Zoho Corp. and Zoho Corp. Pvt., Ltd.
Exhibit 1035 – 001

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J1036 U.S. PTO 60/453307 03/10/03

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Atty. Dkt. No: 5757-00600

Inventor(s):
Thomas A. Dye and Tom Dundon

Title: MEDIA BASED COLLABORATION USING MIXED-MODE PSTN AND INTERNET NETWORKS

CERTIFICATE OF EXPRESS MAIL
UNDER 37 C.F.R. §1.10

"Express Mail" mailing label number: EV234071094US
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Commissioner for Patents
Box Provisional Application
Washington, DC 20231

Derrick Brown

PROVISIONAL PATENT APPLICATION TRANSMITTAL

(For provisional applications under 37 CFR § 1.53(c))

Application Elements

1. Filing Fee Amount $160.00/ $80.00
   - ☒ A Fee Authorization is enclosed.
   - ☒ The Commissioner is hereby authorized to charge filing fees or credit any overpayment to Conley, Rose, & Tayon, P.C., Deposit Account No. 50-1505/5757-00600/JCH.
2. ☒ Specification: 16 page(s)
3. ☒ Drawing(s) 5 sheet(s) (Figs. 1-5)
4. ☒ Small Entity Status is claimed

☐ This invention was made by an agency of the United States Government or under a contract with an agency of the United States Government. Name of U.S. Government agency and Government contract number: .

Inventors:

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[redacted]

Zoho Corp. and Zoho Corp. Pvt., Ltd.
Exhibit 1035 – 002

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03/10/03

307

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

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Inventor(s):
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Washington, DC 20231

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FEE AUTHORIZATION

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The Commissioner is hereby authorized to charge the following fee to Meyertons, Hood, Kivlin, Kowert & Goetzel PC Deposit Account Number 50-1505/5707-00600/JCH:

Fee: $80.00 Provisional Filing Fee
Total Amount: $80.00
Attorney Docket No.: 5707-00600

The Commissioner is also authorized to charge any extension fee or other fees which may be necessary to the same account number.

Respectfully submitted,

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PROVISIONAL PATENT
5757-00600
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March 10, 2003
Date of Mailing

Derrick Brown

MEDIA BASED COLLABORATION USING MIXED-MODE PSTN AND INTERNET NETWORKS

invented by:

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BACKGROUND OF THE INVENTION

Field of the Invention

This invention relates generally to computer system architectures and, more particularly, to audio and video telecommunications for collaboration over hybrid networks.

Description of the Related Art

Ever since they were introduced in the early Eighties, Audio/Video Conferencing (AVC) systems have enabled users to communicate with each other from remote sites using telephone lines based on dedicated and/or switched networks. Recent attempts to develop technologies and products for systems that achieve the same results over Internet Protocol (IP) have met with varied success. Generally, such systems produce low-frame-rates and low quality communications due mainly to an unpredictability factor associated with the Internet. AVC connections established over the Internet have been characterized as having high latencies and limited bandwidth, resulting in jittery video transmission, dropped audio, and quite commonly, loss of lip-sync. Therefore, most video conferencing over IP solutions have relied on dedicated switched networks such as T1/T3, ISDN and/or ATM networks. Video conferencing systems that make use of these networks are at a disadvantage due to higher costs and complexity, and a lack of flexibility arising from interoperability issues, as well as a higher cost of client equipment. High costs are typically manifested in expensive conferencing hardware and dedicated pay-per-minute communications fees. Most often, dedicated communications circuits are implemented as switched circuits with fixed bandwidth allocation.

Modern "voice over IP" (VOIP) telephony systems are generally built around the H.323 recommendations established by the international telecommunications union (ITU). The H.323 recommendations center on the transmission of audio and video information through the Internet or switched private networks. FIG. 1 illustrates a typical H.323 audio and video conferencing system implemented in accordance with prior art. The overall block diagram contains a number of major components coupling to Internet

[redacted]

435. Internet H.323 gateways/bridges 411 are interfacing with wireless phone handsets 412 and land-line phone handsets 413 via telecommunications PSTN 433. Standard Internet router 453 is coupled to an optional gatekeeper 205, a multipoint control unit 203, and a voice over IP network server 201 running the H.323 protocol, which is coupled to multiple I/O and display terminals 455. The system in FIG. 1 is an example of a prior art conferencing system using "hybrid networks" where PSTN 433 is interfacing with Internet 435. Hybrid networks are used to communicate audio on internal LAN and WAN networks as well as transferring audio to existing telephone or PSTN networks. While the H.323 standard allows for video conferencing, prior art systems generally use Private Switched Networks (PSN) to establish transport, which typically requires expensive H.323 bridges between dedicated networks and PSTNs.

Referring again to FIG. 1, network server 201 contains necessary applications to generate the H.323 network protocol. Gatekeeper 205 is responsible for management control functions such as admission control, status determination, and bandwidth management. Network server 201 functions are specified and handled through the ITU-H.225.0RAS recommendations. Multipoint control unit 203 is included to satisfy requirements of the eight-step ITU-H.323 recommendation, which include flexibility to negotiate end points and to determine compatible setups for any conference media correspondents. Multipoint control unit 203 enables communication between three or more end points. An H.323 enabled network may also contain other optional components similar to gatekeeper 205 and multipoint control unit 203. One useful function of the multipoint control unit 203 is to make a determination of whether the audio/video streams are unicast or multicast. As may be well known by those skilled in the art, this determination is typically dependent on the capability of an underlying network and topology of an associated multipoint conference. The multipoint control unit 203 generally determines the capabilities of each client terminal 455 and status of each media stream. Gateways 411 are used to the transcode H.323 network information onto PSTN 433.

