidence of this evolution of internet telephony is best characterized by the number of new products recently become available in the market. Products such as CoolTalk by Netscape Communications Corporation of Mountain View, California; Internet Connection Phone by International Business Machines of Amonk, New York; Intel Internet Phone (IPhone) by Intel Corporation of Santa Clara, California; NetMeeting by Microsoft Corporation, Redmond, Washington; Quarterdeck WebTalk by Quarterdeck Corporation of Marina Del Rey, California; TeleVox by Voxware Incorporated of Princeton, New Jersey; and WebPhone by Netspeak Corporation of Boca

Raton, Florida, are representative of the current state of applications facilitating interent telephony.

Each of these products offers internet based voice communications with a telephone motif, between two users each using the same (or compatible) product on either end of the internet connection. That is, the internet provides the "switching" architecture for the system, while the computer acts as the "handset", or the audio interface. One reason for the proliferation of these applications is a desire to push the technology of the internet to provide a total communications tool. The appeal to users is that, currently, the use of the internet is free of toll charges. Therefore, currently, a user of an internet phone product may communicate with another user located anywhere else in the world without having to pay the long distance charges associated with making a telephone call using the public switched telephone network (PSTN).

However, consumers expecting to completely eliminate their long-distance telephone bills through the exclusive use of internet telephony are in for a disappointment. As they shall soon discover, although innovative in their own right, the current internet based telephony applications identified above have a number of limitations which retard their acceptance as a primary communications tool. One such limitation is that many of the applications identified above require that both users have installed the same software package, or compatible packages and, therefore, provide a relatively low level of interoperability. One reason for this lack of interoperability between internet telephony applications is that the developers of many of these products have incorporated different voice encoders (commonly referred to as a "voice codec", or simply a "codec" by those in the telecommunication arts) in the products. Consequently, as a result of the different codecs used, many internet telephony applications are unable to recognize speech encoded (i.e., digitized) by a codec of a disimilar application.

This problem is alleviated for those products that are upgraded to comply with emerging telephony standards, such as International Telecommunication Union's (ITU) H.323 standard. However, other limitations remain. For example, another limitation associated with many of these products is that they are tied to the internet, often requiring all users to access a common server in order to maintain a directory of available users in which to call. That is to say, many of the applications identified above do not integrate the packet switched network of the internet with the circuit switched public switched telephone network (PSTN). Therefore, although a computer connected to the internet may communicate with another user on the internet, assuming they are both using a common software application (or at least applications with compatible codecs), these applications do not support communication with a user of a Telephone handset.

The reason for this limitation is readily understood by those who appreciate the complexity of the two networks. As alluded to above, the internet is a packet switched network. That is to say, communication over the internet is accomplished by "breaking" the transmitted data into varying-sized packages (or "packets"), based primarily on communication content, and interleaving the various-sized packages to best utilize the bandwidth available at any given time on the internet. When the packets reach their intended destination, they must be reassembled into the originally transmitted data. Loss of packets, and thus data, occur frequently in such a network, and the ability of the network to succesfully transmit information from one point in the network to another determines the quality of the network. For inter-computer communication transactions involving non real-time data, the ability to transmit packets and retransmit any packets that are perceived to have been dropped is not a severe limitation and may not even be perceived by the user of the system. However, in a voice communication transaction, the delay required to retransmit even one data packet may be perceived by a user. At best, such delays are an annoying inconvenience. In practice, the delays actually can become intolerable, as the cumulative latency adds up with successive transmissions.

In contrast to the packet switched network of the internet, the public switched telephone network (PSTN) is a circuit switched network. That is to say that the PSTN assigns a dedicated communication line to a user with which to complete the telephone call, wherein the user can utilize the assigned resource of the PSTN in any way they choose, with the understanding that the user is paying for the use of the dedicated resource of the PSTN. While the circuit switched approach of the PSTN system is not necessarily the most efficient system in terms of call traffic (i.e., it does not make use of the "dead space" common in a conversation), it is relatively easy to ensure that information destined for a particular user is delivered, it simply provides a dedicated line to complete the transaction.

Nonetheless, despite these engineering challanges, a few products have emerged which purport to integrate the PSTN to the internet. Products such as Net2Phone by IDT

Corporation of Hackensack, New Jersey, claim to provide a computer user with the ability to place and receive a phone call to/from a PSTN extension. Unfortunately, none of these products completely solve the problem of integrating the two networks. The limitations perhaps best characterized by way of an example communication session. With these prior art internet telephony applications, a user of an internet telephony enabled client computer initiating a telephone call to a Telephone handset launches the voice call from the client computer by accessing a server (the primary access server), operated by the developer of the internet telephony application that supports internet telecommunications. As the initiator accesses the primary access server, he/she is prompted for a destination address, which takes the form of a PSTN telephone number for an outgoing call to a Telephone handset. Having provided the primary access server with the PSTN telephone number associated with the Telephone handset, the primary server somehow determines[1] which server in a community of similarly enabled servers (i.e., servers with the hardware/software necessary to provide access to the PSTN) is closest to the destination address, and completes the telephone call by routing the telephone call through a number of intermediate servers on the internet to the selected server, which will actually place the voice call to the Telephone handset on behalf of the client computer, facilitating the voice call between the client computer and the Telephone handset. In other words, the user of the client computer is required to have prior knowledge of the destination phone number, which is limiting in many circumstances. For example, in a situation where the user of the client computer is engaged in a data communication session involving a webpage for a corporate entity, the user may wish to speak with someone in a "local office" of the corporate entity. Prior art internet telephony applications require that the

[1] The manner in which the "primary access server" determines the "call originating server" is not known.

telephone number for the "local office" of the corporate entity be provided by the user of the client computer in order to place the telephone call. If the telephone number for the "local office" of the corporate entity is not provided by the webpage, the user of client computer must look it up or have prior knowledge of it.

Additionally, while the prior art approach of simply finding the internet telephony enabled server closest to the destination address may offer the simplest technical solution and a seemingly cheaper connection, it does not ensure the quality of the voice connection. One skilled in the art will appreciate that there are a number of characteristics which may impact the quality of the voice connection. For example, insofar as the internet is a packet switched network, as the number of intermediate routers required to interface the client computer to the selected server increases so, too, does the likelihood that data packets containing voice information could be lost or corrupted. The result of lost or corrupted data packets is broken or garbled speech. Another factor affecting internet telephony communication performance is the bandwidth available on the selected server. If, for example, the selected server is very busy handling a number of other processes, the performance associated with each of the processes begins to degrade (i.e., slow down), which may also result in delayed delivery of data packets containing speech, which in turn results in user perception of poor quality. Therefore, while it is important to some users of internet telephony applications to simply keep the cost down, quality considerations must also be accounted for to enable internet telephony to evolve into a viable communications tool.

Thus, a need exists for a method and apparatus for selecting an internet/PSTN changeover server for a packet based phone call that is unencumbered by the limitations associated with the prior art.

SUMMARY OF THE INVENTION

In accordance with the teachings of the present invention, a method and apparatus for selecting an internet/PSTN changeover server for a packet based phone call is provided. In a first embodiment, the apparatus comprises a storage medium having stored therein a plurality of programming instructions for implementing a set of communication services for facilitating establishment of the voice call to the PSTN extension, and an execution unit, coupled to the storage medium, for executing the plurality of programming instructions. The set of communication services include services for soliciting inputs from one or more other apparatuses on one or more operating characteristics associated with the establishment and support of the voice call to the PSTN extension, and selecting either the apparatus itself, or one of the other apparatuses to place the voice call to the PSTN extension.

