Zoho Corporation
and
Zoho Corporation Pvt., Ltd.

April 25, 2024

IPR2023-0377
(U.S. Patent No. 9,253,332)

IPR2023-00378 & IPR2023-00379
(U.S. Patent No. 9,094,525)

IPR 2023-00380 & IPR2023-00382
(U.S. Patent No. 9,843,612)

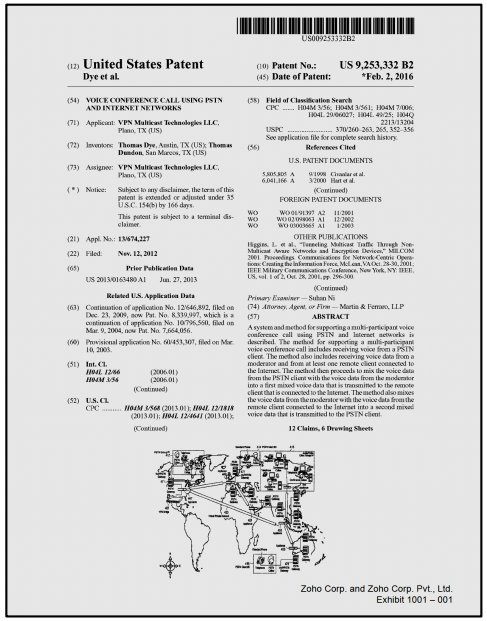
IPR 2023-00371
(U.S. Patent No. 8,339,997)

THE MEETRIX PATENTS

- Related Patents claiming priority to a March 10, 2003 provisional application

The ’332 Patent

“Voice Conference Call Using PSTN and Internet Networks”


US009253332B2

(12) **United States Patent**
Dye et al.

(10) **Patent No.: US 9,253,332 B2**
(45) **Date of Patent: *Feb. 2, 2016**

(54) **VOICE CONFERENCE CALL USING PSTN AND INTERNET NETWORKS**

(71) Applicant: **VPN Multicast Technologies LLC**, Plano, TX (US)

(72) Inventors: **Thomas Dye**, Austin, TX (US); **Thomas Dundon**, San Marcos, TX (US)

(73) Assignee: **VPN Multicast Technologies LLC**, Plano, TX (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 166 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **13/674,227**

(22) Filed: **Nov. 12, 2012**

(65) **Prior Publication Data**

US 2013/0163480 A1 Jun. 27, 2013

Related U.S. Application Data

(63) Continuation of application No. 12/646,892, filed on Dec. 23, 2009, now Pat. No. 8,339,997, which is a continuation of application No. 10/796,560, filed on Mar. 9, 2004, now Pat. No. 7,664,056.

(60) Provisional application No. 60/453,307, filed on Mar. 10, 2003.

(51) **Int. Cl.**
H04L 12/66 (2006.01)
H04M 3/56 (2006.01)
(Continued)

(52) **U.S. Cl.**
CPC ***H04M 3/568*** (2013.01); ***H04L 12/1818*** (2013.01); ***H04L 12/4641*** (2013.01);
(Continued)

(58) **Field of Classification Search**
CPC H04M 3/56; H04M 3/561; H04M 7/006; H04L 29/06027; H04L 49/25; H04Q 2213/13204
USPC 370/260–263, 265, 352–356
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,805,805 A 9/1998 Civanlar et al.
6,041,166 A 3/2000 Hart et al.
(Continued)

FOREIGN PATENT DOCUMENTS

WO WO 01/91397 A2 11/2001
WO WO 02/098063 A1 12/2002
WO WO 03/003665 A1 1/2003

OTHER PUBLICATIONS

Higgins, L. et al., “Tunneling Multicast Traffic Through Non-Multicast Aware Networks and Encryption Devices,” MILCOM 2001, Proceedings Communications for Network-Centric Operations: Creating the Information Force, McLean, VA Oct. 28-30, 2001; IEEE Military Communications Conference, New York, NY: IEEE, US, vol. 1 of 2, Oct. 28, 2001, pp. 296-300.
(Continued)

Primary Examiner — Suhan Ni
(74) *Attorney, Agent, or Firm* — Martin & Ferraro, LLP

(57) **ABSTRACT**

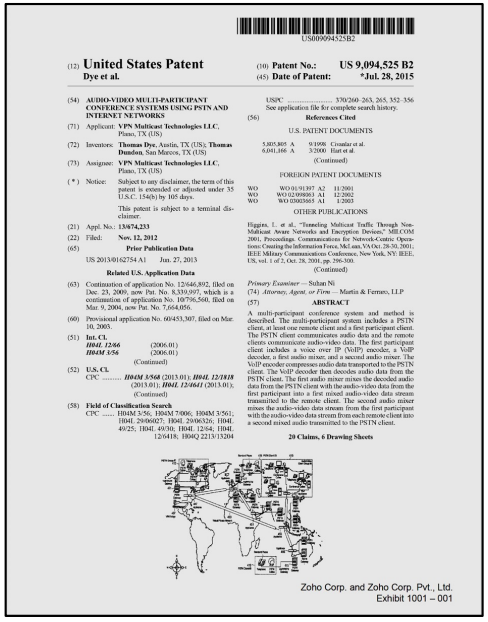
A system and method for supporting a multi-participant voice conference call using PSTN and Internet networks is described. The method for supporting a multi-participant voice conference call includes receiving voice from a PSTN client. The method also includes receiving voice data from a moderator and from at least one remote client connected to the Internet. The method then proceeds to mix the voice data from the PSTN client with the voice data from the moderator into a first mixed voice data that is transmitted to the remote client that is connected to the Internet. The method also mixes the voice data from the moderator with the voice data from the remote client connected to the Internet into a second mixed voice data that is transmitted to the PSTN client.

12 Claims, 6 Drawing Sheets

IPR2023-00377, Ex. 1001

The ’525 Patent

“Audio-Video Multi-Participant Conference Systems Using PSTN and Internet Networks”


US009094525B2

(12) **United States Patent**
Dye et al.

(10) **Patent No.: US 9,094,525 B2**
(45) **Date of Patent: *Jul. 28, 2015**

(54) **AUDIO-VIDEO MULTI-PARTICIPANT CONFERENCE SYSTEMS USING PSTN AND INTERNET NETWORKS**

(71) Applicant: **VPN Multicast Technologies LLC**, Plano, TX (US)

(72) Inventors: **Thomas Dye**, Austin, TX (US); **Thomas Dundon**, San Marcos, TX (US)

(73) Assignee: **VPN Multicast Technologies LLC**, Plano, TX (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 107 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **13/674,233**

(22) Filed: **Nov. 12, 2012**

(65) **Prior Publication Data**

US 2013/0162754 A1 Jun. 27, 2013

Related U.S. Application Data

(63) Continuation of application No. 12/646,892, filed on Dec. 23, 2009, now Pat. No. 8,339,997, which is a continuation of application No. 10/796,560, filed on Mar. 9, 2004, now Pat. No. 7,664,056.

(60) Provisional application No. 60/453,307, filed on Mar. 10, 2003.

(51) **Int. Cl.**
H04L 12/66 (2006.01)
H04M 3/56 (2006.01)
(Continued)

(52) **U.S. Cl.**
CPC ***H04M 3/568*** (2013.01); ***H04L 12/1818*** (2013.01); ***H04L 12/4641*** (2013.01);
(Continued)

(58) **Field of Classification Search**
CPC H04M 3/56; H04M 7/006; H04M 3/561; H04L 29/06027; H04L 29/06326; H04L 49/25; H04L 49/30; H04L 12/64; H04L 12/6418; H04Q 2213/13204
USPC 370/260–263, 265, 352–356
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,805,805 A 9/1998 Civanlar et al.
6,041,166 A 3/2000 Hart et al.
(Continued)

FOREIGN PATENT DOCUMENTS

WO WO 01/91397 A2 11/2001
WO WO 02/098063 A1 12/2002
WO WO 03/003665 A1 1/2003

OTHER PUBLICATIONS

Higgins, L. et al., “Tunneling Multicast Traffic Through Non-Multicast Aware Networks and Encryption Devices,” MILCOM 2001, Proceedings Communications for Network-Centric Operations: Creating the Information Force, McLean, VA Oct. 28-30, 2001; IEEE Military Communications Conference, New York, NY: IEEE, US, vol. 1 of 2, Oct. 28, 2001, pp. 296-300.
(Continued)

Primary Examiner — Suhan Ni
(74) *Attorney, Agent, or Firm* — Martin & Ferraro, LLP

(57) **ABSTRACT**

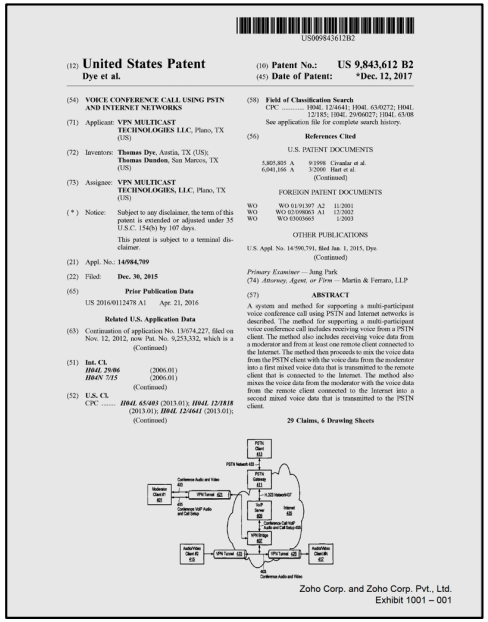
A multi-participant conference system and method is described. The multi-participant system includes a PSTN client, at least one remote client and a first participant client. The PSTN client communicates audio data and the remote clients communicate audio-video data. The first participant client includes a voice over IP (VoIP) encoder, a VoIP decoder, a first audio mixer, and a second audio mixer. The VoIP encoder compresses audio data transported to the PSTN client. The VoIP decoder then decodes audio data from the PSTN client. The first audio mixer mixes the decoded audio data from the PSTN client with the audio-video data from the first participant into a first mixed audio-video data stream transmitted to the remote client. The second audio mixer mixes the audio-video data stream from the first participant with the audio-video data stream from each remote client into a second mixed audio transmitted to the PSTN client.

20 Claims, 6 Drawing Sheets

IPR2023-00378 & IPR2023-00379, Ex. 1001

The ’612 Patent

“Voice Conference Call Using PSTN and Internet Networks”


US009843612B2

(12) **United States Patent**
Dye et al.

(10) **Patent No.: US 9,843,612 B2**
(45) **Date of Patent: *Dec. 12, 2017**

(54) **VOICE CONFERENCE CALL USING PSTN AND INTERNET NETWORKS**

(71) Applicant: **VPN MULTICAST TECHNOLOGIES LLC**, Plano, TX (US)

(72) Inventors: **Thomas Dye**, Austin, TX (US); **Thomas Dundon**, San Marcos, TX (US)

(73) Assignee: **VPN MULTICAST TECHNOLOGIES, LLC**, Plano, TX (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 107 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **14/984,709**

(22) Filed: **Dec. 30, 2015**

(65) **Prior Publication Data**

US 2016/0112478 A1 Apr. 21, 2016

Related U.S. Application Data

(63) Continuation of application No. 13/674,227, filed on Nov. 12, 2012, now Pat. No. 9,253,232, which is a
(Continued)

(51) **Int. Cl.**
H04L 29/06 (2006.01)
H04N 7/15 (2006.01)
(Continued)

(52) **U.S. Cl.**
CPC ***H04L 65/403*** (2013.01); ***H04L 12/1818*** (2013.01); ***H04L 12/4641*** (2013.01);
(Continued)

(58) **Field of Classification Search**
CPC H04L 12/4641; H04L 63/0272; H04L 12/185; H04L 29/06027; H04L 63/08
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,805,805 A 9/1998 Civanlar et al.
6,041,166 A 3/2000 Hart et al.
(Continued)

FOREIGN PATENT DOCUMENTS

WO WO 01/91397 A2 11/2001
WO WO 02/098063 A1 12/2002
WO WO 03/003665 1/2003
(Continued)

OTHER PUBLICATIONS

U.S. Appl. No. 14/590,791, filed Jan. 1, 2015, Dye.
(Continued)

Primary Examiner — Jung Park
(74) *Attorney, Agent, or Firm* — Martin & Ferraro, LLP

(57) **ABSTRACT**

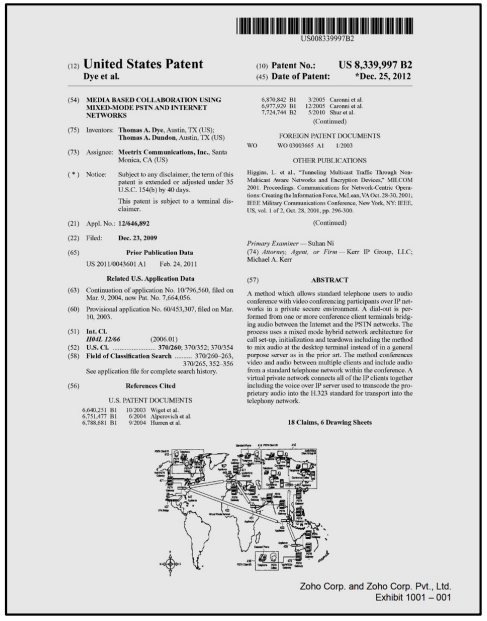
A system and method for supporting a multi-participant voice conference call using PSTN and Internet networks is described. The method for supporting a multi-participant voice conference call includes receiving voice from a PSTN client. The method also includes receiving voice data from a moderator and from at least one remote client connected to the Internet. The method then proceeds to mix the voice data from the PSTN client with the voice data from the moderator into a first mixed voice data that is transmitted to the remote client that is connected to the Internet. The method also mixes the voice data from the remote client connected to the Internet into a second mixed voice data that is transmitted to the PSTN client.

29 Claims, 6 Drawing Sheets

IPR2023-00380 & IPR2023-00382, Ex. 1001

The ’997 Patent

“Media Based Collaboration Using Mixed-Mode PSTN and Internet Networks”


US008339997B2

(12) **United States Patent**
Dye et al.

(10) **Patent No.: US 8,339,997 B2**
(45) **Date of Patent: *Dec. 25, 2012**

(54) **MEDIA BASED COLLABORATION USING MIXED-MODE PSTN AND INTERNET NETWORKS**

(75) Inventors: **Thomas A. Dye**, Austin, TX (US); **Thomas A. Dundon**, Austin, TX (US)

(73) Assignee: **Meetrix Communications, Inc.**, Santa Monica, CA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 40 days.

This patent is subject to a terminal disclaimer.

(21) Appl. No.: **12/646,892**

(22) Filed: **Dec. 23, 2009**

(65) **Prior Publication Data**

US 2011/0043601 A1 Feb. 24, 2011

Related U.S. Application Data

(63) Continuation of application No. 10/796,560, filed on Mar. 9, 2004, now Pat. No. 7,664,056.

(60) Provisional application No. 60/453,307, filed on Mar. 10, 2003.

(51) **Int. Cl.**
H04L 12/66 (2006.01)

(52) **U.S. Cl.** **370/260**; 370/352; 370/354

(58) **Field of Classification Search** 370/260–263, 370/265, 352–356
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

6,640,251 B1 10/2003 Wiget et al.
6,751,671 B1 6/2004 Algarswami et al.
6,788,681 B1 9/2004 Hansen et al.
6,870,842 B1 3/2005 Caronni et al.
6,977,929 B1 12/2005 Caronni et al.
7,724,784 B2 5/2010 Shao et al.
(Continued)

FOREIGN PATENT DOCUMENTS

WO WO 03/003665 A1 1/2003

OTHER PUBLICATIONS

Higgins, L. et al., “Tunneling Multicast Traffic Through Non-Multicast Aware Networks and Encryption Devices,” MILCOM 2001, Proceedings Communications for Network-Centric Operations: Creating the Information Force, McLean, VA Oct. 28-30, 2001; IEEE Military Communications Conference, New York, NY: IEEE, US, vol. 1 of 2, Oct. 28, 2001, pp. 296-300.
(Continued)

Primary Examiner — Suhan Ni
(74) *Attorney, Agent, or Firm* — Kerr IP Group, LLC; Michael A. Kerr

(57) **ABSTRACT**

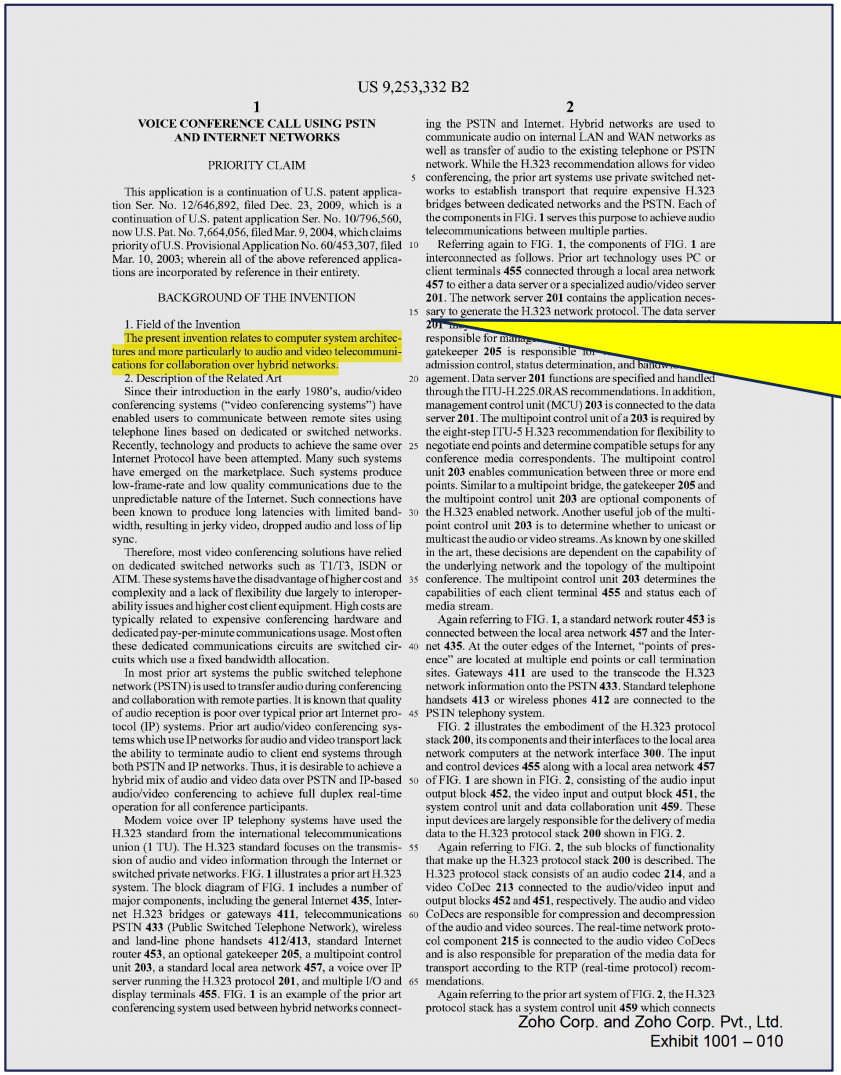
A method which allows standard telephone users to audio conference with video conferencing participants over IP networks in a private secure environment. A dial-out is performed from one or more conference client terminals bridging audio between the Internet and the PSTN networks. The process uses a mixed mode hybrid network architecture for call set-up, initialization and teardown including the method to mix audio at the desktop terminal instead of in a general purpose server as is the prior art. The method conferences video and audio between multiple clients and include audio from a standard telephone network within the conference. A virtual private network connects all of the IP clients together including the voice over IP server used to transcode the proprietary audio into the ULS3 standard for transport into the telephony network.