FIG. 2 illustrates an H.323 protocol stack 200 and its components implemented in accordance with prior art. H 323 protocol stack 200 is coupled to network interface 300. Input and control devices 455 of Figure 1 are shown here in FIG. 2 as audio input/output

(I/O) block 452, video input/output block 451, and system control unit and data collaboration unit 459. These input devices are largely responsible for the delivery of media data to H.323 protocol stack 200. H.323 protocol stack 200 may consist of audio coder/decoder (CODEC) 214 and video CODEC 213 coupled to audio I/O 452 and video I/O 451, respectively. Audio CODEC 214 and video CODEC 213 are typically responsible for compression and decompression of utilized audio and video sources. As shown in FIG. 2, Audio CODEC 214 and video CODEC 213 are also coupled to Real-time network protocol component 215, which is also responsible for preparation of the media data for transport according to real-time protocol (RTP) recommendations.

Again, referring to FIG. 2, H.323 protocol stack 200 is coupled to system control unit 459 which connects to multiple control blocks within H.323 protocol stack 200. System control unit 459 is coupled to RTC Protocol block 217 for real-time transport of control information used in setting up and tearing down the conference. System control unit 459 is also coupled to call-signaling units 221 and 219 for implementing call signaling protocols and media stream packetization applications used in packet-based multimedia communications. System control unit 459 is further coupled to control signaling block 223 used in controlling protocols for multimedia communications. Finally, the H.323 recommendation defines a data collaboration capability outlined in T.120 data collaboration unit 225. H.323 protocol stack 200 interfaces with Transport protocol/network interface unit 300 for transport of data to Internet 435 through a modem or router 453.

As illustrated, prior art systems typically use Public Switched Telephone Networks (PSTN) to transfer audio during conferencing in collaboration with remote parties. The quality of audio reception over IP in prior art systems is widely regarded as being inadequate. Also, current AVC over IP systems lack the ability to transmit audio to client end-systems through both PSTN and IP networks. Thus, it is desirable to achieve a hybrid mix of video and audio data over PSTN and IP based AVC to achieve full duplex real-time operation for all conference participants.

References:

THE METHOD AND PROCESS FOR TRANSMISSION OF INDEPENDENTLY COMPRESSED VIDEO OBJECTS OVER INTERNET PROTOCOL, Dye et al. filed _____,2001

THE METHOD AND PROCESS FOR ASSIGNING PRIORITIZATION DURING ENCODE OF INDEPENDENTLY COMPRESSED OBJECTS, **Dye, et al.** filed on _____, 2001.

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International Telecommunications Union Recommendation H.261, Entitled, "Video Coding for Audio visual services at Px64 kbps."

International Telecommunications Union Recommendation H.263, "Video Coding for low bit-rate communications." February, 1998

Patent #6,426,948 Video conferencing fault management in hybrid networks, Issued July 30th, 2002. Who's inventor is Bowman, et al.

Patent #6,445,682 System and method for multiple mode voice and data communications using intelligent bridged TDM and packet busses and method for performing telephony and data functions using the same. Issued September 3, 2002. Who's inventor is Weitz, et al.

Patent #6,240,449 Method and apparatus for automatic call setpup in different network domains. Issued may 29th, 2001, who's inventor is nadeau, et al.

Patent # 6,385,646 Method and system for establishing voice and data communications in a Internet environment, Issued May 7th, 2002. Who's inventor is Brown, et al.

Patent #6,377,571 Virtual modem for dial-out clients in virtual private networks. Issued April 23, 2002. Who's inventor is Tai, et al.

Patent #6,327,276 Conferencing over LAN/WAN using a hybrid client/server configuration. Issued December 4th, 2001. Who's inventor is Robert, et al.

SUMMARY OF THE INVENTION

The present invention uses a decentralized model for multipoint conferencing. While a multipoint control unit may insure communication capability once the media stream is transcoded to the H.323 standard as practiced today, the present invention is capable of mixing media streams at each terminal point prior to multicasting. One embodiment of the present invention defines an application and process for enabling audio from standard and wireless telephone systems to be mixed with audio, video and collaboration data resident in IP networks, in preparation for transport preferably over virtual private networks. This embodiment teaches how audio data from PSTN may be multiplexed for transport to multiple local and remote client computers connected over IP networks. One application of the present invention allows video and audio collaboration clients to talk with remote telephony devices during audio/video conferencing. In the present invention, audio data may be transported in a hybrid network that included a PSTN and an IP network. In one embodiment of the present invention, a local client initiates a call to a remote telephone, or wireless telephony device, from a local dial-out application located preferably on the client's computer. Call set-up may be initiated as a series of control-packet data transfers to a VOIP server located at a predetermined Internet address on the world-wide-web. Control data packets may be transported to the VOIP server via a secure virtual private network (VPN). In one embodiment, a local client computer compresses the audio data prior to transporting to the VOIP server. The VOIP server may use standard ITU-T, H.323 audio telephony transport protocol on the primary network in preparation for entry into the secondary PSTN. H.323 call instantiation is generally a protocol completed by the VOIP server which may request further transport of the digitized audio stream through a gateway to the PSTN. Once proprietary data packets are handed off to the VOIP server, a standard H.323 process may be used for audio call set-up, tear-down, and transport.

After known H.323 call set-up network connections are established between Hybrid networks, following acceptance of a call by a remote telephone digitized audio

packet data may be transmitted in compressed format between the remote PSTN client (Called) and the Local client (Caller) along with all other remote audio and/or video participants located at various sites connected to the Internet. In one embodiment, compression is accomplished via a standard audio CODEC, such as one specified in the ITU-T G.729 recommendation, or with a proprietary audio CODEC. Thus, audio data transcoders at the VOIP server may be used to match expected audio decoders located at PSTN gateways.

Thus, various embodiments of the invention may provide a means for compressing audio data at a local client computer site prior to a first transport between remote clients and a VOIP server, thus minimizing transport bandwidth during a first mile of a connection to/from the Internet.

BRIEF DESCRIPTION OF THE DRAWINGS

The foregoing, as well as other objects, features, and advantages of this invention may be more completely understood by reference to the following detailed description when read together with the accompanying drawings in which:

FIG. 1 illustrates a typical H.323 audio and video conferencing system implemented in accordance with prior art;

FIG. 2 illustrates an H.323 protocol stack and its components implemented in accordance with prior art;

FIG. 3 illustrates a media based collaboration using mixed-mode PSTN and internet networks implemented in accordance with one set of embodiments of the present invention;

FIG. 4 illustrates one embodiment of audio and video data flow over hybrid networks;

FIG. 5 illustrates local client data mixing implemented in accordance with one set of embodiments of the present invention.