In one embodiment, the set of communication services further include services for calculating a call metric, based on the solicited inputs, for each of the responding apparatus and selecting an apparatus to facilitate the voice call based, in part, on the calculated call metric. In one embodiment, the communication services further include telephony services for cooperating with telephony equipment coupled to the apparatus. In one embodiment, the storage medium stores programming instructions for implementing web server functions. Similarly, in one embodiment, the storage medium stores programming instructions for implementing internet service provider functions.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be described by way of exemplary embodiments, but not limitations, illustrated in the accompanying drawings in which like references denote similar elements, and in which:

Figure 1 is a block diagram illustrating an exemplary communication system incorporating the teachings of the present invention;

Figure 2 is a flow chart illustrating a method for establishing a voice communication session with a PSTN extension for a networked client computer, in accordance with one embodiment of the present invention;

Figure 3 is a block diagram illustrating an exemplary server suitable for use as a changeover server of the present invention; and

Figure 4 is a block diagram illustrating the software architecture of the changeover server of **Figure 3**, in accordance with one embodiment of the present 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 present invention. However, it will be apparent to one skilled in the art that the present invention may be practiced without the specific details. In other instances, well known features are omitted or simplified in order not to obscure the present invention. Furthermore, for ease of understanding, certain method steps are delineated as separate steps, however, these separately delineated steps should not be construed as necessarily order dependent in their performance.

Referring now to **Figure 1**, a block diagram is presented illustrating an exemplary communication system **100** incorporating the teachings of the present invention for connecting a user of an internet telephony enabled client computer with a user of a PSTN endpoint (e.g., telephone handset), and routing the voice call off of the internet. While the present invention will be described in the context of this exemplary computer system, based on the descriptions to follow, those skilled in the art will appreciate that the present invention is not limited to this embodiment, and may be also practiced with intranet (in lieu of the internet) and/or automated/computerized telephony answering equipment (in lieu of telephone handsets).

For the illustrated embodiment, client computer **102** incorporated with the teachings of the present invention, while in data communication with a web server, e.g. web server **128**, through PSTN **140** and Internet **150**, is presented with a Push-To-Talk™[2] option by web server **128**. When client computer **102** selects the Push-To-Talk™ option, a server incorporated with the teachings of the present invention (e.g., bridgeport **162**) automatically determines an appropriate destination PSTN extension, e.g. the PSTN extension of telephone handset **142**, as well as an appropriate one from a "community" internet/PSTN changeover servers (e.g., bridgeports **162** and **165**) to place the voice call to the PSTN extension and facilitate the voice call between the user of client computer **102** and the user of telephone handset **142**. In the context of the example embodiment, for differentiation and ease of explanation, bridgeport **162** will be referred to as a "page" bridgeport, while the selected internet/PSTN changeover server (e.g., bridgeport **165**) will be referred to as a "changeover" bridgeport. In one embodiment, the Push-To-Talk™ option is pre-associated with page bridgeport **162** by web server **128**, and the determination of the destination PSTN extension by page bridgeport **162** is made in accordance with one or more attributes of web server **128**, such as the identity of web server **128**, and optionally, one or more attributes of client computer **102**, such as the zip code of the area client computer **102** is located. In an alternate

[2] Push-To-Talk™ is a Trademark of eFusion™, Incorporated of Beaverton, Oregon.

embodiment, the Push-To-Talk™ option is not pre-associated with bridgeport **162**, but rather a page bridgeport is dynamically selected from the community of bridgeports.

In **Figure 1**, client computer **102**, web servers **120** and **128**, bridgeports **162** and **165**, and handset **142** are communicatively coupled to each other by way of PSTN **140** and internet **150** as shown. More specifically, client computer **102** is coupled to internet **150** by way of an internet service provider (ISP) **112**. Client computer **102** is coupled to ISP **112** through PSTN extension **104**, communication line **106** and PSTN **140**. In other words, for the illustrated embodiment, client computer **102** includes a modulation/demodulation (MODEM) device (not shown) coupled to PSTN extension **104**. However, a client computer may be coupled to ISP **112** through a network connection using a network interface instead, such as client computer **108** using network connection **110**. Alternatively, a client computer may also be directly coupled to internet **150** such as client computer **116** using direct connection **118**.

Web servers **120** and **128** are coupled to internet **150** through connections **122** and **130**. Although not illustrated, web servers **120** and **128** may also be coupled to PSTN **140**. Similarly, bridgeports **162** and **165** of the present invention are coupled to internet **150** through connections **164** and **167**. Bridgeports **162** and **165** are also coupled to PSTN **140** through communication lines **163** and **166** respectively. Handset **142** is coupled to PSTN **140** through PSTN extension **143** and communication line **144**.

Communication lines **106**, **115** and **144** may simply be plain old telephone service (POTS) communication lines, although other types of communication lines may be used. For example, in the case of communication line **106**, it may be an integrated service digital network (ISDN) line, whereas in the case of communication line **115**, it may be a T1 (1.533 Mbps) or an E1 (2.0488 Mbps) trunk line. In the case of communication line **144**, it may be a wireless cellular connection, a Personal Communications Services (PCS) connection, and the like.

PSTN **140** includes a number of Service Switching Points (SSP), Signal Transfer Points (STP), and Service Control Points (SCP) coupled to each other (not shown). PSTN extension **104** through communication line **106** is coupled to a "local" SSP, which in turn is coupled to a number of other "local" PSTN extensions, including e.g. PSTN extension **113** if ISP **112** is a "local" ISP served by the same SSP. In addition, the "local" SSP is also coupled to an associated STP, which in turn is coupled to other "remote" SSPs. Each of the "remote" SSPs is coupled to a number of "remote" PSTN extensions, including e.g. extension **143**, if handset **142** is a "remote" handset served by a "remote" SSP. As is well known in the art, internet **150** includes a number of networks interconnected by routers, interconnecting the various client computers, web servers and bridgeports together. [As described earlier, internet **150** may be a private intranet instead.]

Except for the incorporated teachings of the present invention (to be more fully described below), client computer **102** is intended to represent a broad category of internet telephony enabled computer systems known in the art. An example of such a computer system is a desktop computer system equipped with a high performance microprocessor, such as the Pentium® processor manufactured by Intel Corporation of Santa Clara, CA or the Alpha® processor manufactured by Digital Equipment Corporation of Manard, MA; a number of audio input and output peripherals/interface for inputting, digitizing and compressing outbound audio, and for decompressing and rendering inbound audio; a communication interface for sending and receiving various data packets (including audio data packets) in accordance with certain standard communication protocol, such as a V.42bis compliant modem or an Ethernet adapter card; a windows-based operating system including internetworking communication services providing support for Transmission Control Protocol/Internet Protocol (TCP/IP) (and other Internet Communication Suite protocols) and socket services , such as Windows™ 95 developed by Microsoft Corporation of Redmond, WA; a web communications tool such as Navigator™, developed by Netscape Communications of Mountain View, CA; and an internet telephony application, such as the above described IPhone™[3] developed by Intel Corporation.

In one embodiment, the teachings of the present invention are incorporated in client computer **102** in the form of a client application, e.g., a bridgeport driver. The client bridgeport driver may be made available to client computer **102** in a number of alternate means. For example, the client bridgeport driver may be distributed via diskettes produced by a bridgeport vendor, or downloaded from a web server of the bridgeport vendor. In other embodiments, the teachings of the present invention are incorporated in the browser and/or the operating system of client computer **102**. For ease of understanding, the remaining descriptions will be presented in the context of the client bridgeport driver embodiment.