18 Claims, 6 Drawing Sheets

IPR2023-00371, Ex. 1001

THE MEETRIX PATENTS: CONFERENCING OVER HYBRID NETWORKS

The '332 Patent



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VOICE CONFERENCE CALL USING PSTN AND INTERNET NETWORKS

PRIORITY CLAIM

This application is a continuation of U.S. patent application Ser. No. 12/646,892, filed Dec. 23, 2009, which is a continuation of U.S. patent application Ser. No. 10/796,560, now U.S. Pat. No. 7,664,056, filed Mar. 9, 2004, which claims priority of U.S. Provisional Application No. 60/453,307, filed Mar. 10, 2003; wherein all of the above referenced applications are incorporated by reference in their entirety.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to computer system architectures and more particularly to audio and video telecommunications for collaboration over hybrid networks.

2. Description of the Related Art

Since their introduction in the early 1980's, audio/video conferencing systems ("video conferencing systems") have enabled users to communicate between remote sites using telephone lines based on dedicated or switched networks. Recently, technology and products to achieve the same over Internet Protocol have been attempted. Many such systems have emerged on the marketplace. Such systems produce low-frame-rate and low quality communications due to the unpredictable nature of the Internet. Such connections have been known to produce long latencies with limited bandwidth, resulting in jerky video, dropped audio and loss of lip sync.

Therefore, most video conferencing solutions have relied on dedicated switched networks such as T1/T3, ISDN or ATM. These systems have the disadvantage of higher cost and complexity and a lack of flexibility due largely to interoperability issues and higher cost client equipment. High costs are typically related to expensive conferencing hardware and dedicated pay-per-minute communications usage. Most often these dedicated communications circuits are switched circuits which use a fixed bandwidth allocation.

In most prior art systems the public switched telephone network (PSTN) is used to transfer audio during conferencing and collaboration with remote parties. It is known that quality of audio reception is poor over typical prior art Internet protocol (IP) systems. Prior art audio/video conferencing systems which use IP networks for audio and video transport lack the ability to terminate audio to client end systems through both PSTN and IP networks. Thus, it is desirable to achieve a hybrid mix of audio and video data over PSTN and IP-based audio/video conferencing to achieve full duplex real-time operation for all conference participants.

Modem voice over IP telephony systems have used the H.323 standard from the international telecommunications union (1 TU). The H.323 standard focuses on the transmission of audio and video information through the Internet or switched private networks. FIG. **1** illustrates a prior art H.323 system. The block diagram of FIG. **1** includes a number of major components, including the general Internet **435**, Internet H.323 bridges or gateways **411**, telecommunications PSTN **433** (Public Switched Telephone Network), wireless and land-line phone handsets **412**/**413**, standard Internet router **453**, an optional gatekeeper **205**, a multipoint control unit **203**, a standard local area network **457**, a voice over IP server running the H.323 protocol **201**, and multiple I/O and display terminals **455**. FIG. **1** is an example of the prior art conferencing system used between hybrid networks connecting the PSTN and Internet. Hybrid networks are used to communicate audio on internal LAN and WAN networks as well as transfer of audio to the existing telephone or PSTN network. While the H.323 recommendation allows for video conferencing, the prior art systems use private switched networks to establish transport that require expensive H.323 bridges between dedicated networks and the PSTN. Each of the components in FIG. **1** serves this purpose to achieve audio telecommunications between multiple parties.

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Referring again to FIG. **1**, the components of FIG. **1** are interconnected as follows. Prior art technology uses PC or client terminals **455** connected through a local area network **457** to either a data server or a specialized audio/video server **201**. The network server **201** contains the application necessary to generate the H.323 network protocol. The data server **201** [illegible] responsible for manage[illegible] gatekeeper **205** is responsible for [illegible] admission control, status determination, and bandwidth management. Data server **201** functions are specified and handled through the ITU-H.225.0RAS recommendations. In addition, management control unit (MCU) **203** is connected to the data server **201**. The multipoint control unit of a **203** is required by the eight-step ITU-5 H.323 recommendation for flexibility to negotiate end points and determine compatible setups for any conference media correspondents. The multipoint control unit **203** enables communication between three or more end points. Similar to a multipoint bridge, the gatekeeper **205** and the multipoint control unit **203** are optional components of the H.323 enabled network. Another useful job of the multipoint control unit **203** is to determine whether to unicast or multicast the audio or video streams. As known by one skilled in the art, these decisions are dependent on the capability of the underlying network and the topology of the multipoint conference. The multipoint control unit **203** determines the capabilities of each client terminal **455** and status each of media stream.

Again referring to FIG. **1**, a standard network router **453** is connected between the local area network **457** and the Internet **435**. At the outer edges of the Internet, "points of presence" are located at multiple end points or call termination sites. Gateways **411** are used to the transcode the H.323 network information onto the PSTN **433**. Standard telephone handsets **413** or wireless phones **412** are connected to the PSTN telephony system.

FIG. **2** illustrates the embodiment of the H.323 protocol stack **200**, its components and their interfaces to the local area network computers at the network interface **300**. The input and control devices **455** along with a local area network **457** of FIG. **1** are shown in FIG. **2**, consisting of the audio input output block **452**, the video input and output block **451**, the system control unit and data collaboration unit **459**. These input devices are largely responsible for the delivery of media data to the H.323 protocol stack **200** shown in FIG. **2**.

Again referring to FIG. **2**, the sub blocks of functionality that make up the H.323 protocol stack **200** is described. The H.323 protocol stack consists of an audio codec **214**, and a video CoDec **213** connected to the audio/video input and output blocks **452** and **451**, respectively. The audio and video CoDecs are responsible for compression and decompression of the audio and video sources. The real-time network protocol component **215** is connected to the audio video CoDecs and is also responsible for preparation of the media data for transport according to the RTP (real-time protocol) recommendations.

Again referring to the prior art system of FIG. **2**, the H.323 protocol stack has a system control unit **459** which connects

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Exhibit 1001 – 010

The present invention relates to computer system architectures and more particularly to **audio and video telecommunications for collaboration over hybrid networks**.

IPR2023-00377, Ex. 1001 at 1:15-18.

THE MEETRIX PATENTS: CONFERENCING OVER HYBRID NETWORKS

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to multiple control blocks within the H.323 protocol stack **200**. The system control unit connects to the RTC Protocol block **217** for real time transport of the control information used to set-up and tear down the conference. The system control unit **459** also connects to the call-signaling units **221** and **219** for call signaling protocols and media stream packetization application used for packet-based multimedia communications. The system control unit **459** also connects to the control signaling block **223** used for control of protocols for multimedia communications. Lastly, the H.323 recommendation defines a data collaboration capability as known and outlined in the T.120 data collaboration unit **225**.

All of the defined blocks make up the H.323 protocol network interface to the Transport protocol and network interface unit **300** for transport of data through the modem or router **453** to the Internet **435**.

SUMMARY OF THE INVENTION

A system and method for supporting a multi-participant voice conference call using PSTN and Internet networks is described. The method for supporting a multi-participant voice conference call includes receiving voice from a public switched telephone network (PSTN) client. The method also includes receiving voice data from a moderator and from at least one remote client connected to the Internet. The method then proceeds to mix the voice data from the PSTN client with the voice data from the moderator into a first mixed voice data that is transmitted to the remote client that is connected to the Internet. The method also mixes the voice data from the moderator with the voice data from the remote client connected to the Internet into a second mixed voice data that is transmitted to the PSTN client.

A system for supporting a multi-participant voice conference call is also described. The system includes an audio mixer, a first transport output, a VOIP mixer and a second transport output. The audio mixer mixes voice data from a Public Switched Telephone Network (PSTN) client with voice data from a moderator into a first mixed voice data. The first transport output transmits the first mixed voice data to at least one remote client. The VOIP mixer mixes voice data from the moderator with voice data from the remote client connected to the Internet into a second mixed voice data. The second transport output transmits the second mixed voice data to the PSTN client.

Another system for supporting a multi-participant voice conference call that includes a PSTN client, a first participant client, at least one remote client, a first audio mixer and a second audio mixer is described. The PSTN client, the first participant, and each remote client device are each configured to receive audio. The first audio mixer mixes the audio from the PSTN client with the audio from the first participant into a first mixed audio. The first mixed audio is transmitted to each one remote client connected to the Internet. The second audio mixer mixes audio from the first participant with the audio from each remote client connected to the Internet into a second mixed audio. The second mixed audio is transmitted to the PSTN client.

BRIEF DESCRIPTION OF THE DRAWINGS

A better understanding of the present invention can be obtained when the following detailed description of the preferred embodiment is considered in conjunction with the following drawings, in which:

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

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

FIG. **3** illustrates one embodiment of the present invention;

FIG. **4** illustrates an embodiment using multicast Protocol;

FIG. **5** illustrates the audio and video data flow over hybrid networks; and

FIG. **6** illustrates the local client data mixing used in the preferred embodiment.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Incorporation by Reference

The following applications and references are hereby incorporated by reference as though fully and completely set forth herein.

U.S. application Ser. No. 10/446,407 titled "Transmission Of Independently Compressed Video Objects Over Internet Protocol", Dye et al. filed May 28, 2003

U.S. application Ser. No. 10/620,684 titled "Assigning Prioritization During Encode Of Independently Compressed Objects, Dye, et al. filed on Jul. 16, 2003.

International Telecommunications Union Recommendation H.323, Titled "Packet Based Multimedia Communication System." November, 2000

International Telecommunications Union Recommendation H.261, Titled "Video Coding for Audio Visual Services at Px64 kbps."

International Telecommunications Union Recommendation H.263, Titled "Video Coding for Low Bit-Rate Communications" February, 1998

One embodiment of the present invention uses a decentralized model for multipoint conferencing. The multipoint control unit insures communication capability once the media stream is transcoded to the H.323 standard as known. However, this embodiment mixes media streams at each terminal prior to multicast.

FIG. **3** illustrates one embodiment of the invention. This embodiment allows audio video and data collaboration information to be securely transferred between a plurality of local and remote clients preferably within a virtual private network. This embodiment provides the ability for a moderator (single member of the conference) to dial out from a desktop computer or terminal (using a novel hybrid network structure) connecting an external telephone user's audio into the audio/video conference. The embodiment integrates full duplex audio, video, and data connections between clients conferencing on the Internet and clients conferencing on standard telephone systems. The Internet/PSTN hybrid network is the medium used for transport. FIG. **3** depicts the necessary equipment and protocols to complete the dial out to PSTN network method and process.

Now referring to FIG. **3**, the voice over IP moderator **401** (call initiator or caller) typically has a number of peripherals used for real input output devices at the desktop. These include a client computing devices such as a PC or other computer **459**, a client terminal **455** including a keyboard and mouse for input output control, a standard desktop telephone **457**, a video input device or camera **451** and the audio input device, microphone **452**. In one embodiment each conference call connected to the Internet will have similar peripheral hardware devices. FIG. **3** illustrates a multi-party virtual conference connected over the Internet. Internet clients include audio video client **415**, audio video client **418**, and audio video client number and **417**. In addition, FIG. **3** shows two possible telephony clients using standard wired **413** or wire-

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Exhibit 1001 – 011

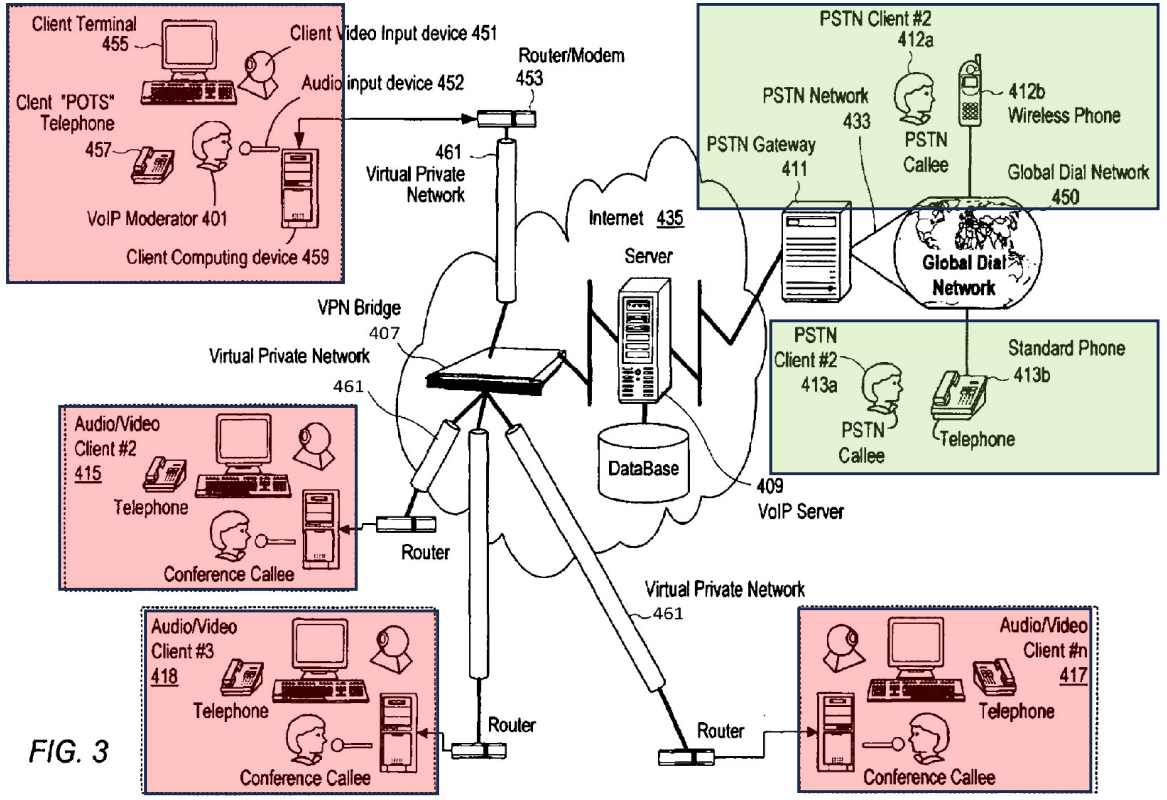


FIG. 3

FIG. 3 illustrates one embodiment of the invention. This embodiment allows audio video and data collaboration information to be securely transferred between a plurality of local and remote clients . . . The embodiment integrates full duplex audio, video, and data connections between clients conferencing on the Internet and clients conferencing on standard telephone systems. The Internet/PSTN hybrid network is the medium used for transport.

IPR2023-00377, Ex. 1001 at 4:39-53; FIG. 3 (highlights added).

STATE OF THE ART: CONFERENCING OVER HYBRID NETWORKS

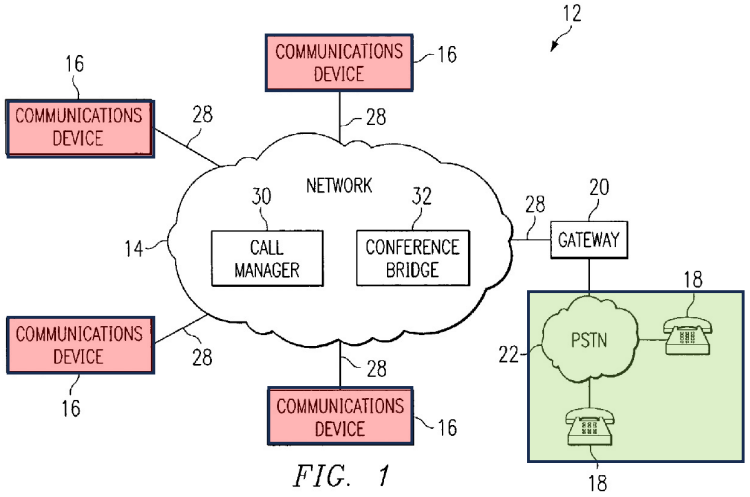


FIG. 1

Ex. 1006 at FIG. 1.

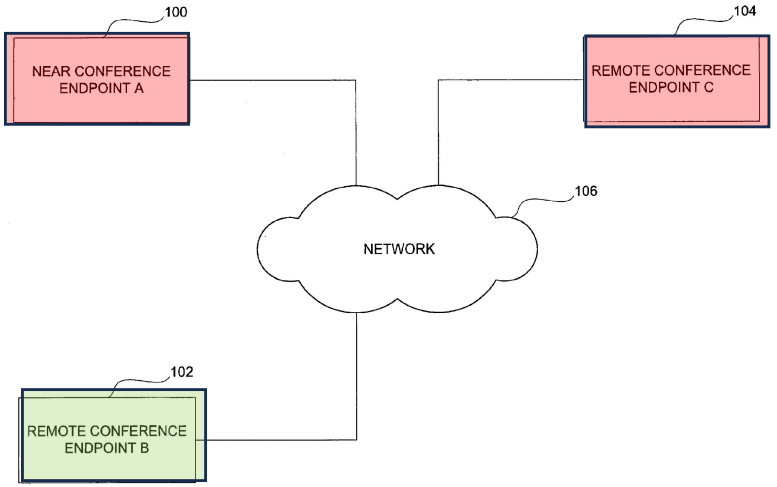


FIG. 1

Alternatively, network **106** may comprise a packet switched network, such as the Internet. Although a single network **106** is shown, the invention contemplates the use of two or more networks (for example, the PSTN and the Internet) to connect conference endpoints utilizing different communication protocols.

Ex. 1008 at FIG. 1, 3:33-38.

STATE OF THE ART: VPN

6. Virtual Private Network

91. A Virtual Private Network (VPN) is "defined loosely as a network in which customer connectivity amongst multiple sites is deployed on a shared infrastructure with the same access or security policies as a private network." *See* VPN Textbook (Ex. 1010) at 115; *see also* Yee Decl. (Ex. 1011) ¶¶ 16, 23, 95-111. VPNs were known to be used to securely exchange data between different locations over the Internet. VPN Textbook at 118-119; *see also* U.S. Patent No. 6,980,526 (the "'526 patent") (Ex. 1020) at 2:65-3:3, 5:52-55. It was also known that VPNs may be implemented over the public Internet using IPsec tunnels between sites.