While the invention is susceptible to various modifications and alternative forms, specific embodiments thereof are shown by way of example in the drawings and will herein be described in detail. It should be understood, however, that the drawings and detailed description thereto are not intended to limit the invention to the particular form disclosed, but on the contrary, the intention is to cover all modifications, equivalents, and alternatives falling within the spirit and scope of the present invention as defined by the appended claims. Note, the headings are for organizational purposes only and are not meant to be used to limit or interpret the description or claims. Furthermore, note that the word "may" is used throughout this application in a permissive sense (i.e., having the potential to, being able to), not a mandatory sense (i.e., must)." The term "include", and derivations thereof, mean "including, but not limited to". The term "connected" means "directly or indirectly connected", and the term "coupled" means "directly or indirectly connected".

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

FIG. 3 illustrates a media based collaboration using mixed-mode PSTN and Internet networks implemented in accordance with one set of embodiments of the present invention. In one embodiment, the current invention allows audio video and data collaboration information to be securely transferred between a plurality of local and remote clients, preferably within a VPN. In this embodiment, a moderator (single member of a conference) may be able to dial out from a desktop computer or terminal (using a novel hybrid network structure), connecting audio from an external telephone user into the audio/video conference. Full duplex audio, video and data connections may be integrated between clients conferencing on the Internet and clients conferencing on standard telephone systems. An Internet/PSTN hybrid network may be the transport medium. FIG. 3 illustrates necessary equipment and protocols for one embodiment of a method and process to complete a dial out to PSTN.

Referring to FIG. 3, VOIP moderator 401 (call initiator or caller) may use a number of peripherals for real I/O devices at a desktop. These peripherals may include a client computing device such as a PC or other computer 459, a client terminal 455 that may include a keyboard and a mouse for I/O control, a standard desktop telephone 457, a video input device or camera 451 and an audio input device, for example a microphone, 452. Each conference call connected to the Internet may have similar peripheral hardware devices. FIG. 3 indicates a multitude of clients connected in a conference over the Internet. Internet clients may include audio video client 415, audio video client 418, and audio video client 417. In addition, two possible telephony clients may be added, using standard wired 412b or wireless 413b telephone systems. PSTN client 412a may be connected to wireless cell-phone 412b, which in turn may be connected to global dial network 450 as specified by PSTN 433. Remote telephony user client 413a may be connected to standard telephone handset 413b, which also may be connected to global dial network 450 based on PSTN 433.

In one embodiment, and referring again to FIG. 3, Internet based clients 401 415 418 and 417 are connected through routers/modems 453, preferably in a VPN configuration 461. A VPN bridge 407 may be used to connect local and remote clients together within a secure private network. A local connection from VPN bridge 407 to VOIP server 409 may be used for a transfer of conference audio from any one participant to any other participant within the conference. VOIP server 409 may be responsible for transcoding audio information from VPN 461 to PSTN gateway 411, and from PSTN gateway 411 to VPN 461, thus bridging PSTN gateway 411 and VPN 461 together.

Figure 4 illustrates one embodiment of audio and video data flow over hybrid networks. In this embodiment, moderator client 401 initiates a call using application code running on VOIP server 409. Call initiation and call transfer may be accomplished through VPN tunnel 421 connected to moderator client 401. Two connections to Moderator client 401 through VPN tunnel 421 may be established. A first connection may couple VOIP conference data for call initiation, set-up and control 405. A second connection may couple conference audio and video 403 between moderator client 401 and multiple remote clients 415 417 413 connected to Internet 435. VPN tunnel 421 may be coupled to VPN bridge 407, which may be located within Internet 435 at either a local or a remote site. In one embodiment, VPN bridge 407 is responsible for connecting and establishing a virtual private network used for secure conferencing, as it bridges all the tunnels for data transfer using a secure multicast protocol as taught by Lehmann et al in patent #_____. Thus, VPN tunnel 421, VPN tunnel 423 and VPN tunnel 425 may be on a same virtual private network. Alternate embodiments may include a plurality of tunnels connected through a single VPN bridge or multiple VPN bridges based on scalability of the entire system. An additional tunnel containing conference VOIP audio and call set-up data 405 may be connected to separate VOIP server 409. VOIP server 409 may be responsible for transcoding VOIP audio and call set-up control 405 in preparation for data transfer across H.323 network 437. H.323 network 437 may traverses Internet 435 to one of many PSTN gateways 411. PSTN gateways 411 may form a bridge between Internet 435 and PSTN 433. VOIP gateways are typically located at the local exchange carrier (LEC) in a multitude of individual points of presence throughout the world. Audio telephony calls may typically be terminated at VOIP client

413. Termination points may be located throughout the world. Thus, one embodiment of the present invention, as illustrated in FIG. 4, allows for dial-out to standard phones from a client terminal with audio and video capability over IP networks, allowing conferencing between multiple remote sites including secure VOIP audio components over a PSTN.

Figure 5 illustrates local client data mixing implemented in accordance with one set of embodiments of the present invention, with multiple network domains, software applications and operating system boundaries, and operations necessary for audio manipulation and transport. In one embodiment, video data accompanies audio data to all conference participants except PSTN client 412. For simplicity of illustration, not shown in FIG. 5 is a video conferencing path. Referring to FIG. 5, local moderator client 401 may be responsible for initiating a dial out for audio conferencing to PSTN client 412. Local moderator client 401 may also be an initiator of the conference meeting. It may be assumed that local moderator client 401 may have set up an audio video conference with remote audio video clients 418 prior to initiating the dial out for audio conferencing to PSTN client 412. In one embodiment, local moderator 401 and remote audio video clients 418 may share audio and video data in a full duplex mode among all participants except PSTN client 412. PSTN client 412 may share audio from a standard telephone or wireless telephone with all participants in the conference including local client 401 and remote audio video clients 418. Likewise, remote audio video clients 418 and local moderator client 401 may share audio with remote PSTN client 412. Thus, as indicated in FIG. 5, a VOIP call placed through a standard telephone system may bring a remote telephone user into an audio/video conference with multiple remote participants.