Except for the presentation of web pages having Push-To-Talk™ options pre-associated with the bridgeports of the present invention, web servers **120** and **128** are intended to represent a broad category of web servers, including e.g. corporate presence servers and government presence servers, known in the art. Any number of high performance computer servers may be employed as web servers **120** and **128**, e.g. a computer server equipped with one or more Pentium® Pro processors from Intel Corp., running Mircrosoft's Windows® NT operating system, or a computer server equipped with one or more SPARC® processors from Sun Microsystems of Mountain View, CA, running Sun's Solaris® operating system.

Similarly, ISP **112** is intended to represent a broad category of internet service providers. An ISP may be a "small" local internet access provider, or one of a number of

[3] Note that it is not necessary for the internet telephony application to explicitly support voice calls with telephone handsets, as is the case with IPhone and many of the prior art internet telephony applications.

point of presence providers offered by a "large" ISP. It is also anticipated that ISP **112** may be incorporated with an SSP of PSTN **140**. Handset **142** is intended to represent a broad category of conventional handsets known in the art, including but not limited to desktop handsets, cordless handsets, and wireless handsets. No special features are required of handset **142** for it to be called and connected to internet telephony enabled client computer **102**, in accordance with the present invention. [As described earlier, handset **142** may also be automated/computerized telephony answering equipment.]

Before we proceed to describe bridgeports **162** and **165** in further detail, it should be noted that one skilled in the art of, for example, telecommunications, will appreciate that the communication system illustrated in **Figure 1**, is significantly more complex than that which is depicted. For example, each SSP of PSTN **140** may service thousands of PSTN extensions, and there are numerous SSPs, STPs and SCPs in a common PSTN implementation. Internet **150** includes well over several hundred thousand of networks. Together, PSTN **140** and internet **150** interconnect millions of client computers and web servers. Nonetheless, **Figure 1** does capture a number of the more relevant components of a communication system necessary to illustrate the interrelationship between client computer **102**, web server **128**, bridgeports **162** and **168**, and handset **142**, such that one skilled in the art may practice the present invention. Also, while the present invention is being described in the context of client computer **102** being engaged in data communication with web server **128**, as will be readily apparent from the description to follow, the present invention may be practiced with any "client" computer engaged in data communication with any "web" or "info" server.

Turning now to **Figure 2**, a flow chart illustrating one embodiment of the method steps of the present invention for establishing and facilitating a voice call to a PSTN extension for a networked client computer is shown. For ease of explanation, the method of **Figure 2** will be developed in the context of an example implementation, wherein a user of client computer **102** is engaged in a data communication session involving a webpage, projected by web server **128**, which incorporates a Push-To-Talk™ feature wherein the user of the webpage may "push" a displayed Push-To-Talk™ button to cause a voice connection with a local office, retail center and the like, associated with the web server, to be established, enabling the user of client computer **102** to engage in a voice call with a user of a PSTN endpoint (e.g., handset **142**) located at the "local office". [Those skilled in the art will appreciate that the terms "push" and "pushing" are metaphoric descriptions of the action taken by a user. The action is in actuality accomplished, e.g., by the user clicking a mouse button, upon moving a cursor over the displayed Push-To-Talk™ button.]

With reference to **Figure 2**, the method begins at step **202** with a user of client computer **102** "pushing" the Push-To-Talk™ button projected with the webpage. As described earlier, the Push-To-Talk™ button is pre-associated with a bridgeport, e.g. page bridgeport **162**. In one embodiment, the pre-association is accomplished via HyperText Markup Language (HTML) elements embedded in the web page, identifying the Uniform Resource Locator (URL) of page bridgeport **162**. The HTML elements further specify that a Push-To-Talk™ event notice including the URL of web server **128** is to be posted to page bridgeport **162**. Thus, in response to the user's "pushing" of the Push-To-Talk™ button, a HyperText Transmission Protocol (HTTP) connection is temporarily established between client computer **102** and page bridgeport **162**, and a message posting the Push-To-Talk™ event is sent to page bridgeport **162**.

For the illustrated embodiment, in response to the Push-To-Talk™ event notification, page bridgeport **162** identifies itself to client computer **102**, providing client computer **102** with its internet protocol (IP) address, step **204**. The HTTP connection is closed upon sending the return data to client computer **102** by page bridgeport **162**. In one embodiment, the identification and provision of page bridgeport's IP address also includes identification that the information is associated with a Push-To-Talk™ button projected by web server **128**.

More specifically, the URL of web server **128** is also returned to client computer **102**. For the illustrated embodiment, the returned data also includes a command for starting up the client bridgeport driver on client computer **102**.

Next, for the illustrated embodiment, client computer **102** identifies itself to page bridgeport **162**, providing page bridgeport **162** with its IP address and attributes. For the illustrated embodiment, the identification is performed by the launched client bridgeport driver, by way of another HTTP connection. In one embodiment, the identification is also made in the form of a call request to talk to a person, such as an agent, associated with web server **128**. In one embodiment, the attribute information includes a zip code for the area client computer **102** is located. In another embodiment, the attribute information includes a PSTN extension associated with client computer **102**. In one embodiment, the URL of web server **128** is also re-transmitted.

In response, page bridgeport **162** determines a destination PSTN extension for the requested call, step **208**. In one embodiment, the determination is based on attributes of web server **128**, e.g. the URL or the associated corporate name (if provided) of web server **128**. In another embodiment, the determination is also based on attributes of client computer **102**, e.g. the zip code or the telephone area code/prefix associated with client computer **102**. In one embodiment, page bridgeport **162** is equipped with a PSTN extension database having the necessary attributes and PSTN extension information for performing the determination. In an alternate embodiment, page bridgeport **162** is not equipped with such a PSTN extension database, but equipped with services that access external on-line services to make the determination (e.g., on-line directory services, on-line geographic location services, etc.). An example of an on-line geographic location service is MapBlast™ developed by Vicinity Corporation of Palo Alto, CA.

Upon determining the destination PSTN extension, for the illustrated embodiment, page bridgeport **162** selects an internet/PSTN changeover server. In one embodiment, the internet/PSTN changeover server is a bridgeport, such as bridgeport **165** (hereafter, changeover bridgeport), where the requested voice call would be routed off internet **150** and onto PSTN **140**, step **210**. In the context of the example embodiment, the selection of the changeover bridgeport is made from a "community" of bridgeports, to be described more fully below. In one embodiment, the "community" of bridgeports are private bridgeports deployed by the owner of web server **128** (e.g., a corporation). In another embodiment, the "community" of bridgeports are public bridgeports deployed by a service company that offers bridgeport services of the present invention, and subscribed by the corporation of web server **128**. Note that page bridgeport **162** may select itself as the changeover bridgeport, either because of the selection criteria employed dictate the result, or by virtue of a singleton community, i.e. page bridgeport **162** is the only bridgeport of the "community". For ease of understanding, the remaining descriptions will be presented in the context of bridgeport **165** being the selected changeover bridgeport.

Upon selecting changeover bridgeport **165**, for the illustrated embodiment, page bridgeport **162** registers the requested call with changeover bridgeport **165**, step **210**. In one embodiment, the registration reserves bandwidth on changeover bridgeport **165** for the requested call. In one embodiment, the registration includes provision of the source IP address of the voice call, i.e. the IP address of client computer **102**, the source type (e.g. H.323), the destination address, i.e. the destination PSTN extension of handset **142**, and the destination type (e.g. POTS).