IPR2023-00380, -382, Ex. 1003 ¶ 91; *see also* IPR2023-00377, Ex. 1003 ¶ 96; IPR2023-00378, -379, Ex. 1003 ¶ 107 (

STATE OF THE ART: VPN TUNNEL

3. **Tunneling**

88. Tunneling refers to a networking technique whereby a packet is encapsulated in another network packet, typically so that it can traverse an underlying network that does not support the capability required by the encapsulated packet. IP in IP tunneling is described in RFC 1853 which was published in the mid-1990s. Tunneling can be performed in both directions. At the input end of the tunnel, packets are encapsulated, or wrapped, in a way where the source address of the encapsulating header is that of the encapsulating device and the destination address of the encapsulating header is the far end of the tunnel. At the far end of the tunnel, the packets are stripped of the encapsulating header, or unwrapped, and the original packet emerges for further actions such as routing to its destination using the original packet's header.

IPR2023-00377, Ex. 1003 ¶ 88; *see also* IPR2023-00380, -382, Ex. 1003 ¶ 83; IPR2023-00378, -379, Ex. 1003 ¶ 99

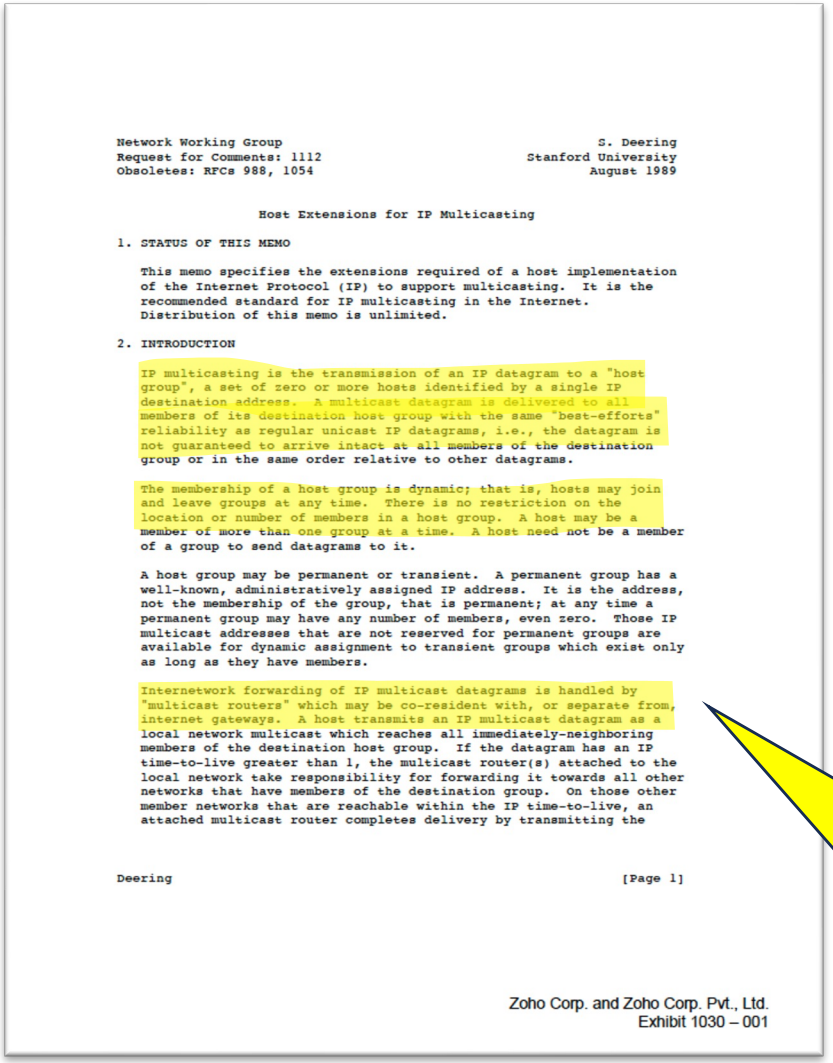
STATE OF THE ART: COMPRESSION

8. **Audio and Video Compression**

102. Audio and video compression were well known concepts, where "codecs" specify algorithms for the coding (the "co" of codec) and decoding (the "dec" of codec) of audio and video data. *See, e.g.*, Cisco Packetized Voice & Data Integration, Robert Caputo (Ex. 1021) at 12-13; U.S. Patent No. 6,850,496 (Ex. 1024) at 3:56-67. Various codecs provide different levels of compression and different levels of perceived quality. G.711 is an early standard for an audio codec, and was available in 1972. G.711 is used in H.323. G.711 specifies that audio be sampled at an 8 kHz rate with 8 bit samples, and thus the bitrate of the digitized encoded audio is 64 kilobits/second. As discussed above, the architecture of the digital PSTN is based around this fundamental base rate of digitized speech. Other voice codecs were known to further reduce the bitrate, and lower bitrate codecs were frequently used in networks to conserve bandwidth.

IPR2023-00377, Ex. 1003 ¶ 102; *see also* IPR2023-00380, -382, Ex. 1003 ¶ 97; IPR2023-00378, -379, Ex. 1003 ¶ 113.

STATE OF THE ART: IP MULTICASTING



Network Working Group
Request for Comments: 1112
Obsoletes: RFCs 988, 1054

S. Deering
Stanford University
August 1989

Host Extensions for IP Multicasting

1. STATUS OF THIS MEMO

This memo specifies the extensions required of a host implementation of the Internet Protocol (IP) to support multicasting. It is the recommended standard for IP multicasting in the Internet. Distribution of this memo is unlimited.

2. INTRODUCTION

IP multicasting is the transmission of an IP datagram to a "host group", a set of zero or more hosts identified by a single IP destination address. A multicast datagram is delivered to all members of its destination host group with the same "best-efforts" reliability as regular unicast IP datagrams, i.e., the datagram is not guaranteed to arrive intact at all members of the destination group or in the same order relative to other datagrams.

The membership of a host group is dynamic; that is, hosts may join and leave groups at any time. There is no restriction on the location or number of members in a host group. A host may be a member of more than one group at a time. A host need not be a member of a group to send datagrams to it.

A host group may be permanent or transient. A permanent group has a well-known, administratively assigned IP address. It is the address, not the membership of the group, that is permanent; at any time a permanent group may have any number of members, even zero. Those IP multicast addresses that are not reserved for permanent groups are available for dynamic assignment to transient groups which exist only as long as they have members.

Internetwork forwarding of IP multicast datagrams is handled by "multicast routers" which may be co-resident with, or separate from, internet gateways. A host transmits an IP multicast datagram as a local network multicast which reaches all immediately-neighboring members of the destination host group. If the datagram has an IP time-to-live greater than 1, the multicast router(s) attached to the local network take responsibility for forwarding it towards all other networks that have members of the destination group. On those other member networks that are reachable within the IP time-to-live, an attached multicast router completes delivery by transmitting the

Deering [Page 1]

Zoho Corp. and Zoho Corp. Pvt., Ltd.
Exhibit 1030 – 001

- "unicast" IP addresses identify an individual host computer
- IP address space was set aside for exclusive use as multicast address space
- a multicast address specifies a "host group," a set of zero or more hosts identified by a single IP address

IP multicasting is the transmission of an IP datagram to a "host group", a set of zero or more hosts identified by a single IP destination address. A multicast datagram is delivered to all members of its destination host group …
The membership of a host group is dynamic; that is hosts may join and leave groups at any time. …
Internetwork forwarding of IP multicast datagrams is handled by "multicast routers" which may be co-resident with, or separate from, internet gateways.

IPR2023-00371, Ex. 1030 at 1; Ex. 1003 at 33-34; Pet. at 22

Instituted Grounds for Cancellation of Claims

THE ’332 PATENT

Ground	Claims	Statutory Basis	Reference(s)
1	1-2, 4-5 and 7	§ 103	Knappe, Elliott, and VPN Textbook
2	3, 6, and 8-11	§ 103	Knappe, Elliott, VPN Textbook, and Drell
3	12	§ 103	Knappe, Elliott, VPN Textbook, Drell, and Hoke

IPR2023-00377, Pet. (Paper No. 1) at 4; Institution Decision (Paper No. 6) at 40-41.

THE ’612 PATENT

Ground	Claims	Statutory Basis	Reference(s)
1	10-18	§ 103	Drell, Knappe, Elliott, Hendricks, and Doganata

IPR2023-00380, Pet. (Paper No. 1) at 4; Institution Decision (Paper No. 6) at 32.

Ground	Claims	Statutory Basis	Reference(s)
1	19-21 and 25-27	§ 103	Knappe, Elliott, VPN Textbook, and Hendricks
2	22-24 and 28-29	§ 103	Knappe, Elliott, VPN Textbook, Hendricks, and Drell

IPR2023-00382, Pet. (Paper No. 1) at 4; Institution Decision (Paper No. 6) at 34.

THE ’525 PATENT

Ground	Claims	Statutory Basis	Reference(s)
1	1-4	§ 103	Drell, Knappe, and Voois
2	5-8	§ 103	Drell, Knappe, Voois, Elliott, and VPN Textbook
3	1-4	§ 103	Drell and Knappe
4	5-8	§ 103	Drell, Knappe, Elliott, and VPN Textbook

IPR2023-00378, Pet. (Paper No. 1) at 4; Institution Decision (Paper No. 6) at 30.

Ground	Claims	Statutory Basis	Reference(s)
1	9-20	§ 103	Drell, Knappe, and Elliott

IPR2023-00379, Pet. (Paper No. 1) at 4; Institution Decision (Paper No. 6) at 36.

THE ’997 PATENT

Ground	Claims	Statutory Basis	Reference(s)
1	1 and 3-7	§ 103	Knappe, Elliott, VPN Textbook, and de Moraes
2	2 and 8-10	§ 103	Knappe, Elliott, VPN Textbook, de Moraes, and Shepherd

IPR2023-00371, Pet. (Paper No. 1) at 4; Institution Decision (Paper No. 6) at 42-43.

KNAPPE

FIG. **1** illustrates a communication system **12** in accordance with one embodiment of the present invention. In this embodiment, the communication system **12** is a distributed system transmitting audio, video, voice, data and other suitable types of real-time and/or non real-time traffic between source and destination endpoints. Communication system **12** may be used to conduct multiple party telephone conference communication sessions. In accordance with a particular embodiment of the present invention, various

Ex. 1006 at 3:26-29

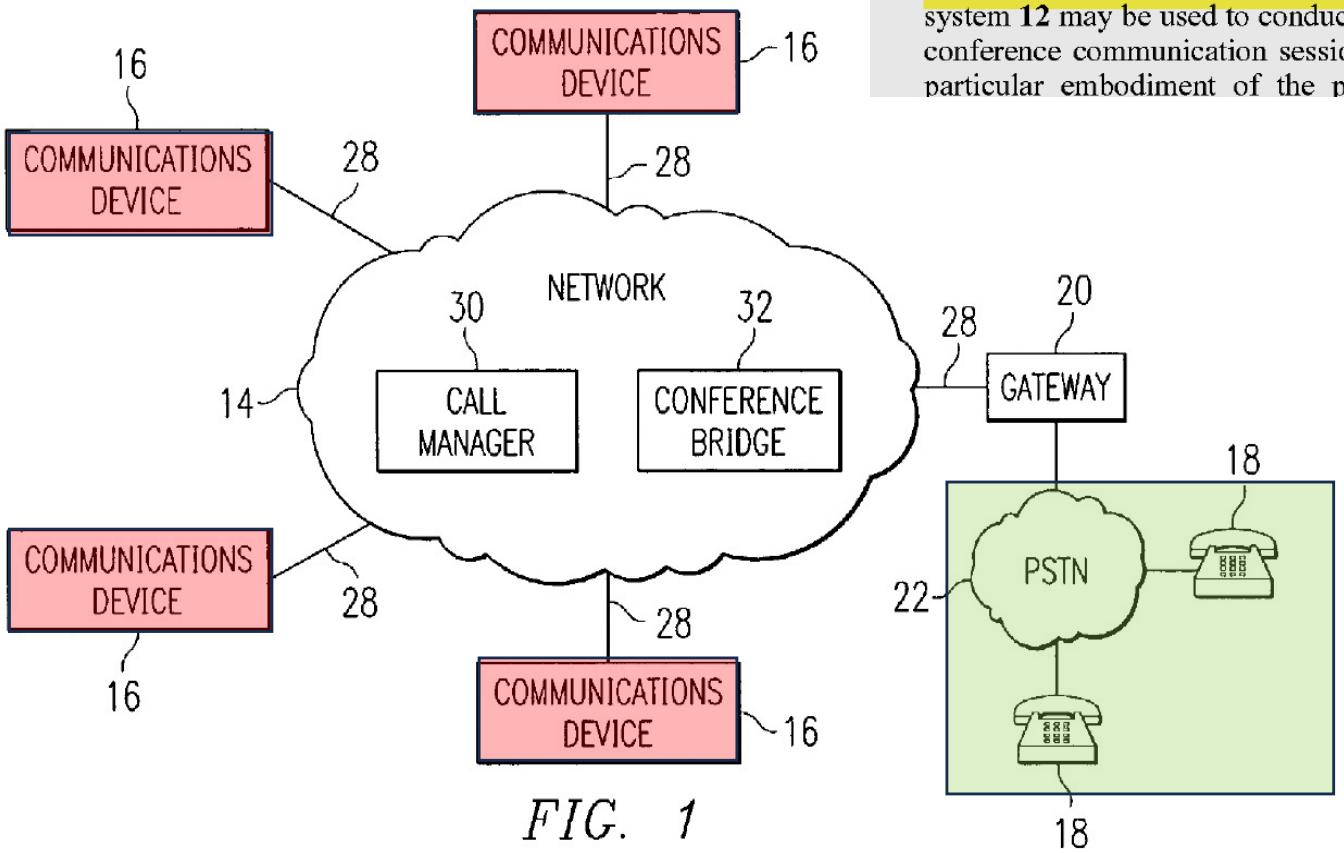
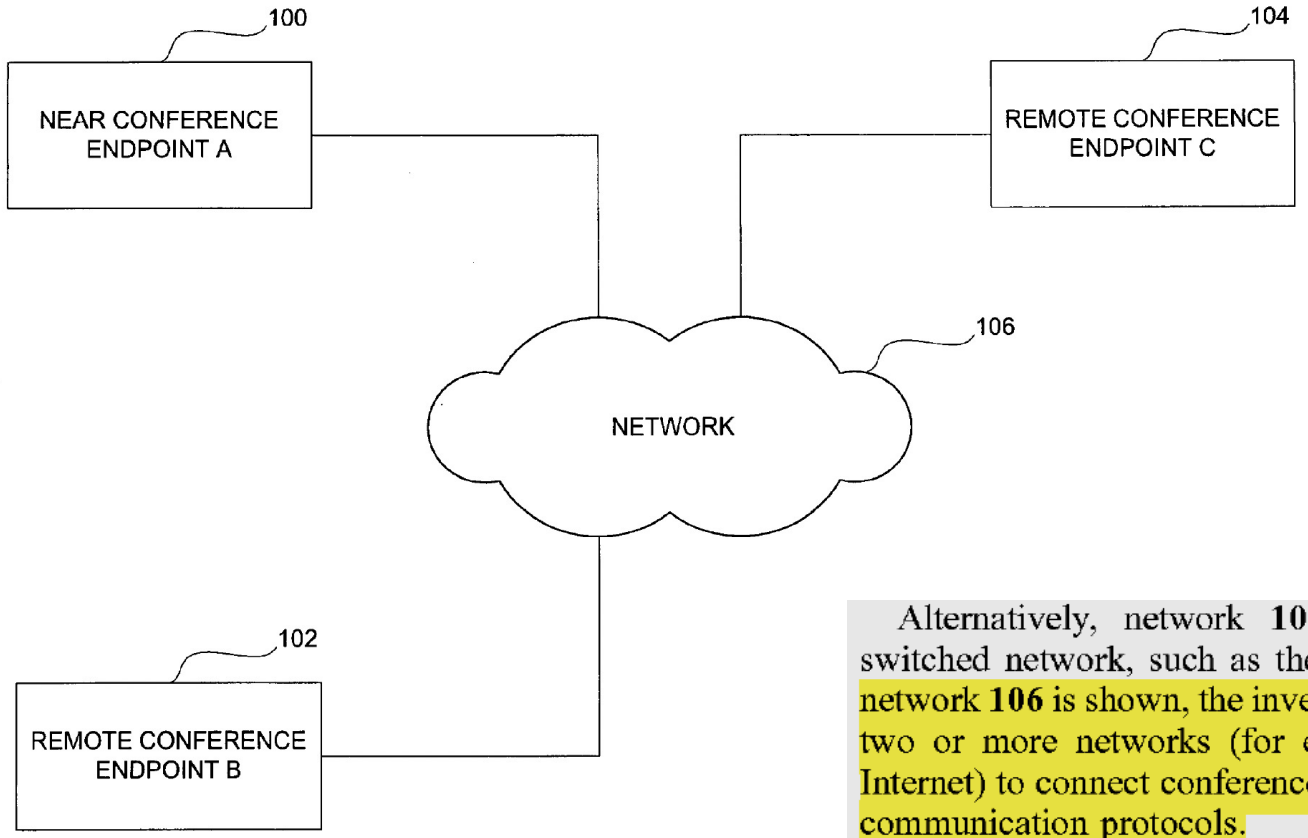


FIG. 1

Ex. 1006 at FIG. 1 (highlighting added).

See IPR2023-00377, Pet. (Paper No. 1) at 14-18, 21;
IPR2023-00380, Pet. (Paper No. 1) at 13-15;
IPR2023-00382, Pet. (Paper No. 1) at 12-14, 17;
IPR2023-00378, Pet. (Paper No. 1) at 15-16;
IPR2023-00379, Pet. (Paper No. 1) at 14-16;
IPR2023-00371, Pet. (Paper No. 1) at 17-18, 23-24.

DRELL



Ex. 1008 at FIG. 1.

FIG. 1

Alternatively, network **106** may comprise a packet switched network, such as the Internet. Although a single network **106** is shown, the invention contemplates the use of two or more networks (for example, the PSTN and the Internet) to connect conference endpoints utilizing different communication protocols.

Ex. 1008 at 3:33-38.

See IPR2023-00377, Pet. (Paper No. 1) at 50-53;
IPR2023-00380, Pet. (Paper No. 1) at 12-13;
IPR2023-00382, Pet. (Paper No. 1) at 63-66;
IPR2023-00378, Pet. (Paper No. 1) at 14-15;
IPR2023-00379, Pet. (Paper No. 1) at 13-14.