In one embodiment, functions and features illustrated in FIG. 5 are implemented through the use of general-purpose hardware and appropriate software for performing all required tasks. In other embodiments, various elements shown in FIG. 5 may be built as specialized hardware for improved speed performance. Again referring to FIG. 5 and starting with local moderator client 401, a process of call set-up is first performed. Local moderator client 401 may use a computer terminal coupled to a local area network, which in turn may be coupled to a wide area network, and finally, preferably coupled to VPN 461. As depicted in FIG. 5 local moderator client 401 may be equipped with proprietary software to operate as a dial-out to PSTN application. An application interface allows for

establishing dial out phone numbers to various possible clients on PSTN 433 with point-and-click ease. Once local moderator client 401 has selected a phone number for remote PSTN client 412, a point and click on a button or hot-link representing PSTN client 412 may initiate a dial-out process where audio information may be transported across hybrid networks. General tones well known in the art according to the ITT standard may be sent from local moderator client's 401 computer/terminal to VOIP server 409 located somewhere within global Internet system 435. In one embodiment, VOIP server 409 is connected to VPN 461. VOIP server 409 may use standard H.323 network protocol to establish communications as known directly to PSTN gateway 433. Once the call set-up is complete both PSTN client 412 and local moderator client 401 may have established a connection. At this time, a connection may not have been established for all the audio participants within the conference. As illustrated in FIG. 5, remote audio video clients 418 are already assumed to have been in a conference with local moderator client 401. In alternate embodiments, a different order of callers may be established. With a foregoing assumption of a conference being established prior to a call-out to PSTN 433, further definition of a VOIP audio path may be specified. Audio paths through three layers of application software 562 564 566, including an audio path through four hybrid network boundaries 510 520 435 and 515 may further be defined.

In one embodiment, transport of audio media in a hybrid mixed network system takes place over a set of distinct boundaries starting from remote client/moderator boundary 510, proceeding to local client VOIP boundary 520, Internet interface boundaries 435 and PSTN telephone network boundary 515. Remote client/moderator boundary 510 may be established as a VPN for transport of audio and video data between local moderator client 401 and remote audio/video clients 418. In an alternate embodiment, the VPN may be replaced with either switched dedicated networks or standard non-secure IP networks. Local client VOIP boundary 520 may also be a VPN coupling audio from local moderator client 401 to a local server or remote VOIP server 409. Local client VOIP boundary 520 may also be established through switched networks or Internet interfaces 435. For security purposes all connections through open Internet interfaces 435 may be protected by encryption running within a VPN. In alternate embodiments, encryption and VPNs, including public non-encrypted

information and public Internet interfaces may be excluded and private switched networks may be employed. Information above PSTN boundary 515 (as indicated in FIG. 5) may be assumed to be traveling within local client LANs, within remote client LANs, or on WANs through the Internet. PSTN boundary 515 may be considered a final boundary for network transport, where PSTN transport 433 takes place between a WAN (Internet) and gateways that transmit data to and from PSTN client 412.

Again referring to FIG. 5, using as an example a PSTN dial out call established in accordance with prior art (based on ITU H.323), an interface between conference application boundary 562 and operating system interface boundary 564, and VOIP application boundary 566 may be considered. In order to achieve full duplex operation, real time audio processing may be desired. In one embodiment, standard audio mixing takes place within operating system sound interface and API boundaries 564. Audio may be mixed by local moderator client 401 before being transported to PSTN client 412 and to remote audio video clients 418. Conference application boundary 562 may contain applications that controls audio mixing to operating system sound interface 564. Operating system sound interface 564 may also perform interfacing and mixing for VOIP application boundary 566. Information from local moderator client 401 may be input into one of many possible devices. One device, moderator audio input 550, may be connected to moderator VOIP record mixer 568. Another device, local microphone 452 from moderator client 401, may also be connected to audio mixer 534. Moderator VOIP record mixer 568 may mix together audio information obtained from audio decompressors 525 and local moderator client 401, in preparation for transport to VOIP encoder 522. In addition, audio mixer 534 may mix together audio information obtained from local moderator client 401 and VOIP decoder 524, for output to audio encoder 520. In one embodiment, audio encoder 520 combines audio information obtained from PSTN client 412 with audio information obtained from local moderator client 401, then encodes and compresses the resulting combined signal in preparation for transport across VPN network transport 461. Application software employed in audio encoder 520 may deliver audio obtained from PSTN client 412 and local moderator client 401 to remote audio video clients 418.

In order for local moderator client 401 to receive audio from PSTN client 412, moderator VOIP player mixer 569 may mix audio information previously decoded by VOIP decoder 524 with audio information obtained from remote clients 418 for presentation to local speaker 454. This presentation results in remote clients 418 hearing audio originating from PSTN client 412. In other words, PSTN client 412 may transport audio information through PSTN 433 to Internet based VOIP server 409. VOIP server 409 may transcode the audio information into a format suitable for transport onto VOIP application boundary 566. In anticipation of transport across VPN network transport 461, audio information from remote clients 418 may first need to be prepared. Therefore, audio information from remote clients 418 may be decompressed using audio decompressors 525 prior to mixing. The audio information obtained from remote clients 418 may then be mixed together with audio information obtained from local moderator client 401 in preparation for compression by VOIP encoder 522. This mixed audio information may then be placed on VOIP server 409 for transport over PSTN transport 433, terminating at PSTN client 412.