Next, for the illustrated embodiment, page bridgeport **162** identifies changeover bridgeport **165** to client computer **102**, providing client computer **102** with the IP address of changeover bridgeport **165**, step **212**. In one embodiment, where the call request is made through a HTTP connection, step **212** also includes closing the HTTP connection. In one embodiment, the identification also includes provision of the URL of web server **128**. In one embodiment, both the identification, i.e. the IP address of changeover bridgeport **165**, and the URL of web server **128** are provided to client bridgeport application.

Next, for the illustrated embodiment, client computer **102** places a net call to changeover bridgeport **165**, step **214**. In one embodiment, the net call is a H.323 call placed by an internet telephony application. In one embodiment, step **214** also includes automatic launching of the internet telephony application to place the net call, if an internet telephony application has not been previously launched. For the illustrated embodiment, the automatic launching is performed by the client bridgeport driver.

In response, changeover bridgeport **165** places a voice call to the PSTN extension of handset **142**, and bridges the net and the voice calls, step **216**, thereby allowing a user of client computer **102** to communicate with a user of handset **142**. In bridging the two calls, changeover bridgeport **165** digitizes and compresses inbound call signals received from handset **142**, and delivers the encoded call signals to client computer **102** via the previously established H.323 connection. The compressed inbound call signals are decompressed by the communication interface of client computer **102** and rendered by the internet telephony application. Similarly, outbound call signals emanating from client computer **102** are digitized by the audio interface, compressed by the communication interface of client computer **102** and delivered to changeover bridgeport **165** via the H.323 connection, wherein they are decompressed, and upon conversion, forwarded to handset **142**. In other words, changeover bridgeport **165** converts the voice information between PSTN and IP protocols and delivers voice call information to/from handset **142** and client computer **102** until call completion.

Note that steps **204 - 206** are all performed automatically in response to step **202**, without requiring any intervention from the user of client computer **102**. In particular, it does not require the user of client computer **102** to enter the PSTN extension of handset **142**, nor the IP address of changeover bridgeport **165**. It does not even require the user of client computer **102** to know this information. All that is required of the user is the metaphorical "pushing" of the Push-To-Talk™ button projected by web server **128**.

It should also be noted that the voice connection has minimal impact on establishing any additional data connections with any number of web servers **120** and **128**. In other words, client computer **102** may continue to browse web pages offered by web servers **120** and **128**, while simultaneously supporting the voice connection with handset **142** via changeover bridgeport **165**. In addition, although there may be a number of intermediate routers in Internet **150** between changeover bridgeport **165** and client computer **102**, changeover bridgeport **165** is the only server charged with supporting both the voice connection and the H.323 connection, while the voice connection is transparent to the intermediate routers. In other words, the information exchanged between changeover bridgeport **165**, through the plurality of intermediate routers of Internet **150**, to client computer **102** will appear as normal data packets to the intermediate routers.

Returning now to step **210**, as described earlier, page bridgeport **162** selects changeover bridgeport **165** from a community of bridgeports. In one embodiment, page bridgeport **162** first solicits input on a number of call characteristics from each bridgeport member of the community. The call characteristics may include the number of intermediate servers (Is) required to connect client computer **102** with the responding bridgeport member, the toll charge (Tc) that may be incurred by placing the telephone call from the responding bridgeport, the bandwidth (B) currently available on the responding bridgeport, the number of PSTN connections (P) supported by the responding bridgeport, service premiums (S), if any, charged by the responding bridgeport, and so forth. In the context of the example implementation, one member bridgeport may respond with an indication that there are no intermediaries between itself and client computer **102**, which may result in a higher quality of service, however, given its connection point to PSTN **140**, there may be a significant toll charge incurred in placing the telephone call to handset **142** from this member bridgeport. Alternatively, another member bridgeport may respond with a low toll charge, but with a higher number of intermediaries, which may result in a lower quality voice connection as the voice data will have to route through a large number of routers.

In any event, for the illustrated embodiment, page bridgeport **162** calculates a Call Metric (CM) for each responding bridgeport that is representative of the bridgeport's ability to establish and facilitate the voice call between client computer **102** and handset **142**. Equation (1) is an example of an equation used to calculate the Call Metric for each of the responding bridgeport, wherein the bridgeport with the lowest CM is determined to be able to provide the best all around service. It should be noted that equation (1) is merely illustrative, as one skilled in the art will appreciate that suitable alternative equations may be beneficially employed to calculate alternative call metrics.

$$CM_i = W_1(Is_i) + W_2(Tc_i) + \frac{W_3}{B_i} + \frac{W_4}{P_i} + W_5(S_i) \tag{1}$$

where:	Is	number of intermediate servers
	Tc	toll charge estimate
	B	available bandwidth on responding server
	P	number of PSTN ports available on responding server
	S	premium service charge
	W	weighting factor

In one embodiment of the present invention, the number of intermediate servers (Is) is determined by each of the responding bridgeports from the community of bridgeports through the use of a "traceroute" function, common to the UNIX network operating environment. As one skilled in the art will appreciate, the execution of a "traceroute" command by a bridgeport will produce a result quantifying the number of intermediate routers between the execution bridgeport and a destination address (provided in the command line). Accordingly, in one embodiment, the execution of the command: "traceroute (IP_address)", will return a number representative of the intermediate routers required for the responding bridgeport to communicate with the source address, represented by IP_address.

In one embodiment, an estimate of the toll charge is determined locally at the responding bridgeport by accessing a toll rate table stored on the responding bridgeport. An example toll rate table is depicted in **Table 1**. As illustrated in the example toll rate table of **Table 1**, an estimate toll charge is determined by analyzing the components of the destination telephone number against a toll charge hierarchy. If none of the elements of the destination telephone number match (i.e., hit) the elements of the toll rate table, a high toll charge estimate is returned. If, however, there is a hit on the area code and the telephone number prefix, a much lower toll charge or even toll free estimate is returned. In alternate embodiments of the present invention, the responding bridgeport may query the local SSP (i.e., the SSP to which it is coupled) to ascertain a specific toll charge value. In such a case, the communication between the responding bridgeport and the local SSP is conducted via an out-of-band signaling protocol such as Signaling System 7 (SS7).

Elements of Destination Telephone No.	Relative Toll Charge Estimate
No Hit	$ $ $ $
Area Code Hit	$ $
Area Code and PSTN Prefix Hit	0

TABLE 1: TOLL CHARGE LOOKUP TABLE

In an alternate embodiment, page bridgeport **162** may also involve client computer **102** in the selection of a suitable changeover bridgeport, by presenting the solicited responses to the client computer **102**, in either an uncondensed or a condensed format. Whether client computer **102** should be involved in the selection process is a design choice, a trade off between ease of use (without requiring intervention from the user of client computer **102**) and functionality (allowing the user of client computer **102** to veto or influence the selection). Between the two distinct choices, a number of hybrid implementations may be employed without deviating from the spirit and scope of the present invention.

Although bridgeport **162** is being described as the page bridgeport and bridgeport **165** is being described as the changeover bridgeport, it should be recognized that they are all members of the community of bridgeports and, as such, may be beneficially employed as a page bridgeport at one point in time, and as a changeover bridgeport at another point in time, or, in one embodiment, they may be deployed as a page/changeover bridgeport simultaneously. As will be readily apparent from the descriptions to follow, bridgeports **162** and **165** of the present invention can be practiced with computer servers programmed to perform the above described bridgeport functions, thus it is expected that a bridgeport may be integrated with other equipment in a variety of manners, for example, with a web server, an ISP, an STP, and so forth.