ELLIOTT

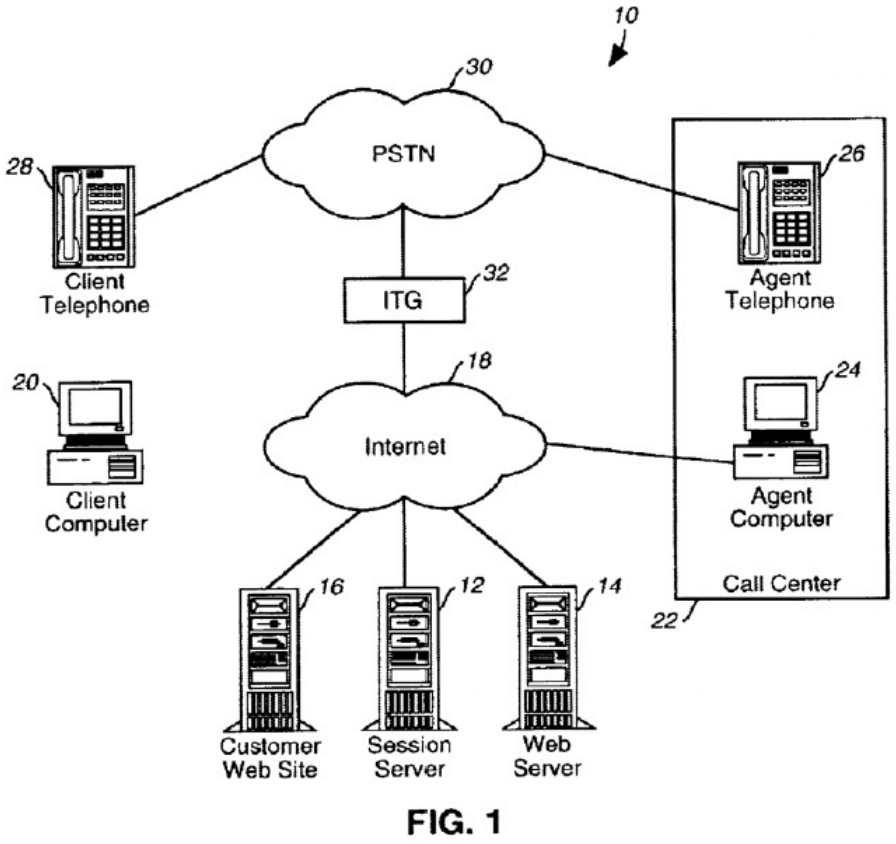


FIG. 1

Ex. 1007 at FIG. 1.

SUMMARY OF THE INVENTION

The present invention provides a method and system for communicating with a plurality of remote parties. This method and system allows for multi-media collaboration. A first party, which may be an agent in a call center, receives requests for access from first and second remote parties

Ex. 1007 at 1:51-54.

Protocol. An internet which is owned by a single entity sometimes referred to as an intranet. Network **18** can comprise an intranet, which is or is not connected to the Internet. For example, network **18** can be either a public or a carrier supported virtual private network (VPN) or local area network (LAN) which uses IP or IP-tunneling. Other protocols can alternatively be utilized.

Ex. 1007 at 4:29-34.

Client computer **20** may also be equipped with other software such as Internet telephone software, for placing Internet telephony calls. A client may also have a traditional telephone **28** for placing calls to call center agents over the public switched telephone network (PSTN) **30**.

Session server **12** is software that manages communica-

Ex. 1007 at 5:34-39.

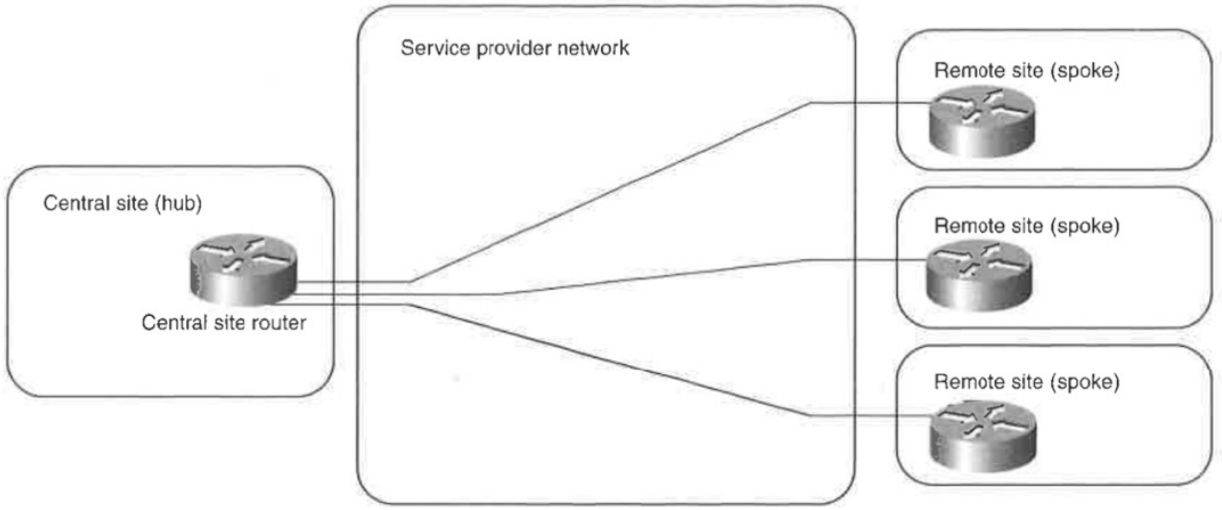
See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21;
IPR2023-00380, Pet. (Paper No. 1) at 15, 22;
IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18;
IPR2023-00378, Pet. (Paper No. 1) at 54-56;
IPR2023-00379, Pet. (Paper No. 1) at 16, 46;
IPR2023-00371, Pet. (Paper No. 1) at 19-20.

VPN TEXTBOOK

Hub-and-spoke Topology

The most commonly encountered topology is a hub-and-spoke topology, where a number of remote offices (spokes) are connected to a central site (hub), similar to the setup in Figure 7-10. The remote offices usually can exchange data (there are no explicit security restrictions on inter-office traffic), but the amount of data exchanged between them is negligible. The hub-and-spoke topology is used typically in organizations with strict hierarchical structures, for example, banks, governments, retail stores, international organizations with small in-country offices, and so on.

Figure 7-10 *Hub-and-spoke Topology*



Ex. 1010 at FIG. 7-10 (p. 130).

See IPR2023-00377, Pet. (Paper No. 1) at 20;
IPR2023-00382, Pet. (Paper No. 1) at 15-16;
IPR2023-00378, Pet. (Paper No. 1) at 54;
IPR2023-00371, Pet. (Paper No. 1) at 20-21.

The VPN Textbook Is Relevant Prior Art for the Board to Consider in the IPRs

IPR2023-0377, Grounds 1-3
(U.S. Patent No. 9,253,332)

IPR2023-00378, Grounds 2, 4
(U.S. Patent No. 9,094,525)

IPR2023-00382, Grounds 1-2
(U.S. Patent No. 9,843,612)

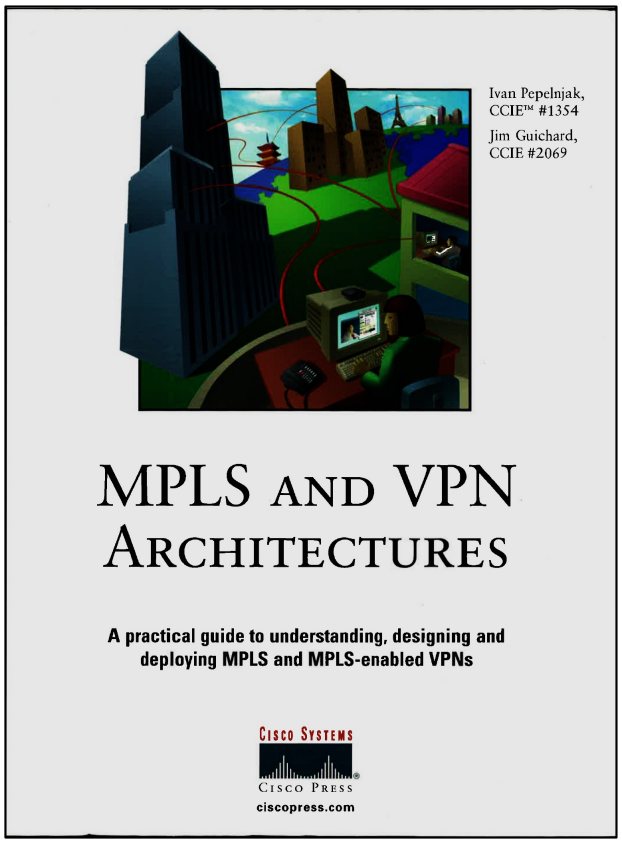
IPR 2023-00371. Grounds 1-2
(U.S. Patent No. 8,339,997)

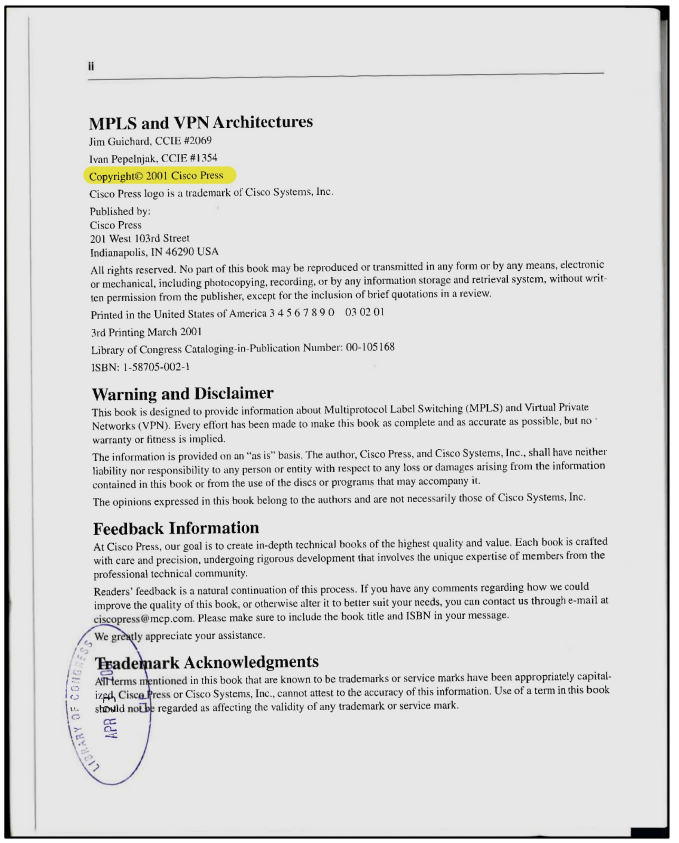
VPN TEXTBOOK (EX. 10) IS RELEVANT PRIOR ART TO BE CONSIDERED BY THE BOARD

- Certain instituted grounds rely on VPN Textbook with respect to "VPN tunnel" limitations:
  - '332 patent, claims 1-12 (IPR2023-00377)
  - '612 patent, claims 19-29 (IPR2023-00382)
  - '525 patent, claims 5-8 (IPR2023-00378)
  - '997 patent, claims 1-10 (IPR2023-00371)
- No Prior Finding by the Board Regarding the VPN Textbook or Its Teachings
  - Consideration of other Cisco published textbooks during prosecution has no bearing on the disclosures in the VPN Textbook. *See also Hum Indus. Tech., Inc. v. Amsted Rail Co., Inc.*, IPR 2023-00539, Paper No. 10 (PTAB 2023) (declining to exercise discretion under § 325(d) even where primary references were cited in IDS during prosecution).

See IPR2023-00377, Reply (Paper No. 13) at 3-4;
IPR2023-00382, Reply (Paper No. 13) at 3-5;
IPR2023-00378, Reply (Paper No. 14) at 9-11;
IPR2023-00371, Reply (Paper No. 13) at 3-4.

VPN TEXTBOOK IS RELEVANT PRIOR ART TO BE CONSIDERED BY THE BOARD





ii

MPLS and VPN Architectures

Jim Guichard, CCIE #2069

Ivan Pepelnjak, CCIE #1354

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Ex. 1010 at 001, 003.

- VPN Textbook is prior art published two years before the patents' earliest claimed priority date

See IPR2023-00377, Pet. (Paper No. 1) at 4;
IPR2023-00382, Pet. (Paper No. 1) at 4;
IPR2023-00378, Pet. (Paper No. 1) at 4;
IPR2023-00371, Pet. (Paper No. 1) at 4.

The Prior Art Renders Obvious the "VPN" Limitations

IPR2023-0377, Grounds 1-3
(U.S. Patent No. 9,253,332)

IPR2023-00380, Ground 1
(U.S. Patent No. 9,843,612)

IPR2023-00382, Grounds 1-2
(U.S. Patent No. 9,843,612)

IPR2023-00378, Grounds 2, 4
(U.S. Patent No. 9,094,525)

IPR2023-00379, Ground 1
(U.S. Patent No. 9,094,525)

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE/DRELL WITH ELLIOTT

KNAPPE

(Primary reference for IPR2023-00377, -00382, -00371)

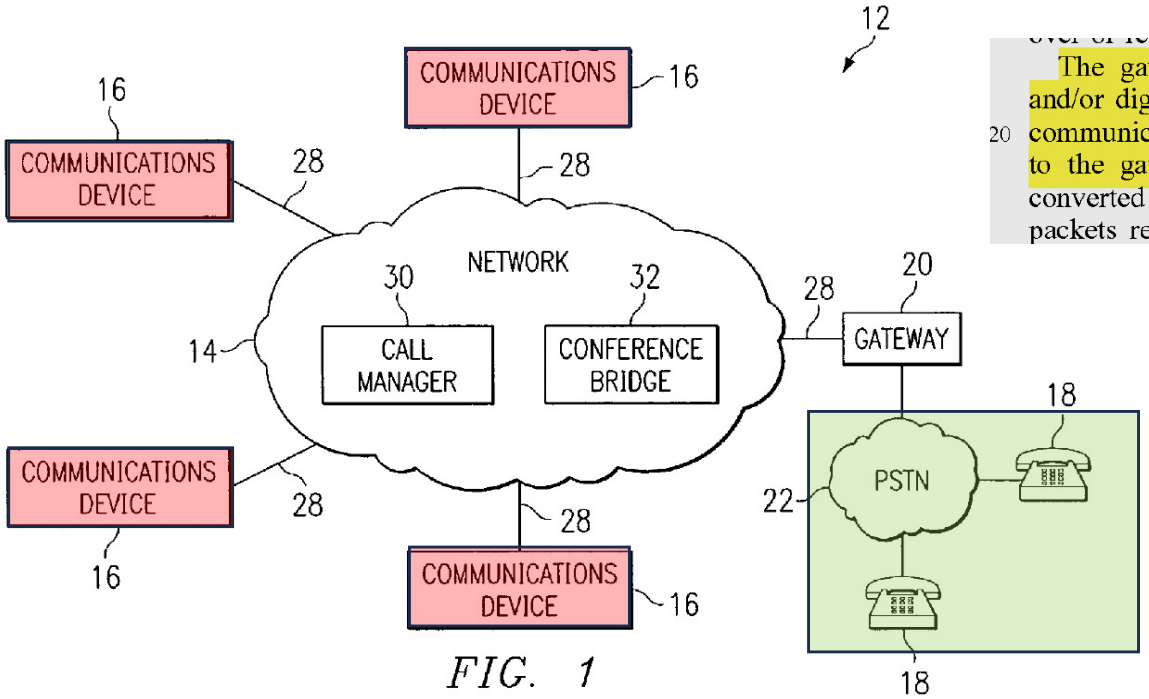


FIG. 1

The gateway **20** provides conversion between analog and/or digital formats. The standard analog telephones **18** communicate standard telephony signals through PSTN **22** to the gateway **20**. At the gateway **20**, the signals are converted to IP packets in the VoIP format. Similarly, VoIP packets received from the network **14** are converted into

Ex. 1006 at 4:19-21.

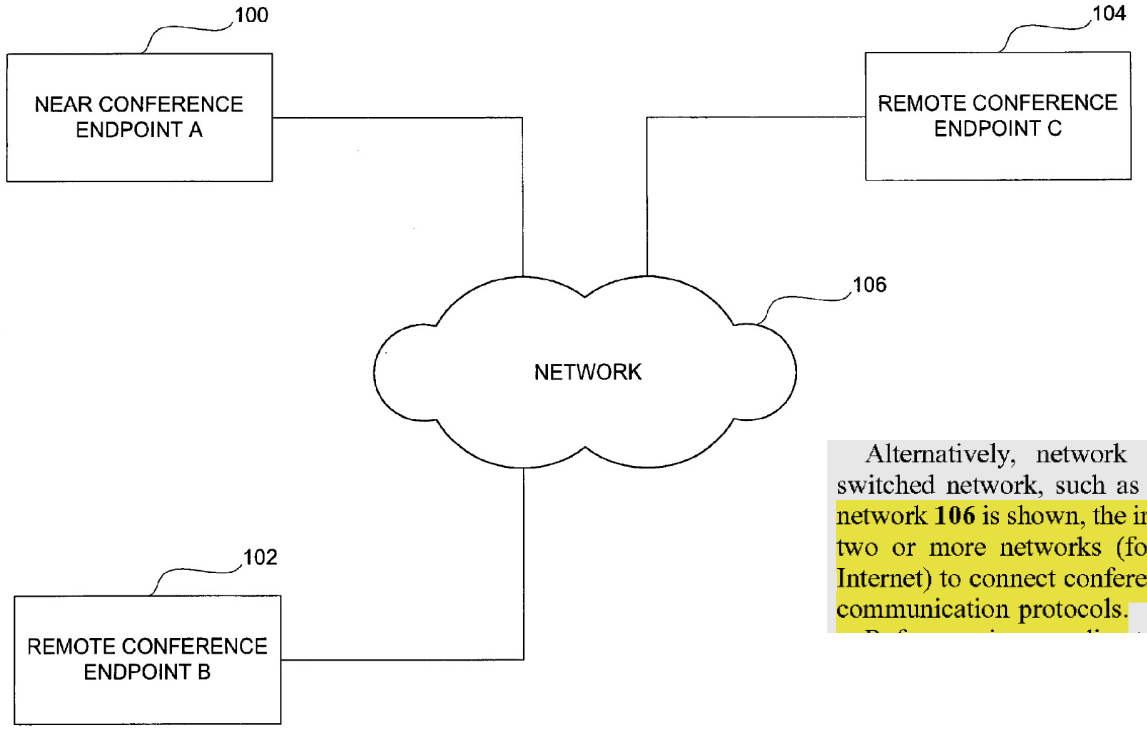
Ex. 1006 at FIG. 1 (highlighting added).

See IPR2023-00377, Pet. (Paper No. 1) at 14-18, 21, 23-25;
IPR2023-00382, Pet. (Paper No. 1) at 12-14, 17, 20-22;
IPR2023-00371, Pet. (Paper No. 1) at 17-18, 23-24, 28-30.

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE/DRELL WITH ELLIOTT

DRELL

(Primary reference for IPR2023-00380, -00378, -00379 in combination with Knappe)



Ex. 1008 at FIG. 1.

FIG. 1

Alternatively, network **106** may comprise a packet switched network, such as the Internet. Although a single network **106** is shown, the invention contemplates the use of two or more networks (for example, the PSTN and the Internet) to connect conference endpoints utilizing different communication protocols.

Ex. 1008 at 3:33-38.

See IPR2023-00380, Pet. (Paper No. 1) at 12-13, 21-22, 24-25;
IPR2023-00378, Pet. (Paper No. 1) at 14-15, 55-57;
IPR2023-00379, Pet. (Paper No. 1) at 13-14, 53-54.