FIG. 5 outlines multiple application software boundaries for mixing audio between local and remote clients in hybrid data networks as indicated by multiple protocol boundaries 562, 564, and 566. Prior art techniques, such as those known in the ITU H.323 recommendations, have audio encoder 522 and audio decompressors 524 (from FIG. 5) located within a VOIP server running the H.323 network system protocol as indicated in FIG. 2 (audio codec 214). This generally poses a problem for low bit-rate networks, especially when video and audio are already part of transport data. Preferred embodiments of the present invention use highly compressed audio that may be compressed and decompressed at a client site. Thus, the VOIP server may be located anywhere within Internet 435 (FIG. 3, FIG. 4, and FIG. 5) without concern about limited bandwidth over the course of the first and last mile of data transmission. In addition, embodiments of the present invention allow for requirement of a single server for multiple conferences. The prior art systems, as shown in FIG. 1, place at least one or more VOIP servers behind a firewall and corporate router, for transcoding information to the H.323 network. This typically requires additional cost as a separate server is needed in each location to run the H.323 standard. Preferred embodiments of the present

invention do not require a separate server at each site, but instead may require that the desktop computer or terminal compress data prior to transport.

Thus, various embodiments of the systems and methods described above may facilitate enhanced audio/video conferencing between multiple clients, with an added capability of dialing out to a remote telephone user located within a global dial-up network.

Although the embodiments above have been described in considerable detail, other versions are possible. Numerous variations and modifications will become apparent to those skilled in the art once the above disclosure is fully appreciated. It is intended that the following claims be interpreted to embrace all such variations and modifications. Note the section headings used herein are for organizational purposes only and are not meant to limit the description provided herein or the claims attached hereto.

We claim:

1. A method comprising:

a process for allowing secure data conferencing in a hybrid network system containing multiple remote clients connected to one network type and multiple remote clients connected to a different network type.

2. The Method of claim 1;

wherein Audio and Video data are transported via a VPN to remote client terminals and,

wherein VOIP audio is transported via a VPN to a VOIP server and,

wherein the VOIP data is transcoded into a standard audio format in preparation for transport over standard PSTN telephone networks.

3. The method of claim 2, wherein said VOIP data is compressed by a local client and mixed with multiple streams of previously compressed audio for transport on the standard PSTN networks.

4. The method of claim 3, wherein audio from the standard PSTN networks is encoded and transported to both a moderator client terminal and one or more remote client terminals.

5. The method of claim 1, wherein encoder and audio mixing functions are performed at a desktop client computer instead of at a central server.

ABSTRACT OF THE DISCLOSURE

The present invention allows for a method and process that enables standard telephone users to audio conference with others who are conferencing over IP networks in a private secure environment. A dial-out from one or more conference client terminals is used, bridging audio between the Internet and PSTN networks. The process includes call set-up, initialization and tear-down, including a method to mix audio at the desktop terminal instead of at a general purpose server. The invention also allows for audio/video conferencing between multiple clients featuring audio from a standard telephone network within the conference. Standard operating system calls are employed for audio mixing between multiple clients prior to encoding and network transport. A VPN connects the IP clients together, including a VOIP server used for transcoding proprietary audio into the H.323 standard for transport into the telephony network. Thus, the method and process of the present invention brings mobile clients into a multi-way conference without the burden of changing to a common network connected between all conference participants.

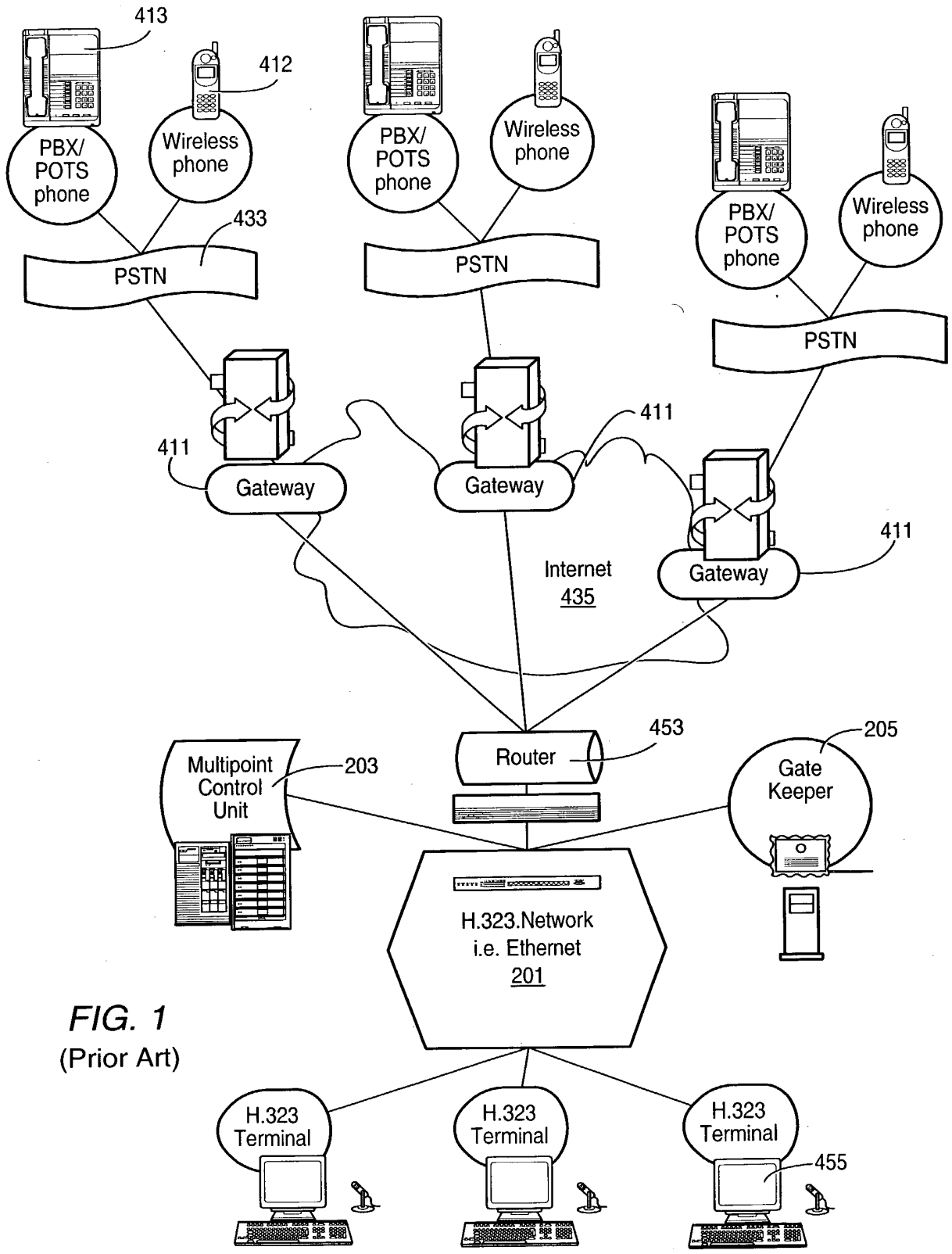


FIG. 1
(Prior Art)