Turning now to **Figures 3** and **4**, two block diagrams illustrating the hardware and software elements of an exemplary computer server **300** suitable to be employed as a bridgeport are depicted. As illustrated, exemplary computer server **300** is comprised of multiple processors **302a - 302n** and memory subsystem **308** coupled to processor bus **304** as depicted. Additionally, computer server **300** is comprised of a second bus **310**, a third bus **312** and a fourth bus **314**. In one embodiment, buses **312** and **314** are Peripheral Component Interconnect (PCI) buses, while bus **310** is an Industry Standard Architecture (ISA) bus. PCI buses **312** and **314** are bridged by bus bridge **316**, and bridged to ISA bus **310** and processor bus **304** by I/O controller **306**. Coupled to PCI bus **312** are network interface **318** and display interface **320**, which in turn is coupled to display **322**. Coupled to PCI bus **314** is computer telephony interface (CTI) **324**, PSTN interface **326** and SS7 Interface **328**. Coupled to ISA bus **310** are hard disk interface **330**, which in turn is coupled to a hard drive **332**. Additionally, coupled to ISA bus **310**. keyboard and cursor control device **334**, which in turn is coupled keyboard **336** and cursor control device **338**.

CTI interface **324** provides the necessary hardware to interface exemplary computer server **300** to telephony equipment, such as private branch exchange (PBX) equipment. PSTN interface **326** provides the necessary hardware to interface exemplary computer server **300** to a plurality of PSTN communication lines (e.g., T1, E1 or POTS), wherein the actual number of PSTN communication lines interfaced will be implementation dependent. Additionally, PSTN interface **326** provides advanced DSP-based voice, dual-tone multiple frequency (DTMF) and call progress functionality, which allows for downloadable DSP protocol and voice processing algorithms, thereby providing CODEC support locally on the interface. Examples of supported codecs include the Global System for Mobile Communications (GSM) codec and the ITU-T G.723.1 protocol codecs, the specification for which are commonly available from the GSM consortium and the International Telecommunications Union, respectively. Similarly, SS7 interface **328** provides the hardware necessary to interface exemplary computer server **300** with PSTN trunk lines (e.g., ISDN) supporting the out-of-band communication protocol (e.g., SS7)) used between PSTN network elements (i.e., SSP-SSP, SSP-STP, STP-SCP, etc.). In one embodiment, PSTN interface **326** is preferably an AG-T1™ (for U.S. implementations, while an AG-E1 may be seamlessly substituted for European implementations), while SS7 interface **328** is preferably the TX3000™, both of which, along with their accompanying software drivers, are manufactured by and commonly available from Natural MicroSystems of Natick, Massachusetts. Otherwise, all other elements, processors **302***, memory system **308** and so forth perform their conventional functions known in the art. Insofar as their constitutions are generally well known to those skilled in the art, they need not be further described.

From a software perspective, **Figure 4** illustrates the software elements of exemplary computer server **300**. In particular, exemplary computer server **300** is shown comprising an application layer consisting of a Bridgeport Management Driver **402**, Hop Off™[4] driver **404**, and other drivers **406**. Hop Off™ driver **404**, supported by Management Driver **402**, optional drivers **406**, and abstracted service layer **408** implements the method steps of **Figure 2** that are the responsibility of the community of bridgeports (i.e., bridgeports **162,** and **165**).

The Service Abstraction Layer (SAL) **408** is shown comprising SS7 services **410**, CTI Services **411**, Management Services **412**, Connection Services **414**, Streaming Services **416**, and Data Services **418**. The protocol/service layer is shown comprising Telephony Application Programming Interface (TAPI) **420**, Telephony Connection Protocol **422,** PSTN Data Interface **424,** CODEC **426**, Real Time (Streaming) Protocol **428**, and HTTP server **434**. Also shown in this "layer" are configuration management data **419** maintained by management service **412**, and codec services **426** employed by streaming services **416**. The driver layer is shown comprising SS7 driver **427**, CTI driver **429**, PSTN driver **430** and socket service **432**. Data and control information are exchanged between these elements in the fashion depicted.

Within the context of the present invention, one purpose of SAL **408** is to provide an Application Programming Interface (API) for all the available bridgeport and related services in exemplary computer server **300**. The API abstracts out the actual modules used for providing services such as connection establishment **(414)**, streaming and data exchange services (**416** and **418**). Additionally, SAL **408** provides the common operation tools such as queue management, statistics management, state management and the necessary interface

[4] Hop Off™ is a Trademark of eFusion™, Incorporated of Beaverton, Oregon.

between the plug-in services (i.e., drivers in the driver layer). SAL **408** is also responsible for loading and unloading the appropriate drivers as appropriate.

Connection service **414** includes a connection establishment and tear-down mechanism facilitating the interconnection to the PSTN **140**. Additionally, for the illustrated embodiment, connection service **414** employs connection and compatibility services which facilitate interoperation between communication equipment that support industry standards, thereby allowing a variety of communication equipment manufactured by different vendors to be benefited from the present invention. Connection services **414** include, in particular, services for supporting standard video telephony, such as ITU-T's H.323 video telephony, and standard data communication, such as ITU-T's T.120 data communication protocol. Examples of the connection establishment and tear-down mechanisms supported by connection service layer **414** include opening and starting PSTN ports, call control, DTMF collection, and tone generation, to name but a few.

Streaming service **416** is responsible for interfacing with the components that provide the real-time streaming functionality for the multimedia data. Once the connection has been established between the connection points (i.e., PSTN, H.323, etc.), streaming service **416** will take over the management and streaming of data between the two connected parties, until the connection is terminated. Codec service **426** facilitates the above described compression and transmission of inbound call signals from handset **142** as well as decompression and transmission of outbound call signals from client computer **102**.

Data service **418** is responsible for providing non real-time peer to peer (i.e., computer-computer) messaging and data exchange between exemplary computer server **300** and other Internet and perhaps PSTN based applications. Sending messages to exemplary computer server end-points (i.e., other similarly equipped bridgeports on the Internet) or other servers within the PSTN is accomplished via data service **418**.

CTI services **411** service all communications and automatic call distribution (ACD) necessary for Private Branch Exchange (PBX) based systems. SS7 services **410** service all out of band communications with STPs and/or SCPs of PSTN **140**.

PSTN driver **430** is equipped to accommodate particularized PSTN interfaces **326**, whereas CTI driver **429** is equipped to support particularized ACD and PBX equipment. Similarly, SS7 driver **427** is equipped to support particularized SS7 interface **328**.

While the method and apparatus of the present invention has been described in terms of the above illustrated embodiments, those skilled in the art will recognize that the invention is not limited to the embodiments so described. The present invention can be practiced with modification and alteration within the spirit and scope of the appended claims. Accordingly, the description is to be regarded as illustrative instead of restrictive on the present invention.

Thus, a method and apparatus for selecting an internet/PSTN changeover server for a packet based phone call has been described.

CLAIMS

We claim:

1. An apparatus comprising:

a storage medium having stored therein a plurality of programming instructions for implementing a set of communication services for establishing and facilitating a voice call to a PSTN extension for a client computer, wherein the communication services include services for soliciting inputs from one or more other apparatuses on one or more operating characteristics associated with the establishment and support of the voice call to the PSTN extension, and selecting either the apparatus itself or one of the other apparatuses to place the voice call to the PSTN extension; and

an execution unit, coupled to the storage medium, for executing the plurality of programming instructions.