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE/DRELL WITH ELLIOTT

ELLIOTT

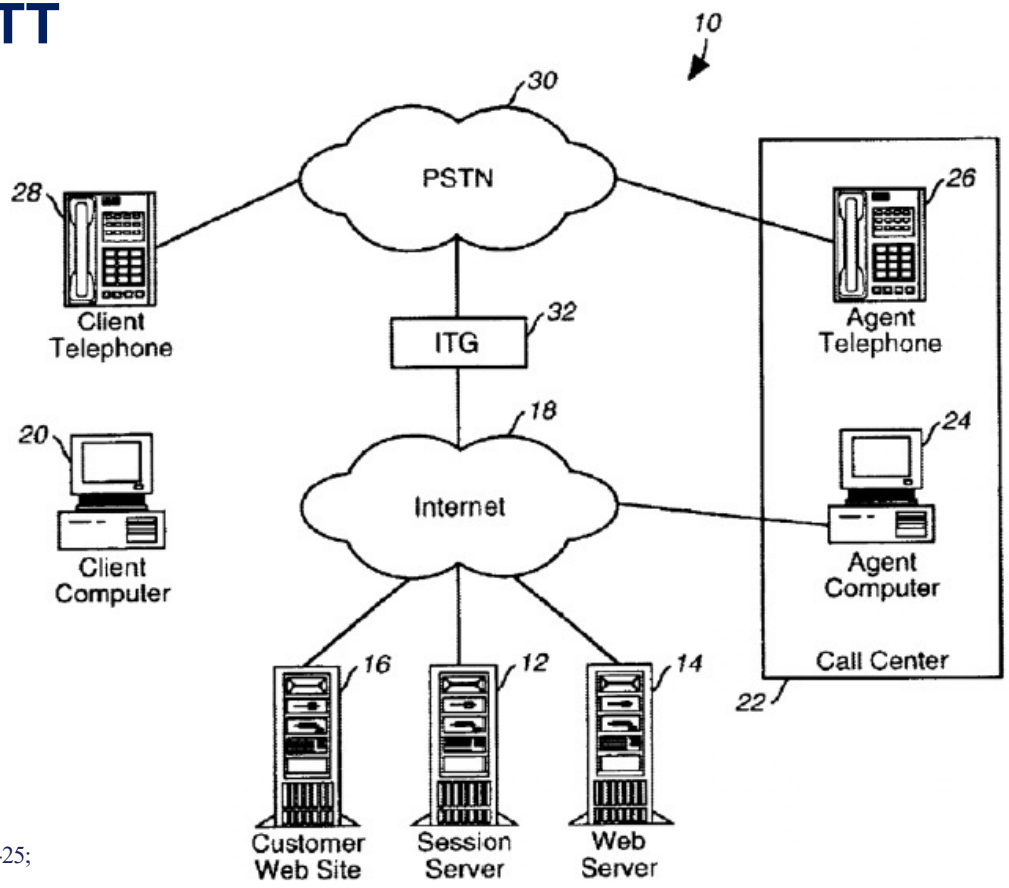
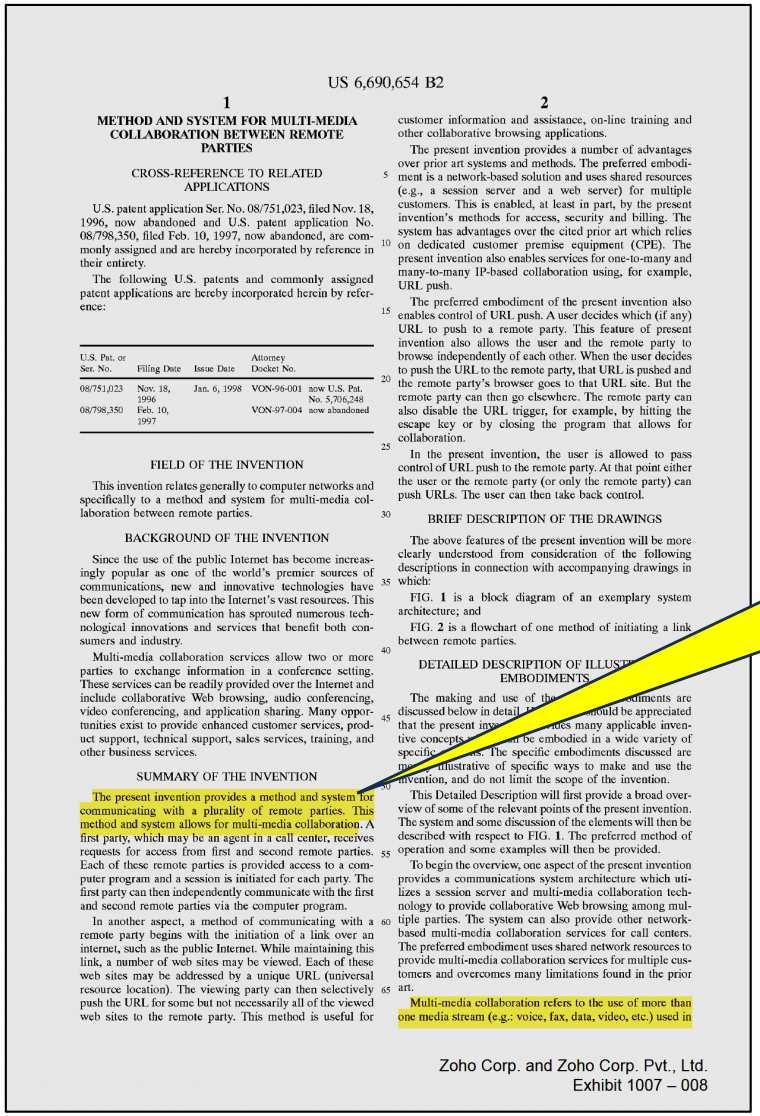


FIG. 1

See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21, 23-25;
IPR2023-00380, Pet. (Paper No. 1) at 15, 21-22, 24-25;
IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18; 20-22;
IPR2023-00378, Pet. (Paper No. 1) at 54-57;
IPR2023-00379, Pet. (Paper No. 1) at 16, 46, 53-54;
IPR2023-00371, Pet. (Paper No. 1) at 19-20, 23-24, 28-30.

Ex. 1007 at FIG. 1.

ELLIOTT DISCLOSES A SYSTEM FOR MULTI-PARTY CONFERENCING



US 6,690,654 B2

1

METHOD AND SYSTEM FOR MULTI-MEDIA COLLABORATION BETWEEN REMOTE PARTIES

CROSS-REFERENCE TO RELATED APPLICATIONS

U.S. patent application Ser. No. 08/751,023, filed Nov. 18, 1996, now abandoned and U.S. patent application No. 08/798,350, filed Feb. 10, 1997, now abandoned, are commonly assigned and are hereby incorporated by reference in their entirety.

The following U.S. patents and commonly assigned patent applications are hereby incorporated herein by reference:

U.S. Pat. or Ser. No.	Filing Date	Issue Date	Attorney Docket No.	
08/751,023	Nov. 18, 1996	Jan. 6, 1998	VON-96-001	now U.S. Pat. No. 5,706,248
08/798,350	Feb. 10, 1997		VON-97-004	now abandoned

FIELD OF THE INVENTION

This invention relates generally to computer networks and specifically to a method and system for multi-media collaboration between remote parties.

BACKGROUND OF THE INVENTION

Since the use of the public Internet has become increasingly popular as one of the world's premier sources of communications, new and innovative technologies have been developed to tap into the Internet's vast resources. This new form of communication has sprouted numerous technological innovations and services that benefit both consumers and industry.

Multi-media collaboration services allow two or more parties to exchange information in a conference setting. These services can be readily provided over the Internet and include collaborative Web browsing, audio conferencing, video conferencing, and application sharing. Many opportunities exist to provide enhanced customer services, product support, technical support, sales services, training, and other business services.

SUMMARY OF THE INVENTION

The present invention provides a method and system for communicating with a plurality of remote parties. This method and system allows for multi-media collaboration. A first party, which may be an agent in a call center, receives requests for access from first and second remote parties. Each of these remote parties is provided access to a computer program and a session is initiated for each party. The first party can then independently communicate with the first and second remote parties via the computer program.

In another aspect, a method of communicating with a remote party begins with the initiation of a link over an internet, such as the public Internet. While maintaining this link, a number of web sites may be viewed. Each of these web sites may be addressed by a unique URL (universal resource location). The viewing party can then selectively push the URL for some but not necessarily all of the viewed web sites to the remote party. This method is useful for

2

customer information and assistance, on-line training and other collaborative browsing applications.

The present invention provides a number of advantages over prior art systems and methods. The preferred embodiment is a network-based solution and uses shared resources (e.g., a session server and a web server) for multiple customers. This is enabled, at least in part, by the present invention's methods for access, security and billing. The system has advantages over the cited prior art which relies on dedicated customer premise equipment (CPE). The present invention also enables services for one-to-many and many-to-many IP-based collaboration using, for example, URL push.

The preferred embodiment of the present invention also enables control of URL push. A user decides which (if any) URL to push to a remote party. This feature of present invention also allows the user and the remote party to browse independently of each other. When the user decides to push the URL to the remote party, that URL is pushed and the remote party's browser goes to that URL site. But the remote party can then go elsewhere. The remote party can also disable the URL trigger, for example, by hitting the escape key or by closing the program that allows for collaboration.

In the present invention, the user is allowed to pass control of URL push to the remote party. At that point either the user or the remote party (or only the remote party) can push URLs. The user can then take back control.

BRIEF DESCRIPTION OF THE DRAWINGS

The above features of the present invention will be more clearly understood from consideration of the following descriptions in connection with accompanying drawings in which:

FIG. **1** is a block diagram of an exemplary system architecture; and

FIG. **2** is a flowchart of one method of initiating a link between remote parties.

DETAILED DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

The making and use of the [illegible] uments are discussed below in detail. [illegible] should be appreciated that the present inv[illegible] provides many applicable inventive concepts, which may be embodied in a wide variety of specific contexts. The specific embodiments discussed are merely illustrative of specific ways to make and use the invention, and do not limit the scope of the invention.

This Detailed Description will first provide a broad overview of some of the relevant points of the present invention. The system and some discussion of the elements will then be described with respect to FIG. **1**. The preferred method of operation and some examples will then be provided.

To begin the overview, one aspect of the present invention provides a communications system architecture which utilizes a session server and multi-media collaboration technology to provide collaborative Web browsing among multiple parties. The system can also provide other network-based multi-media collaboration services for call centers. The preferred embodiment uses shared network resources to provide multi-media collaboration services for multiple customers and overcomes many limitations found in the prior art.

Multi-media collaboration refers to the use of more than one media stream (e.g.: voice, fax, data, video, etc.) used in

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Exhibit 1007 – 008

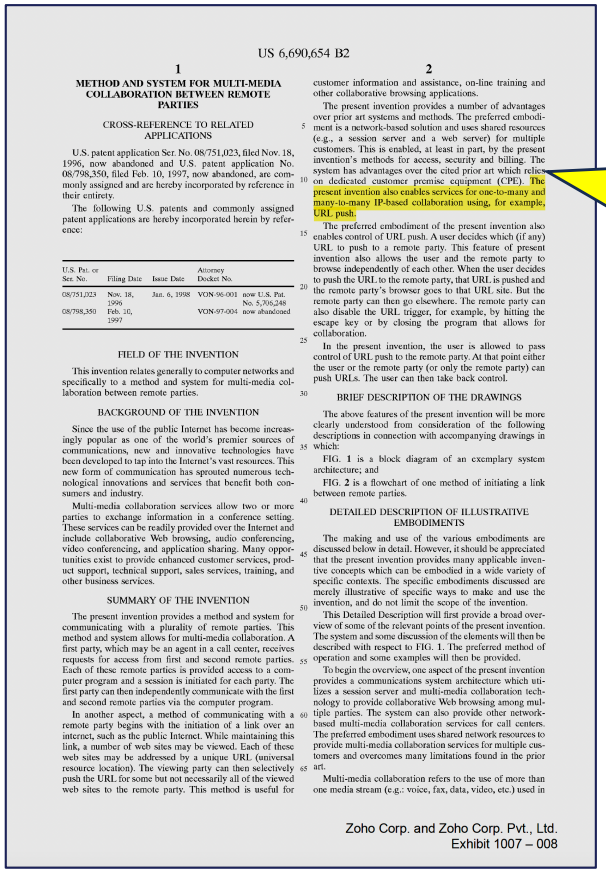
The present invention provides a method and system for **communicating with a plurality of remote parties**. This method and system allows for **multi-media collaboration**

Ex. 1007 at 1:51-54.

See also IPR2023-00380, -382, Ex. 1003 ¶¶ 132-133; IPR2023-00377, Ex. 1003 ¶¶ 128-129; IPR2023-00378, -379, Ex. 1003 ¶¶ 297-298.

See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21; IPR2023-00380, Pet. (Paper No. 1) at 15, 22; IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18; IPR2023-00378, Pet. (Paper No. 1) at 54-56; IPR2023-00379, Pet. (Paper No. 1) at 16, 46; IPR2023-00371, Pet. (Paper No. 1) at 19-20.

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METHOD AND SYSTEM FOR MULTI-MEDIA COLLABORATION BETWEEN REMOTE PARTIES

CROSS-REFERENCE TO RELATED APPLICATIONS

U.S. patent application Ser. No. 08/751,023, filed Nov. 18, 1996, now abandoned and U.S. patent application No. 08/798,350, filed Feb. 10, 1997, now abandoned, are commonly assigned and are hereby incorporated by reference in their entirety.

The following U.S. patents and commonly assigned patent applications are hereby incorporated herein by reference:

U.S. Pat. or Ser. No.	Filing Date	Issue Date	Attorney Docket No.
08/751,023	Nov. 18, 1996	Jan. 6, 1998	VON-96-001 now U.S. Pat. No. 5,706,248
08/798,350	Feb. 10, 1997		VON-97-004 now abandoned

FIELD OF THE INVENTION

This invention relates generally to computer networks and specifically to a method and system for multi-media collaboration between remote parties.

BACKGROUND OF THE INVENTION

Since the use of the public Internet has become increasingly popular as one of the world's premier sources of communications, new and innovative technologies have been developed to tap into the Internet's vast resources. This new form of communication has sprouted numerous technological innovations and services that benefit both consumers and industry.

Multi-media collaboration services allow two or more parties to exchange information in a conference setting. These services can be readily provided over the Internet and include collaborative Web browsing, audio conferencing, video conferencing, and application sharing. Many opportunities exist to provide enhanced customer services, product support, technical support, sales services, training, and other business services.

SUMMARY OF THE INVENTION

The present invention provides a method and system for communicating with a plurality of remote parties. This method and system allows for multi-media collaboration. A first party, which may be an agent in a call center, receives requests for access from first and second remote parties. Each of these remote parties is provided access to a computer program and a session is initiated for each party. The first party can then independently communicate with the first and second remote parties via the computer program.

In another aspect, a method of communicating with a remote party begins with the initiation of a link over an internet, such as the public Internet. While maintaining this link, a number of web sites may be viewed. Each of these web sites may be addressed by a unique URL (universal resource location). The viewing party can then selectively push the URL for some but not necessarily all of the viewed web sites to the remote party. This method is useful for

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customer information and assistance, on-line training and other collaborative browsing applications.

The present invention provides a number of advantages over prior art systems and methods. The preferred embodiment is a network-based solution and uses shared resources (e.g., a session server and a web server) for multiple customers. This is enabled, at least in part, by the present invention's methods for access, security and billing. The system has advantages over the cited prior art which relies on dedicated customer premise equipment (CPE). The present invention also enables services for one-to-many and many-to-many IP-based collaboration using, for example, URL push.

The preferred embodiment of the present invention also enables control of URL push. A user decides which (if any) URL to push to a remote party. This feature of present invention also allows the user and the remote party to browse independently of each other. When the user decides to push the URL to the remote party, that URL is pushed and the remote party's browser goes to that URL site. But the remote party can then go elsewhere. The remote party can also disable the URL trigger, for example, by hitting the escape key or by closing the program that allows for collaboration.

In the present invention, the user is allowed to pass control of URL push to the remote party. At that point either the user or the remote party (or only the remote party) can push URLs. The user can then take back control.

BRIEF DESCRIPTION OF THE DRAWINGS

The above features of the present invention will be more clearly understood from consideration of the following descriptions in connection with accompanying drawings in which:

FIG. **1** is a block diagram of an exemplary system architecture; and

FIG. **2** is a flowchart of one method of initiating a link between remote parties.

DETAILED DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

The making and use of the various embodiments are discussed below in detail. However, it should be appreciated that the present invention provides many applicable inventive concepts which can be embodied in a wide variety of specific contexts. The specific embodiments discussed are merely illustrative of specific ways to make and use the invention, and do not limit the scope of the invention.

This Detailed Description will first provide a broad overview of some of the relevant points of the present invention. The system and some discussion of the elements will then be described with respect to FIG. 1. The preferred method of operation and some examples will then be provided.

To begin the overview, one aspect of the present invention provides a communications system architecture which utilizes a session server and multi-media collaboration technology to provide collaborative Web browsing among multiple parties. The system can also provide other network-based multi-media collaboration services for call centers. The preferred embodiment uses shared network resources to provide multi-media collaboration services for multiple customers and overcomes many limitations found in the prior art.

Multi-media collaboration refers to the use of more than one media stream (e.g.: voice, fax, data, video, etc.) used in

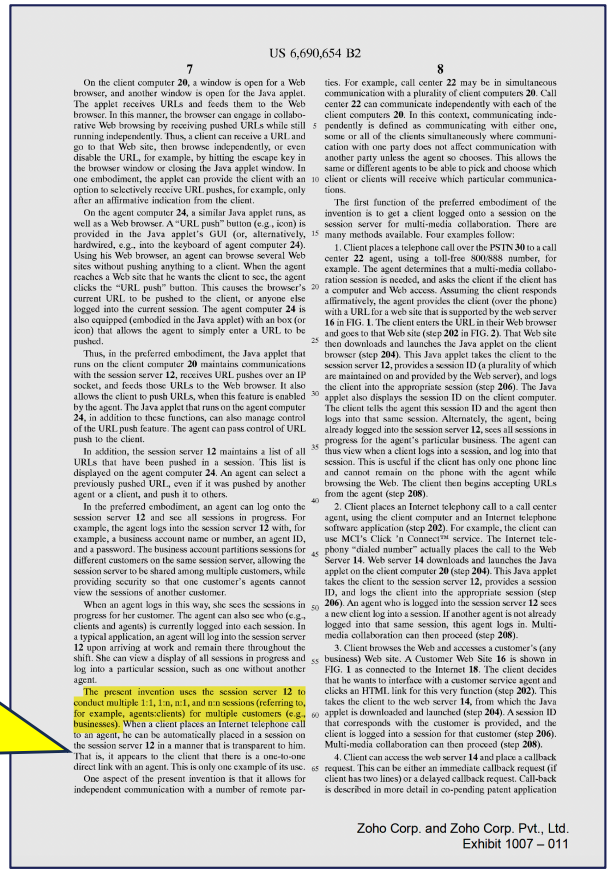
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Exhibit 1007 – 008

The present invention also enables services for one-to-one and **many-to-many IP-based collaboration**[.]