6045307.031003

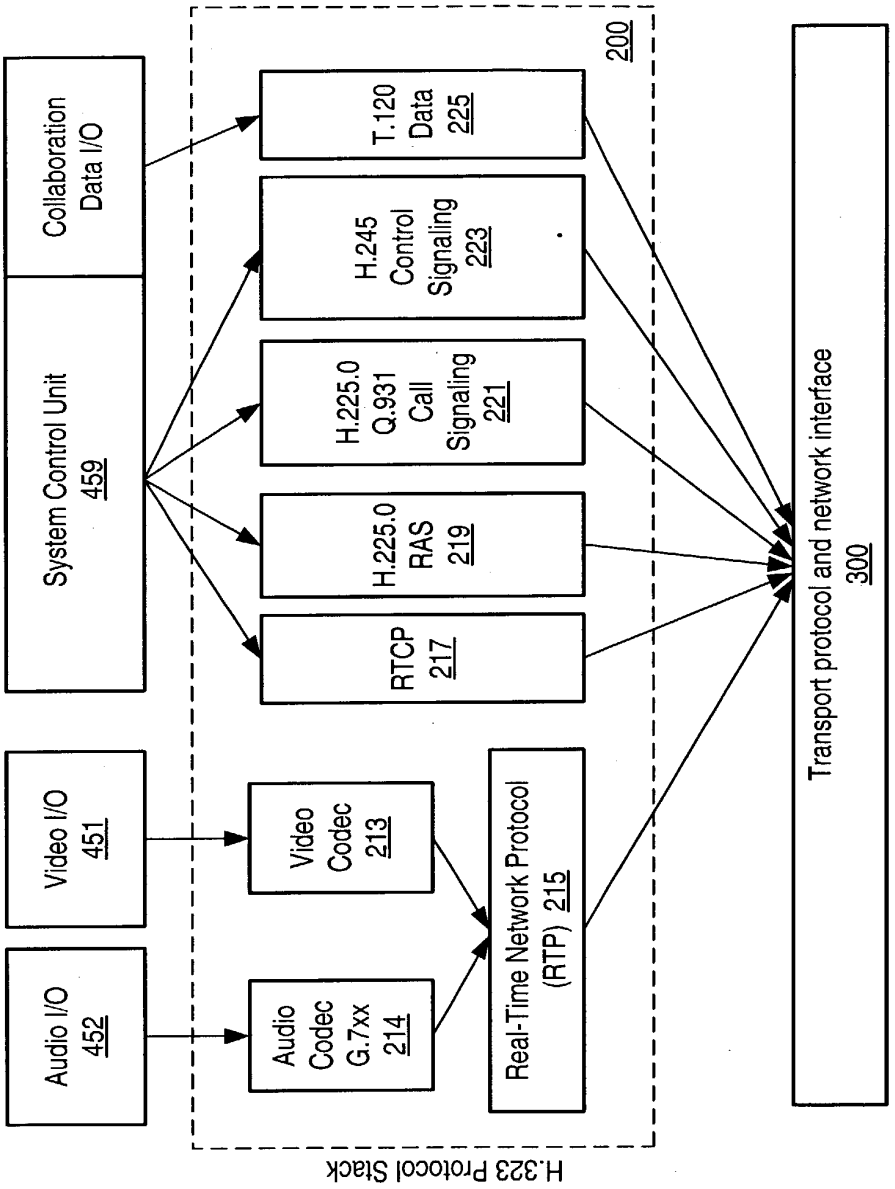


FIG. 2 (Prior Art)