2. The apparatus of claim 1, wherein the inputs on one or more operating characteristics include an estimate of the number of intermediaries required to connect the client computer to a server.

3. The apparatus of claim 1, wherein the inputs on one or more operating characteristics include an estimate for toll charges associated with placing the voice call from the selected apparatus to the PSTN extension.

4. The apparatus of claim 1, wherein the communication services calculate corresponding call metrics based on the solicited inputs for the repsonding apparatuses, and use the calculated call metrics to make the apparatus selection.

5. The apparatus of claim 1, wherein the communication services further include telephony services for cooperating with telephony equipment coupled to the apparatus.

6. The apparatus of claim 1, wherein the storage medium further having stored therein programming instructions for implementing web server functions.

7. The apparatus of claim 1, wherein the storage medium further having stored therein programming instructions for implementing internet service provider functions.

8. A method for selecting a server to route a packet based phone call, destined for a PSTN extension, off a network and onto a PSTN, the method comprising the steps of:

(a) soliciting inputs from a community of servers on one or more operating characteristics associated with routing the packet based phone call off the network and onto the PSTN at the various servers; and

(b) selecting a server from the community of servers to route the packet based phone call off the network and onto the PSTN based on the solicited inputs. .

9. The method of claim 8, wherein step (a) comprises soliciting from each server an estimate on the number of intermediaries required to connect the client computer to the server.

10. The method of claim 8, wherein step (a) comprises soliciting from each server a toll charge estimate for placing a voice call from the server to the PSTN handset.

11. The method of claim 8, wherein step (b) comprises calculating corresponding call metrics based on the solicited inputs for the repsonding servers, and use the calculated call metrics to make the selection.

12. An apparatus comprising:

a storage medium having stored therein a plurality of programming instructions for implementing a set of communication services for establishing and facilitating a voice call to a PSTN extension for a client computer, wherein the communication services include services for receiving solicitation for inputs from another apparatus on one or more operating characteristics associated with the establishment and support of the voice call to the PSTN extension, determining the one or more operating characteristics, and then responding with the determined one or more operating characteristics; and

an execution unit, coupled to the storage medium, for executing the plurality of programming instructions.

13. The apparatus of claim 12, wherein the inputs on one or more operating characteristics include an estimate of the number of intermediaries required to connect the client computer to a server.

14. The apparatus of claim 13, wherein the estimate of the number of intermediaries required to connect the client computer to a server is derived from execution of a traceroute function.

15. The apparatus of claim 12, wherein the inputs on one or more operating characteristics include an estimate for toll charges associated with placing the voice call from the selected apparatus to the PSTN extension.

16. The apparatus of claim 15, wherein the toll rate estimate is retrieved from a local toll rate table.

17. The apparatus of claim 16, wherein the toll rate table is located within the apparatus.

18. The apparatus of claim 16, wherein the toll rate table is located at a signal control point (SCP) of the PSTN.

19. The apparatus of claim 18, wherein the toll rate table located at the SCP is accessed by the apparatus with an out-of-band signalling protocol.

20. The apparatus of claim 19, wherein the out-of-band signalling protocol is a Signalling System 7 (SS7) standard protocol.

21. The apparatus of claim 12, wherein the communication services calculate corresponding call metrics based on the solicited inputs for the repsonding apparatuses, and use the calculated call metrics to make the apparatus selection.

22. The apparatus of claim 12, wherein the communication services further include telephony services for cooperating with telephony equipment coupled to the apparatus.

23. The apparatus of claim 12, wherein the storage medium further having stored therein programming instructions for implementing web server functions.

24. The apparatus of claim 12, wherein the storage medium further having stored therein programming instructions for implementing internet service provider functions.

25. A method for selecting a server to route a packet based phone call, destined for a PSTN extension, off a network and onto a PSTN, the method comprising the steps of:

(a) receiving solicitation inputs for one or more operating characteristics associated with routing the packet based phone call off the network and onto the PSTN;

(b) determining the one or more operating characteristics; and

(c) responding the determined one or more operating characteristics;. .

26. The method of claim 25, wherein the one or more operating characteristics include an estimate on the number of intermediaries required to connect the client computer to the server.

27. The method of claim 26, wherein step (b) comprises executing a trace route function.

28. The method of claim 25, wherein the one or more operating characteristics include a toll charge estimate for placing a voice call from the server to the PSTN handset.

29. The method of claim 28, wherein step (b) comprises accessing a toll rate table.

30. The method of claim 29, wherein the toll rate table is located locally.

31. The method of claim 29, wherein the toll rate table is located at a signal control point (SCP) of the PSTN.

32. The method of claim 31, wherein step (b) further comprises sending and receving a pluralitys of out-of-band signalling protocol packets.

33. The method of claim 32, wherein the out-of-band signalling protocol is a Signalling System 7 (SS7) standard protocol.

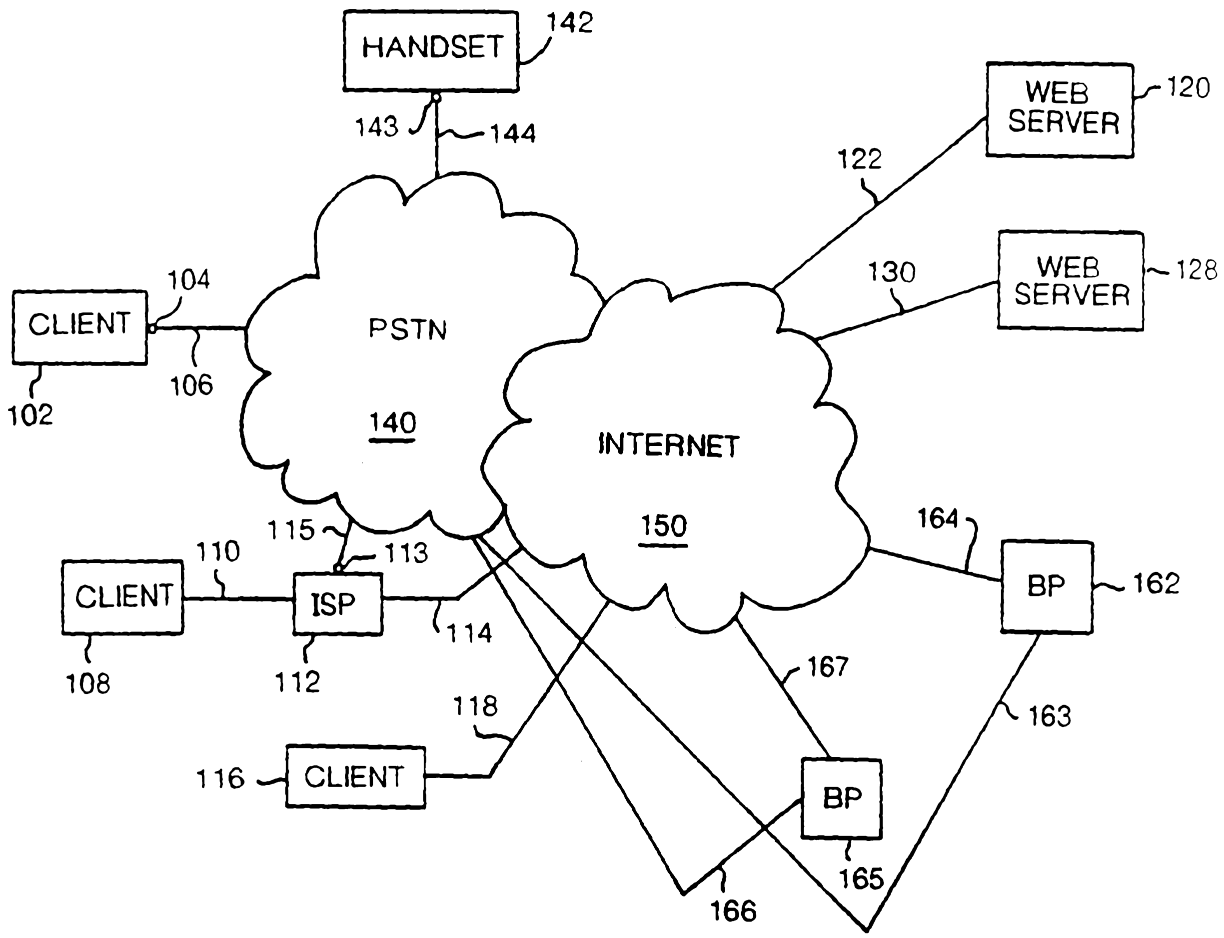
HANDSET
142
143
144
WEB SERVER
120
122
WEB SERVER
128
130
CLIENT
104
106
102
PSTN
140
INTERNET
150
115
113
110
CLIENT
ISP
114
108
112
164
BP
162
167
163
118
CLIENT
116
BP
165
166