Ex. 1007 at 2:10-13.

The present invention uses the session server 12 to conduct multiple 1:1, **1:n, n:1, and n:n sessions** . . . for multiple customers[.]

Ex. 1007 at 7:58-61.



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On the client computer **20**, a window is open for a Web browser, and another window is open for the Java applet. The applet receives URLs and feeds them to the Web browser. In this manner, the browser can engage in collaborative Web browsing by receiving pushed URLs while still running independently. Thus, a client can receive a URL and go to that Web site, then browse independently, or even disable the URL, for example, by hitting the escape key in the browser window or closing the Java applet window. In one embodiment, the applet can provide the client with an option to selectively receive URL pushes, for example, only after an affirmative indication from the client.

On the agent computer **24**, a similar Java applet runs, as well as a Web browser. A "URL push" button (e.g., icon) is provided in the Java applet's GUI (or, alternatively, hardwired, e.g., into the keyboard of agent computer **24**). Using his Web browser, an agent can browse several Web sites without pushing anything to a client. When the agent reaches a Web site that he wants the client to see, the agent clicks the "URL push" button. This causes the browser's current URL to be pushed to the client, or anyone else logged into the current session. The agent computer **24** is also equipped (embodied in the Java applet) with an box (or icon) that allows the agent to simply enter a URL to be pushed.

Thus, in the preferred embodiment, the Java applet that runs on the client computer **20** maintains communications with the session server **12**, receives URL pushes over an IP socket, and feeds those URLs to the Web browser. It also allows the client to push URLs, when this feature is enabled by the agent. The Java applet that runs on the agent computer **24**, in addition to these functions, can also manage control of the URL push feature. The agent can pass control of URL push to the client.

In addition, the session server **12** maintains a list of all URLs that have been pushed in a session. This list is displayed on the agent computer **24**. An agent can select a previously pushed URL, even if it was pushed by another agent or a client, and push it to others.

In the preferred embodiment, an agent can log onto the session server **12** and see all sessions in progress. For example, the agent logs into the session server **12** with, for example, a business account name or number, an agent ID, and a password. The business account partitions sessions for different customers on the same session server, allowing the session server to be shared among multiple customers, while providing security so that one customer's agents cannot view the sessions of another customer.

When an agent logs in this way, she sees the sessions in progress for her customer. The agent can also see who (e.g., clients and agents) is currently logged into each session. In a typical application, an agent will log into the session server **12** upon arriving at work and remain there throughout the shift. She can view a display of all sessions in progress and log into a particular session, such as one without another agent.

The present invention uses the session server **12** to conduct multiple 1:1, 1:n, n:1, and n:n sessions (referring to, for example, agents:clients) for multiple customers (e.g., businesses). When a client places an Internet telephone call to an agent, he can be automatically placed in a session on the session server **12** in a manner that is transparent to him. That is, it appears to the client that there is a one-to-one direct link with an agent. This is only one example of its use.

One aspect of the present invention is that it allows for independent communication with a number of remote parties. For example, call center **22** may be in simultaneous communication with a plurality of client computers **20**. Call center **22** can communicate independently with each of the client computers **20**. In this context, communicating independently is defined as communicating with either one, some or all of the clients simultaneously where communication with one party does not affect communication with another party unless the agent so chooses. This allows the same or different agents to be able to pick and choose which client or clients will receive which particular communications.

The first function of the preferred embodiment of the invention is to get a client logged onto a session on the session server for multi-media collaboration. There are many methods available. Four examples follow:

1. Client places a telephone call over the PSTN **30** to a call center **22** agent, using a toll-free 800/888 number, for example. The agent determines that a multi-media collaboration session is needed, and asks the client if the client has a computer and Web access. Assuming the client responds affirmatively, the agent provides the client (over the phone) with a URL for a web site that is supported by the web server **16** in FIG. **1**. The client enters the URL in their Web browser and goes to that Web site (step **202** in FIG. **2**). That Web site then downloads and launches the Java applet on the client browser (step **204**). This Java applet takes the client to the session server **12**, provides a session ID (a plurality of which are maintained on and provided by the Web server), and logs the client into the appropriate session (step **206**). The Java applet also displays the session ID on the client computer. The client tells the agent this session ID and the agent then logs into that same session. Alternately, the agent, being already logged into the session server **12**, sees all sessions in progress for the agent's particular business. The agent can thus view when a client logs into a session, and log into that session. This is useful if the client has only one phone line and cannot remain on the phone with the agent while browsing the Web. The client then begins accepting URLs from the agent (step **208**).

2. Client places an Internet telephony call to a call center agent, using the client computer and an Internet telephone software application (step **202**). For example, the client can use MCI's Click 'n Connect℠ service. The Internet telephony "dialed number" actually places the call to the Web Server **14**. Web server **14** downloads and launches the Java applet on the client computer **20** (step **204**). This Java applet takes the client to the session server **12**, provides a session ID, and logs the client into the appropriate session (step **206**). An agent who is logged into the session server **12** sees a new client log into a session. If another agent is not already logged into that same session, this agent logs in. Multi-media collaboration can then proceed (step **208**).

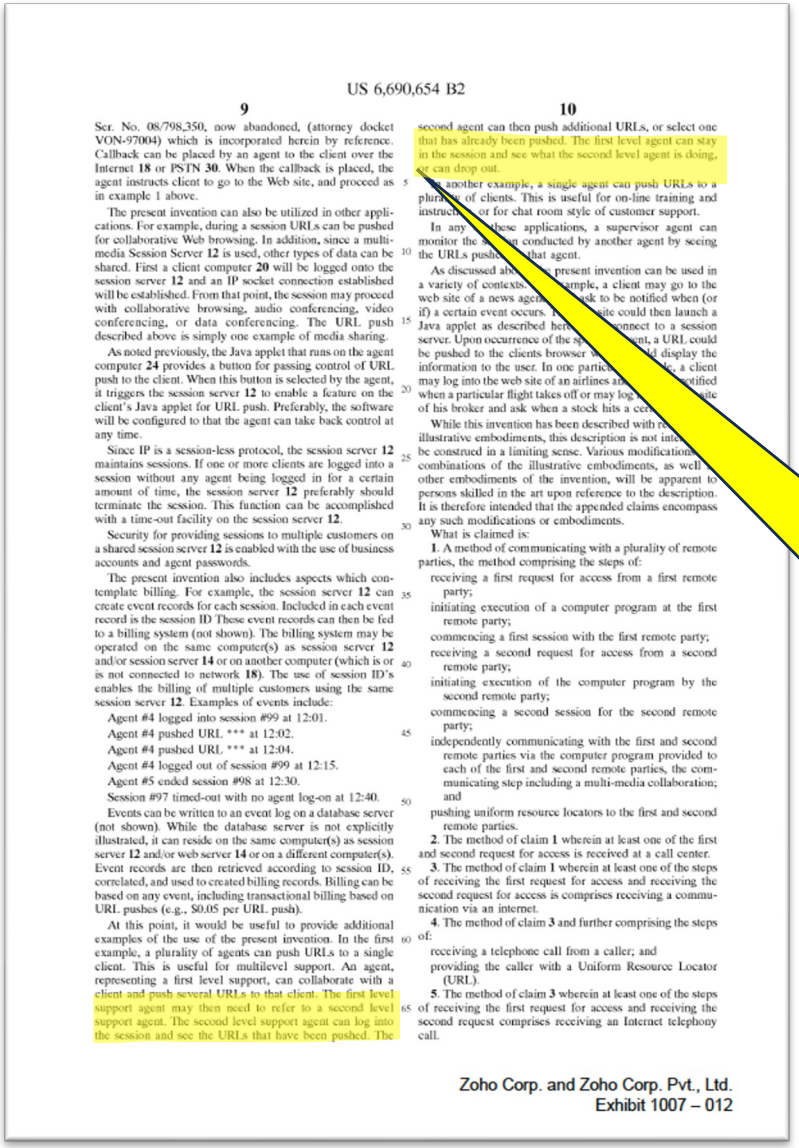
3. Client browses the Web and accesses a customer's (any business) Web site. A Customer Web Site **16** is shown in FIG. **1** as connected to the Internet **18**. The client decides that he wants to interface with a customer service agent and clicks an HTML link for this very function (step **202**). This takes the client to the web server **14**, from which the Java applet is downloaded and launched (step **204**). A session ID that corresponds with the customer is provided, and the client is logged into a session for that customer (step **206**). Multi-media collaboration can then proceed (step **208**).

4. Client can access the web server **14** and place a callback request. This can be either an immediate callback request (if client has two lines) or a delayed callback request. Call-back is described in more detail in co-pending patent application

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See IPR2023-00377, Reply (Paper No. 13) at 9;
IPR2023-00380, Reply (Paper No. 13) at 10;
IPR2023-00382, Reply (Paper No. 13) at 11;
IPR2023-00379, Reply (Paper No. 13) at 9;
see also IPR2023-00371, Reply (Paper No. 13) at 11.

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Ser. No. 08/798,350, now abandoned, (attorney docket VON-97004) which is incorporated herein by reference. Callback can be placed by an agent to the client over the Internet **18** or PSTN **30**. When the callback is placed, the agent instructs client to go to the Web site, and proceed as in example 1 above.

The present invention can also be utilized in other applications. For example, during a session URLs can be pushed for collaborative Web browsing. In addition, since a multimedia Session Server **12** is used, other types of data can be shared. First a client computer **20** will be logged onto the session server **12** and an IP socket connection established will be established. From that point, the session may proceed with collaborative browsing, audio conferencing, video conferencing, or data conferencing. The URL push described above is simply one example of media sharing.

As noted previously, the Java applet that runs on the agent computer **24** provides a button for passing control of URL push to the client. When this button is selected by the agent, it triggers the session server **12** to enable a feature on the client's Java applet for URL push. Preferably, the software will be configured to that the agent can take back control at any time.

Since IP is a session-less protocol, the session server **12** maintains sessions. If one or more clients are logged into a session without any agent being logged in for a certain amount of time, the session server **12** preferably should terminate the session. This function can be accomplished with a time-out facility on the session server **12**.

Security for providing sessions to multiple customers on a shared session server **12** is enabled with the use of business accounts and agent passwords.

The present invention also includes aspects which contemplate billing. For example, the session server **12** can create event records for each session. Included in each event record is the session ID These event records can then be fed to a billing system (not shown). The billing system may be operated on the same computer(s) as session server **12** and/or session server **14** or on another computer (which is or is not connected to network **18**). The use of session ID's enables the billing of multiple customers using the same session server **12**. Examples of events include:

Agent #4 logged into session #99 at 12:01.
Agent #4 pushed URL *** at 12:02.
Agent #4 pushed URL *** at 12:04.
Agent #4 logged out of session #99 at 12:15.
Agent #5 ended session #98 at 12:30.
Session #97 timed-out with no agent log-on at 12:40.

Events can be written to an event log on a database server (not shown). While the database server is not explicitly illustrated, it can reside on the same computer(s) as session server **12** and/or web server **14** or on a different computer(s). Event records are then retrieved according to session ID, correlated, and used to created billing records. Billing can be based on any event, including transactional billing based on URL pushes (e.g., $0.05 per URL push).

At this point, it would be useful to provide additional examples of the use of the present invention. In the first example, a plurality of agents can push URLs to a single client. This is useful for multilevel support. An agent, representing a first level support, can collaborate with a client and push several URLs to that client. The first level support agent may then need to refer to a second level support agent. The second level support agent can log into the session and see the URLs that have been pushed. The

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second agent can then push additional URLs, or select one that has already been pushed. The first level agent can stay in the session and see what the second level agent is doing, or can drop out.

In another example, a single agent can push URLs to a plurality of clients. This is useful for on-line training and instruction or for chat room style of customer support.

In any of these applications, a supervisor agent can monitor the session conducted by another agent by seeing the URLs pushed by that agent.

As discussed above, the present invention can be used in a variety of contexts. For example, a client may go to the web site of a news agency and ask to be notified when (or if) a certain event occurs. The site could then launch a Java applet as described herein to connect to a session server. Upon occurrence of the specified event, a URL could be pushed to the clients browser which would display the information to the user. In one particular example, a client may log into the web site of an airlines and ask to be notified when a particular flight takes off or may log into the web site of his broker and ask when a stock hits a certain price.

While this invention has been described with reference to illustrative embodiments, this description is not intended to be construed in a limiting sense. Various modifications and combinations of the illustrative embodiments, as well as other embodiments of the invention, will be apparent to persons skilled in the art upon reference to the description. It is therefore intended that the appended claims encompass any such modifications or embodiments.

What is claimed is:

1. A method of communicating with a plurality of remote parties, the method comprising the steps of:

receiving a first request for access from a first remote party;

initiating execution of a computer program at the first remote party;

commencing a first session with the first remote party;

receiving a second request for access from a second remote party;

initiating execution of the computer program by the second remote party;

commencing a second session for the second remote party;

independently communicating with the first and second remote parties via the computer program provided to each of the first and second remote parties, the communicating step including a multi-media collaboration; and

pushing uniform resource locators to the first and second remote parties.

2. The method of claim **1** wherein at least one of the first and second request for access is received at a call center.

3. The method of claim **1** wherein at least one of the steps of receiving the first request for access and receiving the second request for access is comprises receiving a communication via an internet.

4. The method of claim **3** and further comprising the steps of:

receiving a telephone call from a caller; and

providing the caller with a Uniform Resource Locator (URL).

5. The method of claim **3** wherein at least one of the steps of receiving the first request for access and receiving the second request comprises receiving an Internet telephony call.

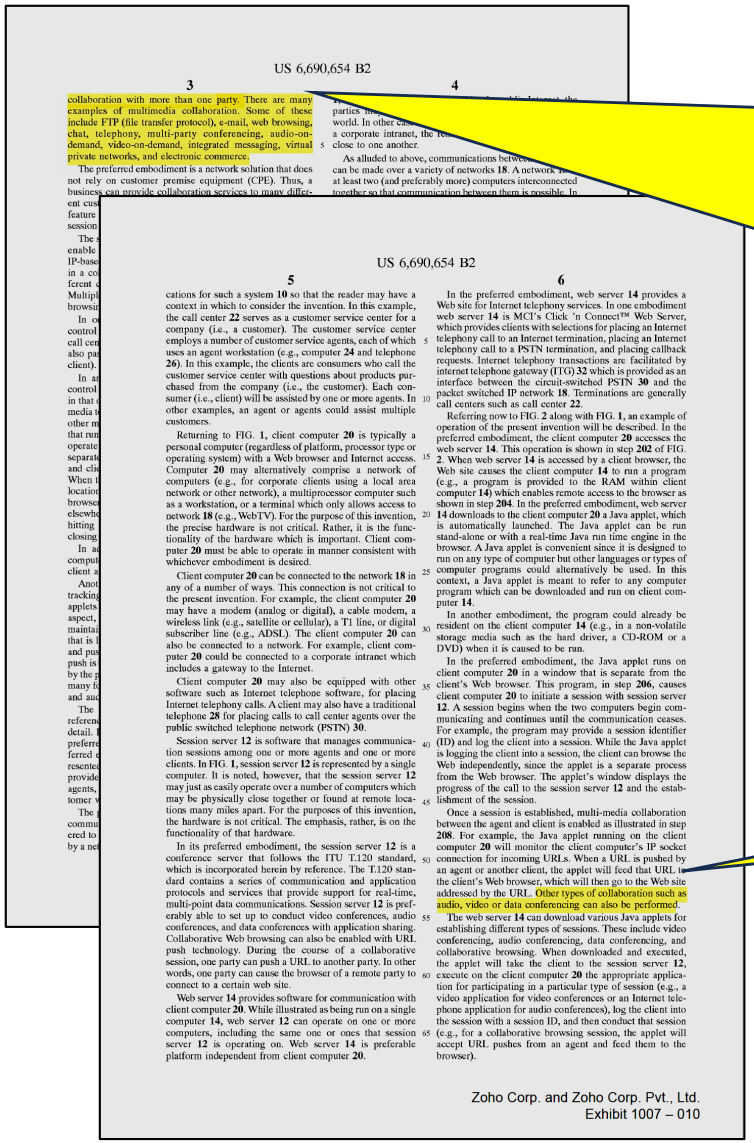
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- Elliott's session ID is not specific to one-to-one sessions:
  - Multiple agents can log to the same conference with the same session ID

The first level support agent may then need to refer to a second level support agent. The second level support agent can log into the session … The first level agent can stay in the session and see what the second level agent is doing or can drop out.

Ex. 1007 at 9:60-10-4; *see also* IPR2023-00371, Reply (Paper No. 13) at 11.

ELLIOTT TEACHES USE OF VPN AND VPN TUNNELING IN MULTI-PARTY CONFERENCING



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collaboration with more than one party. There are many examples of multimedia collaboration. Some of these include FTP (file transfer protocol), e-mail, web browsing, chat, telephony, multi-party conferencing, audio-on-demand, video-on-demand, integrated messaging, virtual private networks, and electronic commerce.

The preferred embodiment is a network solution that does not rely on customer premise equipment (CPE). Thus, a business can provide collaboration services to many differ-

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a corporate intranet, the

close to one another.

As alluded to above, communications betwe

can be made over a variety of networks 18. A network

at least two (and preferably more) computers interconnected

together so that communication between them is possible. In

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cations for such a system 10 so that the reader may have a context in which to consider the invention. In this example, the call center 22 serves as a customer service center for a company (i.e., a customer). The customer service center employs a number of customer service agents, each of which uses an agent workstation (e.g., computer 24 and telephone 26). In this example, the clients are consumers who call the customer service center with questions about products purchased from the company (i.e., the customer). Each consumer (i.e., client) will be assisted by one or more agents. In other examples, an agent or agents could assist multiple customers.

Returning to FIG. 1, client computer 20 is typically a personal computer (regardless of platform, processor type or operating system) with a Web browser and Internet access. Computer 20 may alternatively comprise a network of computers (e.g., for corporate clients using a local area network or other network), a multiprocessor computer such as a workstation, or a terminal which only allows access to network 18 (e.g., WebTV). For the purpose of this invention, the precise hardware is not critical. Rather, it is the functionality of the hardware which is important. Client computer 20 must be able to operate in manner consistent with whichever embodiment is desired.

Client computer 20 can be connected to the network 18 in any of a number of ways. This connection is not critical to the present invention. For example, the client computer 20 may have a modem (analog or digital), a cable modem, a wireless link (e.g., satellite or cellular), a T1 line, or digital subscriber line (e.g., ADSL). The client computer 20 can also be connected to a network. For example, client computer 20 could be connected to a corporate intranet which includes a gateway to the Internet.

Client computer 20 may also be equipped with other software such as Internet telephone software, for placing Internet telephony calls. A client may also have a traditional telephone 28 for placing calls to call center agents over the public switched telephone network (PSTN) 30.