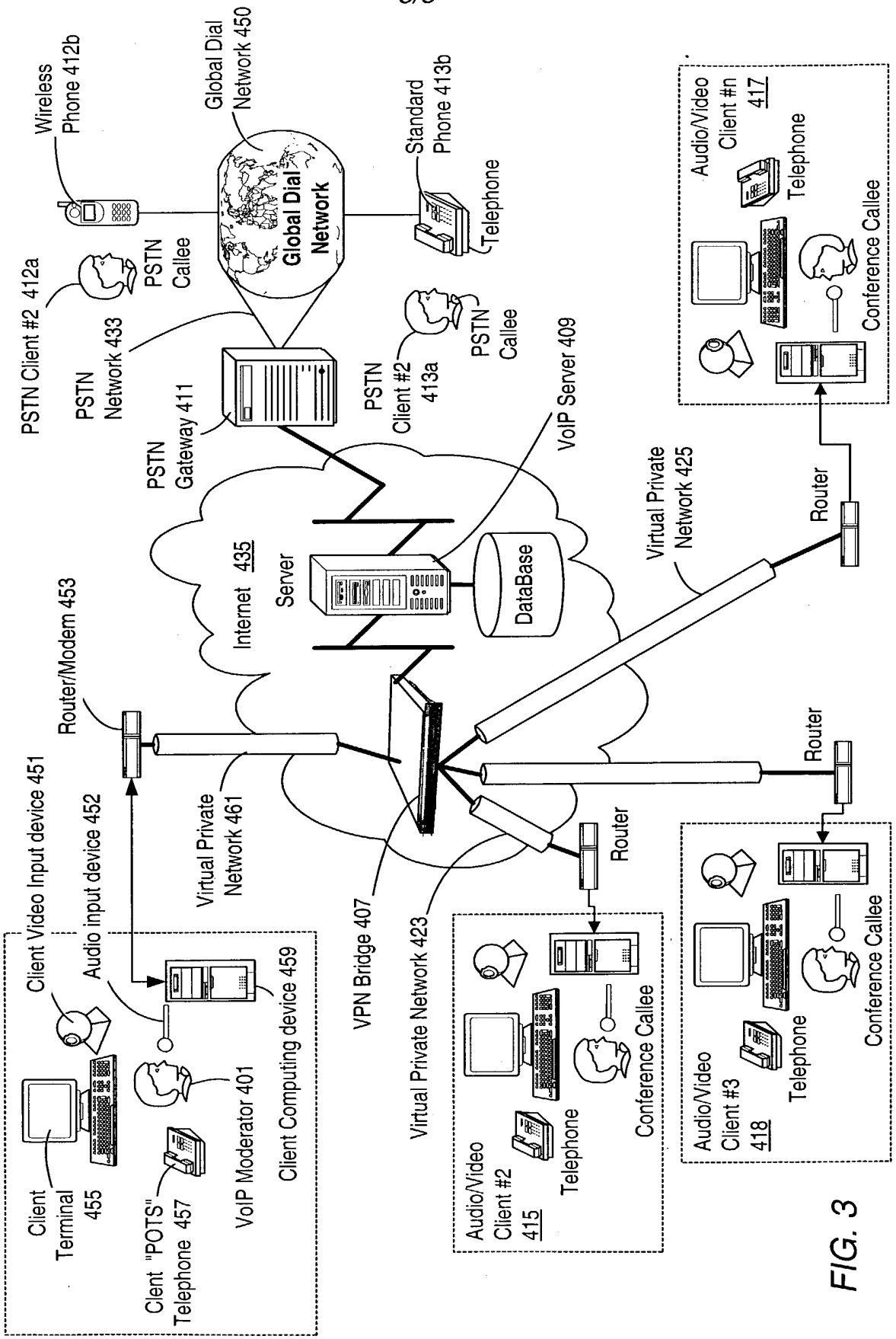

Wireless Phone 412b
Global Dial Network 450
Standard Phone 413b
Global Dial Network
PSTN Client #2 412a
PSTN Callee
Telephone
PSTN Network 433
PSTN Gateway 411
PSTN Client #2 413a
PSTN Callee
VoIP Server 409
Audio/Video Client #n 417
Telephone
Conference Callee
Internet 435
Server
DataBase
Virtual Private Network 425
Router
Router/Modem 453
Client Video Input device 451
Audio input device 452
Virtual Private Network 461
VPN Bridge 407
Virtual Private Network 423
Router
Router
Client Terminal 455
Client "POTS" Telephone 457
VoIP Moderator 401
Client Computing device 459
Audio/Video Client #2 415
Telephone
Conference Callee
Audio/Video Client #3 418
Telephone
Conference Callee