FIG. 1

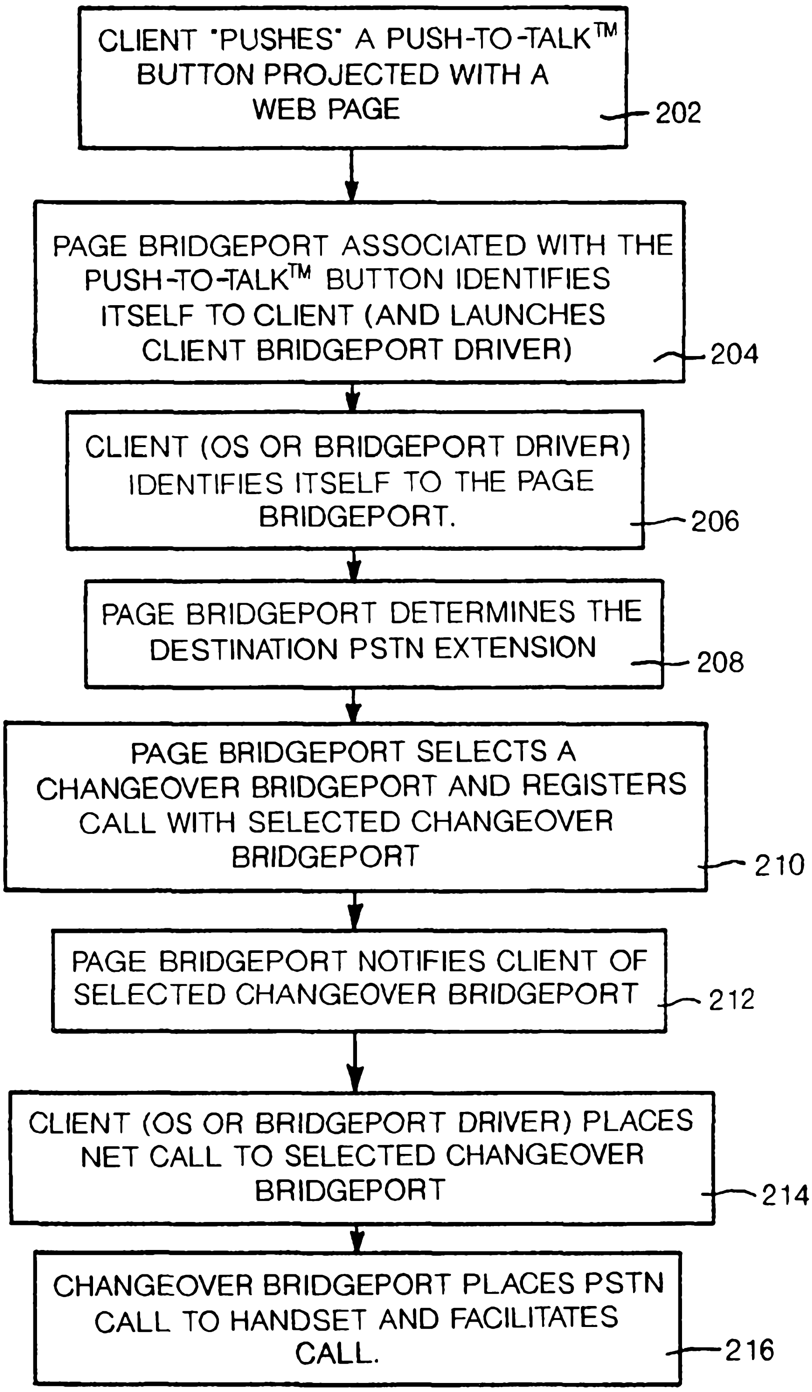
CLIENT "PUSHES" A PUSH-TO-TALK™ BUTTON PROJECTED WITH A WEB PAGE
202
PAGE BRIDGEPORT ASSOCIATED WITH THE PUSH-TO-TALK™ BUTTON IDENTIFIES ITSELF TO CLIENT (AND LAUNCHES CLIENT BRIDGEPORT DRIVER)
204
CLIENT (OS OR BRIDGEPORT DRIVER) IDENTIFIES ITSELF TO THE PAGE BRIDGEPORT.
206
PAGE BRIDGEPORT DETERMINES THE DESTINATION PSTN EXTENSION
208
PAGE BRIDGEPORT SELECTS A CHANGEOVER BRIDGEPORT AND REGISTERS CALL WITH SELECTED CHANGEOVER BRIDGEPORT
210
PAGE BRIDGEPORT NOTIFIES CLIENT OF SELECTED CHANGEOVER BRIDGEPORT
212
CLIENT (OS OR BRIDGEPORT DRIVER) PLACES NET CALL TO SELECTED CHANGEOVER BRIDGEPORT
214
CHANGEOVER BRIDGEPORT PLACES PSTN CALL TO HANDSET AND FACILITATES CALL.
216