Session server 12 is software that manages communication sessions among one or more agents and one or more clients. In FIG. 1, session server 12 is represented by a single computer. It is noted, however, that the session server 12 may just as easily operate over a number of computers which may be physically close together or found at remote locations many miles apart. For the purposes of this invention, the hardware is not critical. The emphasis, rather, is on the functionality of that hardware.

In its preferred embodiment, the session server 12 is a conference server that follows the ITU T.120 standard, which is incorporated herein by reference. The T.120 standard contains a series of communication and application protocols and services that provide support for real-time, multi-point data communications. Session server 12 is preferably able to set up to conduct video conferences, audio conferences, and data conferences with application sharing. Collaborative Web browsing can also be enabled with URL push technology. During the course of a collaborative session, one party can push a URL to another party. In other words, one party can cause the browser of a remote party to connect to a certain web site.

Web server 14 provides software for communication with client computer 20. While illustrated as being run on a single computer 14, web server 12 can operate on one or more computers, including the same one or ones that session server 12 is operating on. Web server 14 is preferable platform independent from client computer 20.

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In the preferred embodiment, web server 14 provides a Web site for Internet telephony services. In one embodiment web server 14 is MCI's Click 'n Connect™ Web Server, which provides clients with selections for placing an Internet telephony call to an Internet termination, placing an Internet telephony call to a PSTN termination, and placing callback requests. Internet telephony transactions are facilitated by internet telephone gateway (ITG) 32 which is provided as an interface between the circuit-switched PSTN 30 and the packet switched IP network 18. Terminations are generally call centers such as call center 22.

Referring now to FIG. 2 along with FIG. 1, an example of operation of the present invention will be described. In the preferred embodiment, the client computer 20 accesses the web server 14. This operation is shown in step 202 of FIG. 2. When web server 14 is accessed by a client browser, the Web site causes the client computer 14 to run a program (e.g., a program is provided to the RAM within client computer 14) which enables remote access to the browser as shown in step 204. In the preferred embodiment, web server 14 downloads to the client computer 20 a Java applet, which is automatically launched. The Java applet can be run stand-alone or with a real-time Java run time engine in the browser. A Java applet is convenient since it is designed to run on any type of computer but other languages or types of computer programs could alternatively be used. In this context, a Java applet is meant to refer to any computer program which can be downloaded and run on client computer 14.

In another embodiment, the program could already be resident on the client computer 14 (e.g., in a non-volatile storage media such as the hard driver, a CD-ROM or a DVD) when it is caused to be run.

In the preferred embodiment, the Java applet runs on client computer 20 in a window that is separate from the client's Web browser. This program, in step 206, causes client computer 20 to initiate a session with session server 12. A session begins when the two computers begin communicating and continues until the communication ceases. For example, the program may provide a session identifier (ID) and log the client into a session. While the Java applet is logging the client into a session, the client can browse the Web independently, since the applet is a separate process from the Web browser. The applet's window displays the progress of the call to the session server 12 and the establishment of the session.

Once a session is established, multi-media collaboration between the agent and client is enabled as illustrated in step 208. For example, the Java applet running on the client computer 20 will monitor the client computer's IP socket connection for incoming URLs. When a URL is pushed by an agent or another client, the applet will feed that URL to the client's Web browser, which will then go to the Web site addressed by the URL. Other types of collaboration such as audio, video or data conferencing can also be performed.

The web server 14 can download various Java applets for establishing different types of sessions. These include video conferencing, audio conferencing, data conferencing, and collaborative browsing. When downloaded and executed, the applet will take the client to the session server 12, execute on the client computer 20 the appropriate application for participating in a particular type of session (e.g., a video application for video conferences or an Internet telephone application for audio conferences), log the client into the session with a session ID, and then conduct that session (e.g., for a collaborative browsing session, the applet will accept URL pushes from an agent and feed them to the browser).

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Exhibit 1007 – 010

Multi-media collaboration refers to the **use of more than one media stream** (e.g.: voice, fax, data, video, etc.) **used in collaboration with more than one party**. There are many **examples of multimedia collaboration**. Some of these include FTP . . . email, web browsing, chat, telephony, **multi-party conferencing**, audio-on-demand, video-on-demand, integrated messaging, **virtual private networks**, and electronic commerce.

Ex. 1007 at 2:66-3:6.

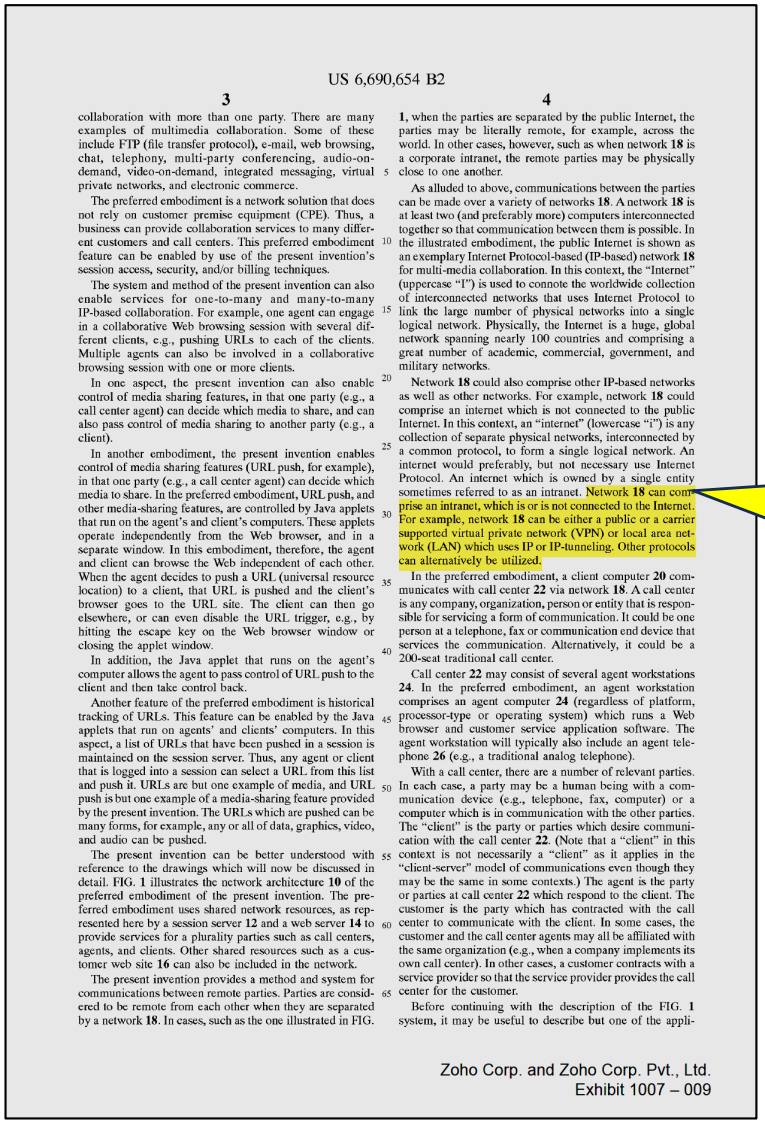
Other types of collaboration such as audio, video, or data conferencing can also be performed.

Ex. 1007 at 6:53-54.

See also IPR2023-00380, -382, Ex. 1003 ¶¶ 132-133;
IPR2023-00377, Ex. 1003 ¶¶ 128-129;
IPR2023-00378, -379, Ex. 1003 ¶¶ 297-298.

See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21;
IPR2023-00380, Pet. (Paper No. 1) at 15, 22;
IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18;
IPR2023-00378, Pet. (Paper No. 1) at 54-56;
IPR2023-00379, Pet. (Paper No. 1) at 16, 46;
IPR2023-00371, Pet. (Paper No. 1) at 19-20.

ELLIOTT TEACHES USE OF VPN AND VPN TUNNELING IN MULTI-PARTY CONFERENCING



US 6,690,654 B2

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collaboration with more than one party. There are many examples of multimedia collaboration. Some of these include FTP (file transfer protocol), e-mail, web browsing, chat, telephony, multi-party conferencing, audio-on-demand, video-on-demand, integrated messaging, virtual private networks, and electronic commerce.

The preferred embodiment is a network solution that does not rely on customer premise equipment (CPE). Thus, a business can provide collaboration services to many different customers and call centers. This preferred embodiment feature can be enabled by use of the present invention's session access, security, and/or billing techniques.

The system and method of the present invention can also enable services for one-to-many and many-to-many IP-based collaboration. For example, one agent can engage in a collaborative Web browsing session with several different clients, e.g., pushing URLs to each of the clients. Multiple agents can also be involved in a collaborative browsing session with one or more clients.

In one aspect, the present invention can also enable control of media sharing features, in that one party (e.g., a call center agent) can decide which media to share, and can also pass control of media sharing to another party (e.g., a client).

In another embodiment, the present invention enables control of media sharing features (URL push, for example), in that one party (e.g., a call center agent) can decide which media to share. In the preferred embodiment, URL push, and other media-sharing features, are controlled by Java applets that run on the agent's and client's computers. These applets operate independently from the Web browser, and in a separate window. In this embodiment, therefore, the agent and client can browse the Web independent of each other. When the agent decides to push a URL (universal resource location) to a client, that URL is pushed and the client's browser goes to the URL site. The client can then go elsewhere, or can even disable the URL trigger, e.g., by hitting the escape key on the Web browser window or closing the applet window.

In addition, the Java applet that runs on the agent's computer allows the agent to pass control of URL push to the client and then take control back.

Another feature of the preferred embodiment is historical tracking of URLs. This feature can be enabled by the Java applets that run on agents' and clients' computers. In this aspect, a list of URLs that have been pushed in a session is maintained on the session server. Thus, any agent or client that is logged into a session can select a URL from this list and push it. URLs are but one example of media, and URL push is but one example of a media-sharing feature provided by the present invention. The URLs which are pushed can be many forms, for example, any or all of data, graphics, video, and audio can be pushed.

The present invention can be better understood with reference to the drawings which will now be discussed in detail. FIG. **1** illustrates the network architecture **10** of the preferred embodiment of the present invention. The preferred embodiment uses shared network resources, as represented here by a session server **12** and a web server **14** to provide services for a plurality parties such as call centers, agents, and clients. Other shared resources such as a customer web site **16** can also be included in the network.

The present invention provides a method and system for communications between remote parties. Parties are considered to be remote from each other when they are separated by a network **18**. In cases, such as the one illustrated in FIG.

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1, when the parties are separated by the public Internet, the parties may be literally remote, for example, across the world. In other cases, however, such as when network **18** is a corporate intranet, the remote parties may be physically close to one another.

As alluded to above, communications between the parties can be made over a variety of networks **18**. A network **18** is at least two (and preferably more) computers interconnected together so that communication between them is possible. In the illustrated embodiment, the public Internet is shown as an exemplary Internet Protocol-based (IP-based) network **18** for multi-media collaboration. In this context, the "Internet" (uppercase "I") is used to connote the worldwide collection of interconnected networks that uses Internet Protocol to link the large number of physical networks into a single logical network. Physically, the Internet is a huge, global network spanning nearly 100 countries and comprising a great number of academic, commercial, government, and military networks.

Network **18** could also comprise other IP-based networks as well as other networks. For example, network **18** could comprise an internet which is not connected to the public Internet. In this context, an "internet" (lowercase "i") is any collection of separate physical networks, interconnected by a common protocol, to form a single logical network. An internet would preferably, but not necessary use Internet Protocol. An internet which is owned by a single entity sometimes referred to as an intranet. Network **18** can comprise an intranet, which is or is not connected to the Internet. For example, network **18** can be either a public or a carrier supported virtual private network (VPN) or local area network (LAN) which uses IP or IP-tunneling. Other protocols can alternatively be utilized.

In the preferred embodiment, a client computer **20** communicates with call center **22** via network **18**. A call center is any company, organization, person or entity that is responsible for servicing a form of communication. It could be one person at a telephone, fax or communication end device that services the communication. Alternatively, it could be a 200-seat traditional call center.

Call center **22** may consist of several agent workstations **24**. In the preferred embodiment, an agent workstation comprises an agent computer **24** (regardless of platform, processor-type or operating system) which runs a Web browser and customer service application software. The agent workstation will typically also include an agent telephone **26** (e.g., a traditional analog telephone).

With a call center, there are a number of relevant parties. In each case, a party may be a human being with a communication device (e.g., telephone, fax, computer) or a computer which is in communication with the other parties. The "client" is the party or parties which desire communication with the call center **22**. (Note that a "client" in this context is not necessarily a "client" as it applies in the "client-server" model of communications even though they may be the same in some contexts.) The agent is the party or parties at call center **22** which respond to the client. The customer is the party which has contracted with the call center to communicate with the client. In some cases, the customer and the call center agents may all be affiliated with the same organization (e.g., when a company implements its own call center). In other cases, a customer contracts with a service provider so that the service provider provides the call center for the customer.

Before continuing with the description of the FIG. **1** system, it may be useful to describe but one of the appli-

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Exhibit 1007 – 009

For example, **network 18 can be either a public or a carrier supported virtual private network (VPN) or a local area network (LAN) which uses IP or IP-tunneling.** Other protocols can alternatively be utilized.

Ex. 1007 at 4:29-34.
See also IPR2023-00380, -382, Ex. 1003 ¶¶ 132-133;
IPR2023-00377, Ex. 1003 ¶¶ 128-129;
IPR2023-00378, -379, Ex. 1003 ¶¶ 297-298.

See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21;
IPR2023-00380, Pet. (Paper No. 1) at 15, 22;
IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18;
IPR2023-00378, Pet. (Paper No. 1) at 54-56;
IPR2023-00379, Pet. (Paper No. 1) at 16, 46;
IPR2023-00371, Pet. (Paper No. 1) at 19-20.

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE/DRELL WITH ELLIOTT

> 169. As such, one of ordinary skill in the art would have been motivated to apply the techniques of using VPN tunneling for conferencing, as Elliott teaches, to Drell's conferencing system because VPN tunneling provides secure communications while also reducing or avoiding costs associated with leased private data lines. Indeed, the combination of Drell and Elliott is merely the ordinary use of a common technique (using VPN tunneling for conferencing data) in the similar conferencing system of Drell to yield predictable results of (1) securing the transmitted data, and (2) reducing or avoiding costs associated with leasing private data lines. Each of these benefits separately and together would have motivated a POSITA.

IPR2023-00380, -382, Ex. 1003 ¶ 169 *(see also* ¶¶ 166-172); IPR2023-00378, -379, Ex. 1003 ¶ 314 *(see also* ¶¶ 311-317).

See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21, 23-25;
IPR2023-00380, Pet. (Paper No. 1) at 15, 21-22, 24-25;
IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18; 20-22;
IPR2023-00378, Pet. (Paper No. 1) at 54-57;
IPR2023-00379, Pet. (Paper No. 1) at 16, 46, 53-54;
IPR2023-00371, Pet. (Paper No. 1) at 19-20, 23-24, 28-30.

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE/DRELL WITH ELLIOTT

> 360. Indeed, the combination of Knappe and Elliott is merely the ordinary use of the common technique of VPN tunneling of conferencing data, as disclosed in Elliott, in the similar conferencing system of Knappe to yield predictable results, including securing transmitted data and reducing costs associated with leasing private data lines.

IPR2023-00380, -382, Ex. 1003 ¶ 360 *(see also* ¶¶ 353-361); IPR2023-00377, Ex. 1003 ¶ 152 *(see also* ¶¶ 145-153)).

See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21, 23-25;
IPR2023-00380, Pet. (Paper No. 1) at 15, 21-22, 24-25;
IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18; 20-22;
IPR2023-00378, Pet. (Paper No. 1) at 54-57;
IPR2023-00379, Pet. (Paper No. 1) at 16, 46, 53-54;
IPR2023-00371, Pet. (Paper No. 1) at 19-20, 23-24, 28-30.

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE/DRELL WITH ELLIOTT

> "If there's a known technique to address a known problem using 'prior art elements according to their established functions,' then there is a motivation to combine."

Intel Corp. v. PACT XPP Schweiz AG, 61 F.4th 1373, 1380 (Fed. Cir. 2023)

See IPR2023-00377, Pet. (Paper No. 1) at 18-19, 21, 23-25 & Reply (Paper No. 13) at 9-14;
IPR2023-00380, Pet. (Paper No. 1) at 15, 21-22, 24-25;
IPR2023-00382, Pet. (Paper No. 1) at 15, 17-18; 20-22 & Reply (Paper No. 13) at 9-14;
IPR2023-00378, Pet. (Paper No. 1) at 54-57;
IPR2023-00379, Pet. (Paper No. 1) at 16, 46, 53-54 & Reply (Paper No. 12) at 9;
IPR2023-00371, Pet. (Paper No. 1) at 19-20, 23-24, 28-30.

THE VPN LIMITATIONS ARE OBVIOUS

1) The ’612 patent, independent claim 10: (IPR2023-00380, Ground 1)

- Claim 10: “**receiving second audio data** at the local moderator client from a remote **client through a Virtual Private Network (VPN) tunnel.**”

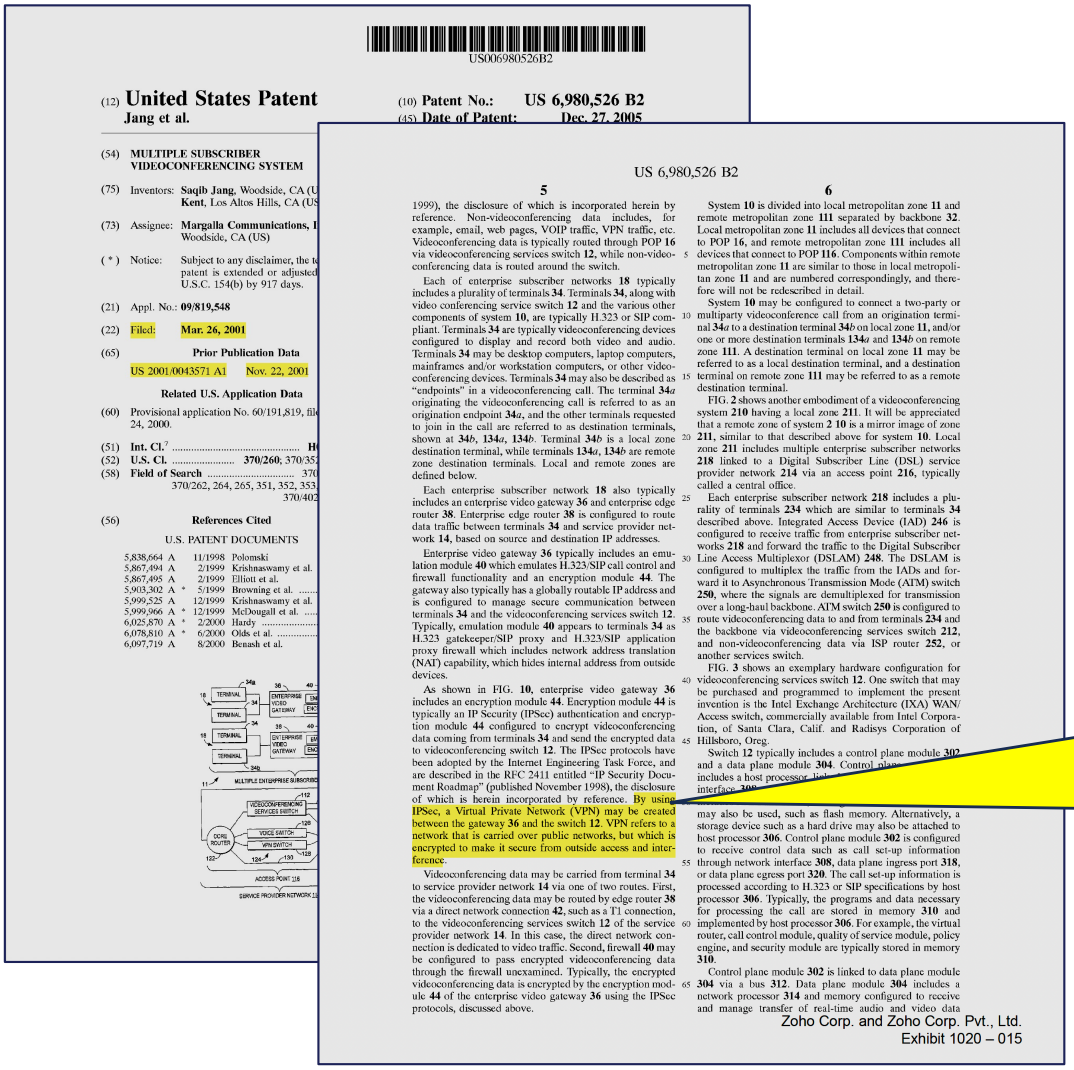
IPR2023-00380, Ex. 1001 at 10:57-59.