FIG. 3

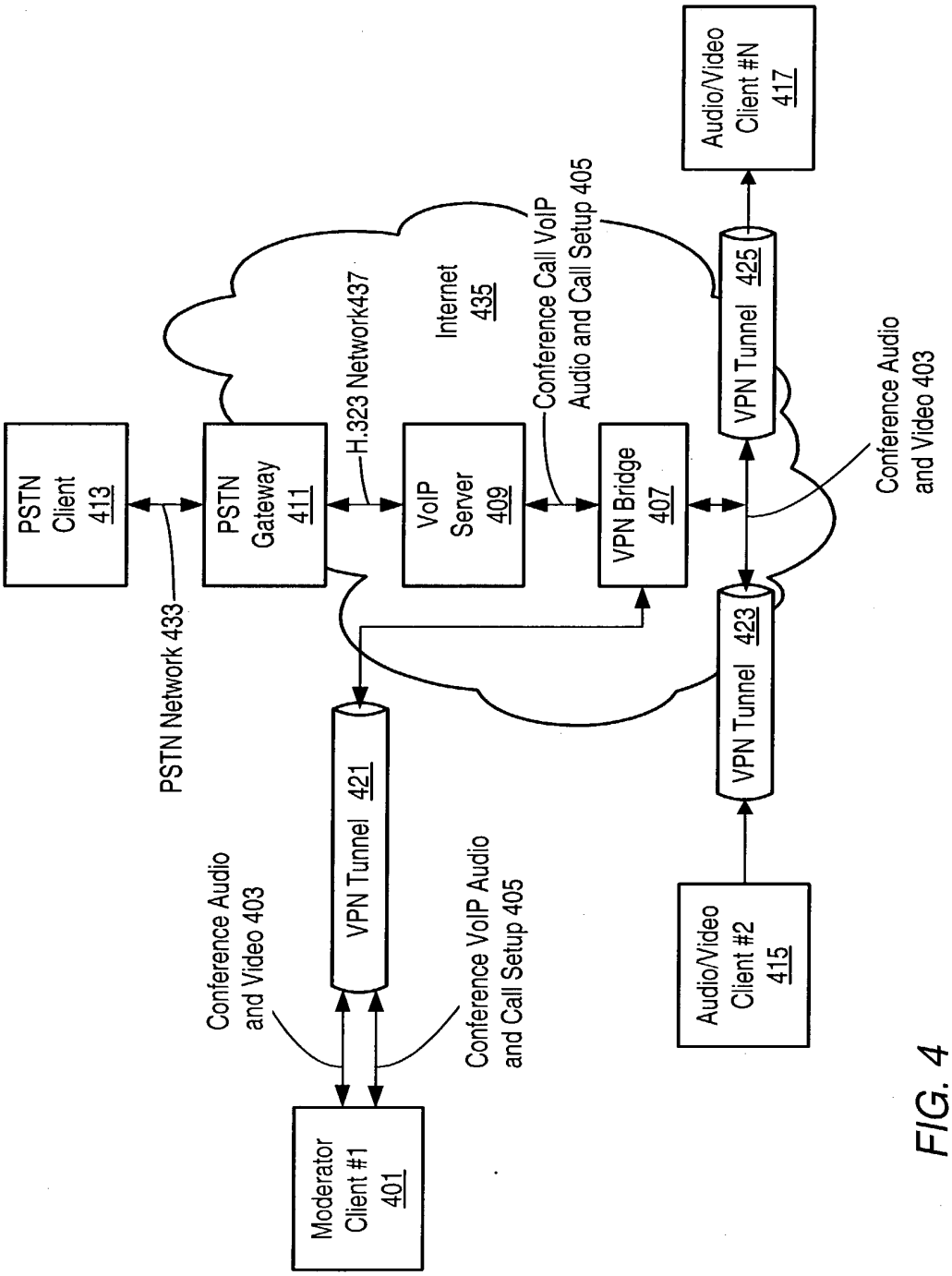
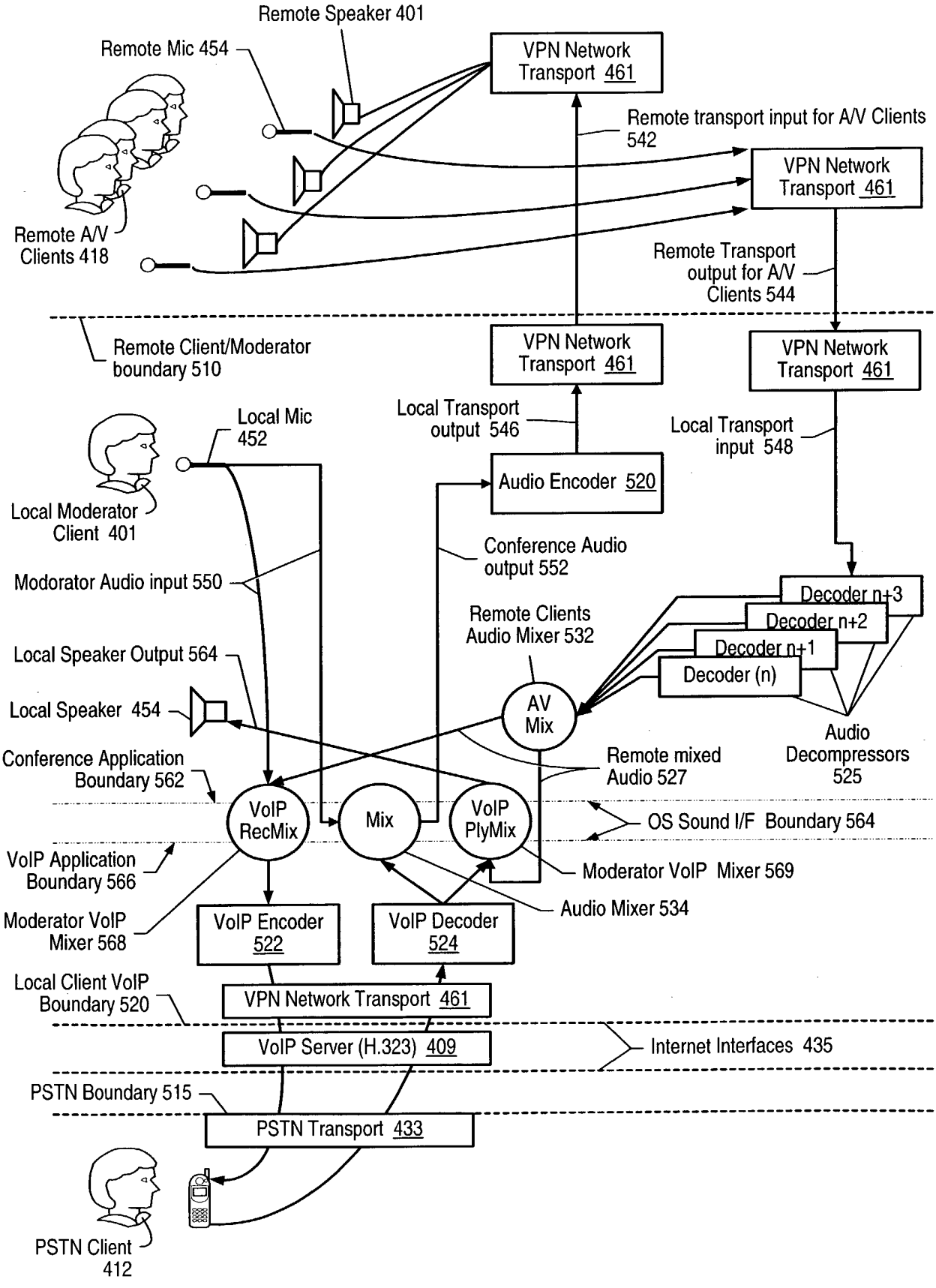

PSTN Client 413
PSTN Gateway 411
H.323 Network437
Internet 435
VoIP Server 409
Conference Call VoIP Audio and Call Setup 405
VPN Bridge 407
VPN Tunnel 425
Audio/Video Client #N 417
Conference Audio and Video 403
PSTN Network 433
VPN Tunnel 423
VPN Tunnel 421
Audio/Video Client #2 415
Conference Audio and Video 403
Conference VoIP Audio and Call Setup 405
Moderator Client #1 401

FIG. 4


Remote Speaker 401
Remote Mic 454
VPN Network Transport 461
Remote transport input for A/V Clients 542
VPN Network Transport 461
Remote A/V Clients 418
Remote Transport output for A/V Clients 544
Remote Client/Moderator boundary 510
VPN Network Transport 461
VPN Network Transport 461
Local Mic 452
Local Transport output 546
Local Transport input 548
Local Moderator Client 401
Audio Encoder 520
Conference Audio output 552
Modorator Audio input 550
Decoder n+3
Decoder n+2
Decoder n+1
Decoder (n)
Remote Clients Audio Mixer 532
Local Speaker Output 564
Local Speaker 454
AV Mix
Audio Decompressors 525
Conference Application Boundary 562
Remote mixed Audio 527
VoIP RecMix
Mix
VoIP PlyMix
OS Sound I/F Boundary 564
VoIP Application Boundary 566
Moderator VoIP Mixer 569
Moderator VoIP Mixer 568
Audio Mixer 534
VoIP Encoder 522
VoIP Decoder 524
Local Client VoIP Boundary 520
VPN Network Transport 461
VoIP Server (H.323) 409
Internet Interfaces 435
PSTN Boundary 515
PSTN Transport 433
PSTN Client 412