FIG. 2

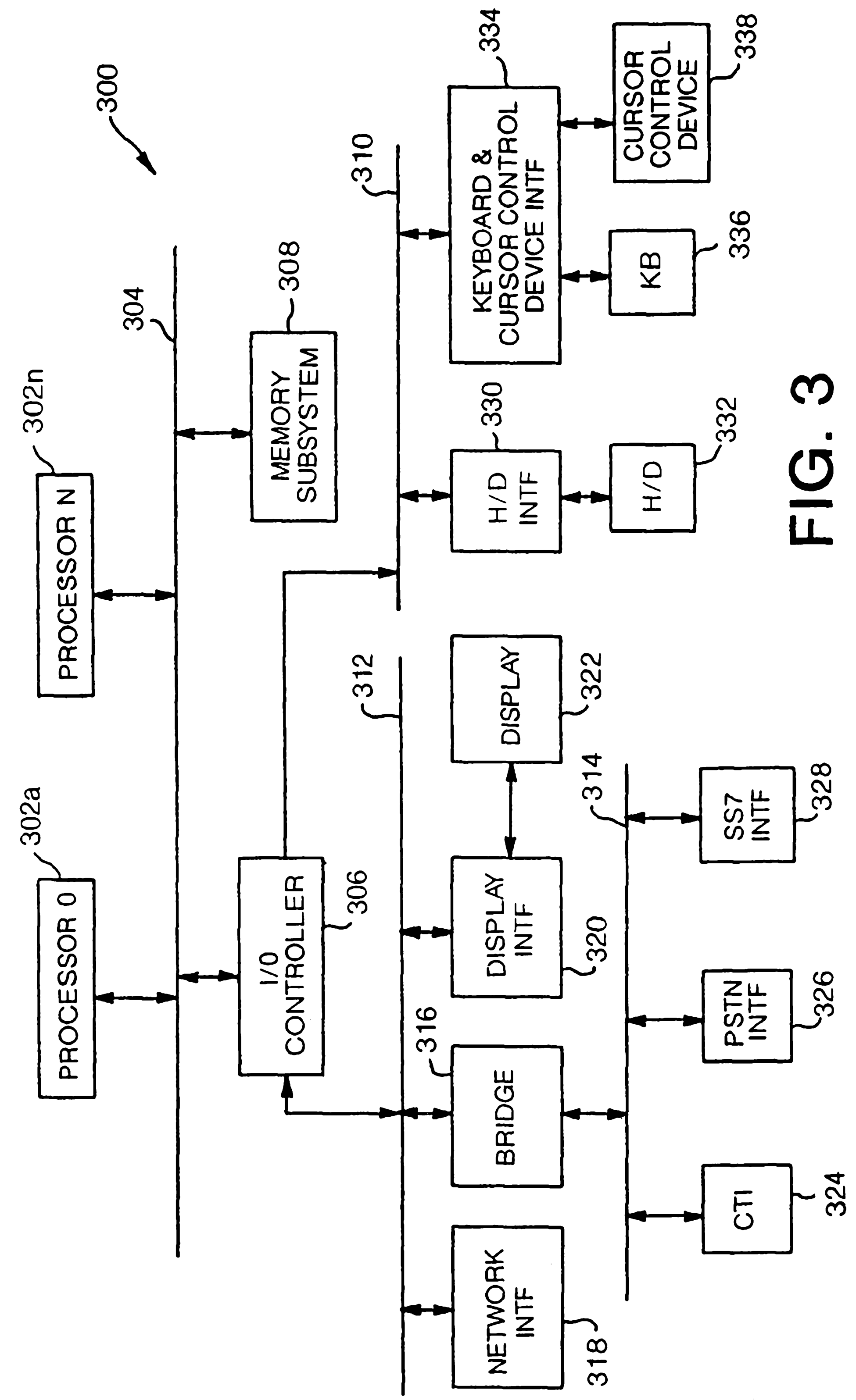
300
302a
PROCESSOR 0
302n
PROCESSOR N
304
MEMORY SUBSYSTEM
308
I/O CONTROLLER
306
310
312
KEYBOARD & CURSOR CONTROL DEVICE INTF
334
CURSOR CONTROL DEVICE
338
KB
336
H/D INTF
330
H/D
332
DISPLAY
322
DISPLAY INTF
320
314
SS7 INTF
328
PSTN INTF
326
CTI
324
BRIDGE
316
NETWORK INTF
318

FIG. 3

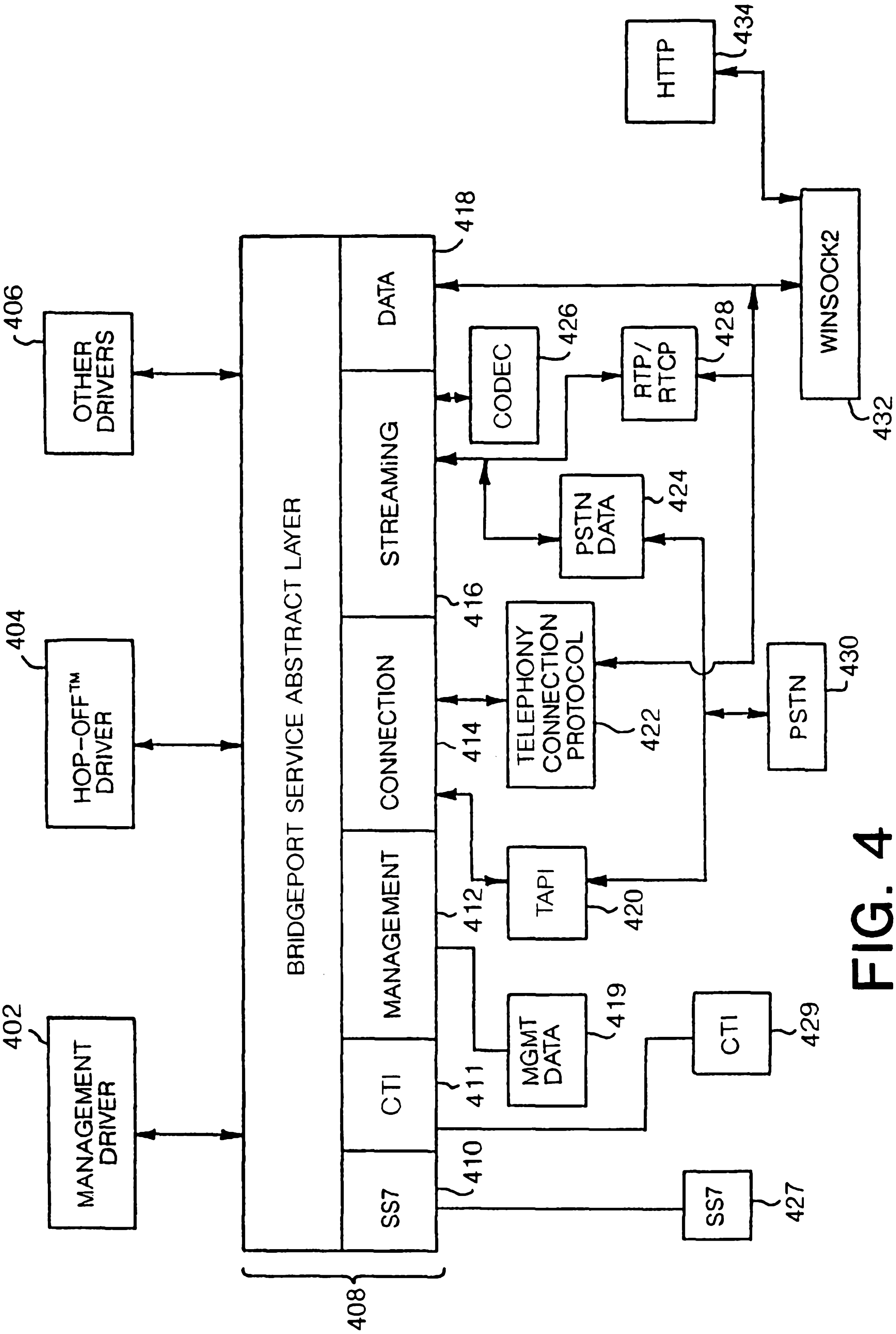
MANAGEMENT DRIVER
402
HOP-OFF™ DRIVER
404
OTHER DRIVERS
406
BRIDGEPORT SERVICE ABSTRACT LAYER
SS7
CTI
MANAGEMENT
CONNECTION
STREAMING
DATA
408
410
411
412
414
416
418
MGMT DATA
419
TAPI
420
TELEPHONY CONNECTION PROTOCOL
422
PSTN DATA
424
CODEC
426
RTP/ RTCP
428
SS7
427
CTI
429
PSTN
430
WINSOCK2
432
HTTP
434

FIG. 4