2) The ’525 patent, dependent claims 13-14, 19-20: (IPR2023-00379, Ground 1)

- Claims 13: “**transmitting** the mixed audio-video **data through a Virtual Private Network (VPN) tunnel**”
- Claim 14: “**transmitting** the mixed audio **data through a Virtual Private Network (VPN) tunnel**”
- Claims 19/20: “wherein the audio **data . . . is transported through a Virtual Private Network (VPN) tunnel**”

IPR2023-00379, Ex. 1001 at 11:18-20, 11:21-23, 12:28-35.

USING VPN IN CONFERENCING SYSTEMS WAS CONVENTIONAL PRACTICE



US006980526B2

(12) **United States Patent**
Jang et al.

(10) **Patent No.: US 6,980,526 B2**
(45) **Date of Patent: Dec. 27, 2005**

(54) **MULTIPLE SUBSCRIBER VIDEOCONFERENCING SYSTEM**

(75) Inventors: **Saqib Jang**, Woodside, CA (U**Kent**, Los Altos Hills, CA (US

(73) Assignee: **Margalla Communications, I**Woodside, CA (US)

(*) Notice: Subject to any disclaimer, the t patent is extended or adjusted U.S.C. 154(b) by 917 days.

(21) Appl. No.: **09/819,548**

(22) Filed: **Mar. 26, 2001**

(65) **Prior Publication Data**

US 2001/0043571 A1 Nov. 22, 2001

Related U.S. Application Data

(60) Provisional application No. 60/191,819, fil 24, 2000.

(51) **Int. Cl.**[7] ..... H
(52) **U.S. Cl.** **370/260**; 370/35
(58) **Field of Search** 370 370/262, 264, 265, 351, 352, 353 370/402

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,838,664	A		11/1998	Polomski
5,867,494	A		2/1999	Krishnaswamy et al.
5,867,495	A		2/1999	Elliott et al.
5,903,302	A	*	5/1999	Browning et al.
5,999,525	A		12/1999	Krishnaswamy et al.
5,999,966	A	*	12/1999	McDougall et al.
6,025,870	A	*	2/2000	Hardy
6,078,810	A	*	6/2000	Olds et al.
6,097,719	A		8/2000	Benash et al.

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1999), the disclosure of which is incorporated herein by reference. Non-videoconferencing data includes, for example, email, web pages, VOIP traffic, VPN traffic, etc. Videoconferencing data is typically routed through POP **16** via videoconferencing services switch **12**, while non-videoconferencing data is routed around the switch.

Each of enterprise subscriber networks **18** typically includes a plurality of terminals **34**. Terminals **34**, along with video conferencing service switch **12** and the various other components of system **10**, are typically H.323 or SIP compliant. Terminals **34** are typically videoconferencing devices configured to display and record both video and audio. Terminals **34** may be desktop computers, laptop computers, mainframes and/or workstation computers, or other videoconferencing devices. Terminals **34** may also be described as "endpoints" in a videoconferencing call. The terminal **34***a* originating the videoconferencing call is referred to as an origination endpoint **34***a*, and the other terminals requested to join in the call are referred to as destination terminals, shown at **34***b*, **134***a*, **134***b*. Terminal **34***b* is a local zone destination terminal, while terminals **134***a*, **134***b* are remote zone destination terminals. Local and remote zones are defined below.

Each enterprise subscriber network **18** also typically includes an enterprise video gateway **36** and enterprise edge router **38**. Enterprise edge router **38** is configured to route data traffic between terminals **34** and service provider network **14**, based on source and destination IP addresses.

Enterprise video gateway **36** typically includes an emulation module **40** which emulates H.323/SIP call control and firewall functionality and an encryption module **44**. The gateway also typically has a globally routable IP address and is configured to manage secure communication between terminals **34** and the videoconferencing services switch **12**. Typically, emulation module **40** appears to terminals **34** as H.323 gatekeeper/SIP proxy and H.323/SIP application proxy firewall which includes network address translation (NAT) capability, which hides internal address from outside devices.

As shown in FIG. **10**, enterprise video gateway **36** includes an encryption module **44**. Encryption module **44** is typically an IP Security (IPSec) authentication and encryption module **44** configured to encrypt videoconferencing data coming from terminals **34** and send the encrypted data to videoconferencing switch **12**. The IPSec protocols have been adopted by the Internet Engineering Task Force, and are described in the RFC 2411 entitled "IP Security Document Roadmap" (published November 1998), the disclosure of which is herein incorporated by reference. By using IPSec, a Virtual Private Network (VPN) may be created between the gateway **36** and the switch **12**. VPN refers to a network that is carried over public networks, but which is encrypted to make it secure from outside access and interference.

Videoconferencing data may be carried from terminal **34** to service provider network **14** via one of two routes. First, the videoconferencing data may be routed by edge router **38** via a direct network connection **42**, such as a T1 connection, to the videoconferencing services switch **12** of the service provider network **14**. In this case, the direct network connection is dedicated to video traffic. Second, firewall **40** may be configured to pass encrypted videoconferencing data through the firewall unexamined. Typically, the encrypted videoconferencing data is encrypted by the encryption module **44** of the enterprise video gateway **36** using the IPSec protocols, discussed above.

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System **10** is divided into local metropolitan zone **11** and remote metropolitan zone **111** separated by backbone **32**. Local metropolitan zone **11** includes all devices that connect to POP **16**, and remote metropolitan zone **111** includes all devices that connect to POP **116**. Components within remote metropolitan zone **11** are similar to those in local metropolitan zone **11** and are numbered correspondingly, and therefore will not be redescribed in detail.

System **10** may be configured to connect a two-party or multiparty videoconference call from an origination terminal **34***a* to a destination terminal **34***b* on local zone **11**, and/or one or more destination terminals **134***a* and **134***b* on remote zone **111**. A destination terminal on local zone **11** may be referred to as a local destination terminal, and a destination terminal on remote zone **111** may be referred to as a remote destination terminal.

FIG. **2** shows another embodiment of a videoconferencing system **210** having a local zone **211**. It will be appreciated that a remote zone of system **2 10** is a mirror image of zone **211**, similar to that described above for system **10**. Local zone **211** includes multiple enterprise subscriber networks **218** linked to a Digital Subscriber Line (DSL) service provider network **214** via an access point **216**, typically called a central office.

Each enterprise subscriber network **218** includes a plurality of terminals **234** which are similar to terminals **34** described above. Integrated Access Device (IAD) **246** is configured to receive traffic from enterprise subscriber networks **218** and forward the traffic to the Digital Subscriber Line Access Multiplexor (DSLAM) **248**. The DSLAM is configured to multiplex the traffic from the IADs and forward it to Asynchronous Transmission Mode (ATM) switch **250**, where the signals are demultiplexed for transmission over a long-haul backbone. ATM switch **250** is configured to route videoconferencing data to and from terminals **234** and the backbone via videoconferencing services switch **212**, and non-videoconferencing data via ISP router **252**, or another services switch.

FIG. **3** shows an exemplary hardware configuration for videoconferencing services switch **12**. One switch that may be purchased and programmed to implement the present invention is the Intel Exchange Architecture (IXA) WAN/Access switch, commercially available from Intel Corporation, of Santa Clara, Calif. and Radisys Corporation of Hillsboro, Oreg.

Switch **12** typically includes a control plane module **302** and a data plane module **304**. Control plane includes a host processor [illegible] interface **308**

may also be used, such as flash memory. Alternatively, a storage device such as a hard drive may also be attached to host processor **306**. Control plane module **302** is configured to receive control data such as call set-up information through network interface **308**, data plane ingress port **318**, or data plane egress port **320**. The call set-up information is processed according to H.323 or SIP specifications by host processor **306**. Typically, the programs and data necessary for processing the call are stored in memory **310** and implemented by host processor **306**. For example, the virtual router, call control module, quality of service module, policy engine, and security module are typically stored in memory **310**.

Control plane module **302** is linked to data plane module **304** via a bus **312**. Data plane module **304** includes a network processor **314** and memory configured to receive and manage transfer of real-time audio and video data

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Exhibit 1020 – 015

Prior art disclosed "**video conferencing system . . . for delivering secure . . . services over an IP network**"

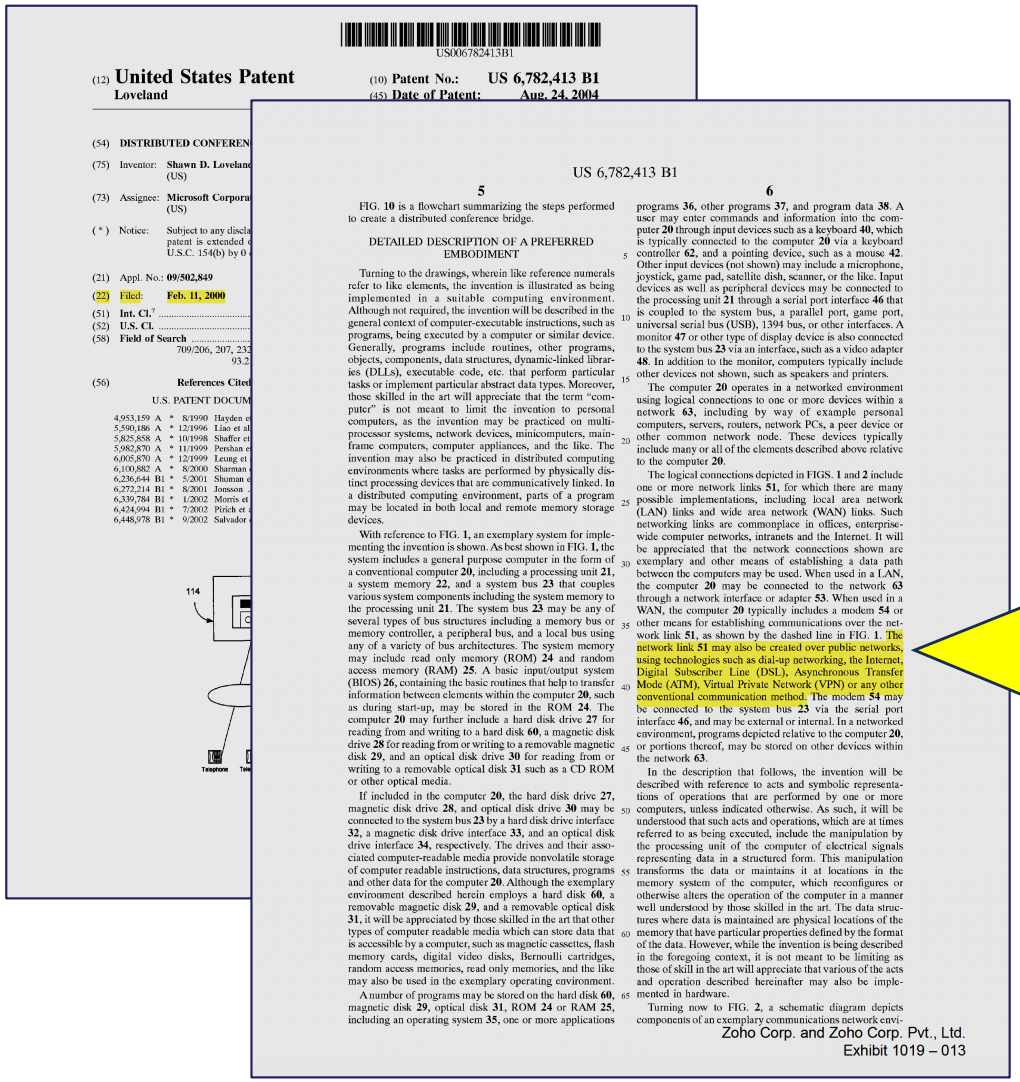
Ex. 1020 at 2:65-3:3.

By using IPSec, **a Virtual Private Network (VPN)** may be created . . . which is **encrypted to make it secure from outside access and interference**.

Ex. 1020 at 5:52-55.

See also IPR2023-00380, -382, Ex. 1003 ¶ 96; IPR2023-00377, Ex. 1003 ¶ 101; IPR2023-00378, -379, Ex. 1003 ¶ 112 (cited at IPR2023-00380, Pet. (Paper No. 1) at 9, IPR2023-00382, Pet. (Paper No. 1) at 9, IPR2023-00377, Pet. (Paper No. 1) at 11, IPR2023-00378, Pet. (Paper No. 1) at 11, IPR2023-00379, Pet. (Paper No. 1) at 10).

USING VPN IN CONFERENCING SYSTEMS WAS CONVENTIONAL PRACTICE



US006782413B1

(12) **United States Patent**
Loveland

(10) **Patent No.: US 6,782,413 B1**
(45) **Date of Patent: Aug. 24, 2004**

(54) DISTRIBUTED CONFEREN
(75) Inventor: Shawn D. Lovelanc (US)
(73) Assignee: Microsoft Corpora (US)
(*) Notice: Subject to any discla patent is extended U.S.C. 154(b) by 0
(21) Appl. No.: **09/502,849**
(22) Filed: **Feb. 11, 2000**
(51) **Int. Cl.**[7] .....
(52) **U.S. Cl.** .....
(58) **Field of Search** 709/206, 207, 232 93.2

(56) **References Cited**

U.S. PATENT DOCUM

4,953,159	A	*	8/1990	Hayden et
5,590,186	A	*	12/1996	Liao et al
5,825,858	A	*	10/1998	Shaffer et
5,982,870	A	*	11/1999	Pershan et
6,005,870	A	*	12/1999	Leung et
6,100,882	A	*	8/2000	Sharman
6,236,644	B1	*	5/2001	Shuman et
6,272,214	B1	*	8/2001	Jonsson
6,339,784	B1	*	1/2002	Morris et
6,424,994	B1	*	7/2002	Pirich et
6,448,978	B1	*	9/2002	Salvador

114

Telephone

US 6,782,413 B1

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FIG. **10** is a flowchart summarizing the steps performed to create a distributed conference bridge.

DETAILED DESCRIPTION OF A PREFERRED EMBODIMENT

Turning to the drawings, wherein like reference numerals refer to like elements, the invention is illustrated as being implemented in a suitable computing environment. Although not required, the invention will be described in the general context of computer-executable instructions, such as programs, being executed by a computer or similar device. Generally, programs include routines, other programs, objects, components, data structures, dynamic-linked libraries (DLLs), executable code, etc. that perform particular tasks or implement particular abstract data types. Moreover, those skilled in the art will appreciate that the term "computer" is not meant to limit the invention to personal computers, as the invention may be practiced on multiprocessor systems, network devices, minicomputers, mainframe computers, computer appliances, and the like. The invention may also be practiced in distributed computing environments where tasks are performed by physically distinct processing devices that are communicatively linked. In a distributed computing environment, parts of a program may be located in both local and remote memory storage devices.

With reference to FIG. **1**, an exemplary system for implementing the invention is shown. As best shown in FIG. **1**, the system includes a general purpose computer in the form of a conventional computer **20**, including a processing unit **21**, a system memory **22**, and a system bus **23** that couples various system components including the system memory to the processing unit **21**. The system bus **23** may be any of several types of bus structures including a memory bus or memory controller, a peripheral bus, and a local bus using any of a variety of bus architectures. The system memory may include read only memory (ROM) **24** and random access memory (RAM) **25**. A basic input/output system (BIOS) **26**, containing the basic routines that help to transfer information between elements within the computer **20**, such as during start-up, may be stored in the ROM **24**. The computer **20** may further include a hard disk drive **27** for reading from and writing to a hard disk **60**, a magnetic disk drive **28** for reading from or writing to a removable magnetic disk **29**, and an optical disk drive **30** for reading from or writing to a removable optical disk **31** such as a CD ROM or other optical media.

If included in the computer **20**, the hard disk drive **27**, magnetic disk drive **28**, and optical disk drive **30** may be connected to the system bus **23** by a hard disk drive interface **32**, a magnetic disk drive interface **33**, and an optical disk drive interface **34**, respectively. The drives and their associated computer-readable media provide nonvolatile storage of computer readable instructions, data structures, programs and other data for the computer **20**. Although the exemplary environment described herein employs a hard disk **60**, a removable magnetic disk **29**, and a removable optical disk **31**, it will be appreciated by those skilled in the art that other types of computer readable media which can store data that is accessible by a computer, such as magnetic cassettes, flash memory cards, digital video disks, Bernoulli cartridges, random access memories, read only memories, and the like may also be used in the exemplary operating environment.

A number of programs may be stored on the hard disk **60**, magnetic disk **29**, optical disk **31**, ROM **24** or RAM **25**, including an operating system **35**, one or more applications

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programs **36**, other programs **37**, and program data **38**. A user may enter commands and information into the computer **20** through input devices such as a keyboard **40**, which is typically connected to the computer **20** via a keyboard controller **62**, and a pointing device, such as a mouse **42**. Other input devices (not shown) may include a microphone, joystick, game pad, satellite dish, scanner, or the like. Input devices as well as peripheral devices may be connected to the processing unit **21** through a serial port interface **46** that is coupled to the system bus, a parallel port, game port, universal serial bus (USB), 1394 bus, or other interfaces. A monitor **47** or other type of display device is also connected to the system bus **23** via an interface, such as a video adapter **48**. In addition to the monitor, computers typically include other devices not shown, such as speakers and printers.

The computer **20** operates in a networked environment using logical connections to one or more devices within a network **63**, including by way of example personal computers, servers, routers, network PCs, a peer device or other common network node. These devices typically include many or all of the elements described above relative to the computer **20**.

The logical connections depicted in FIGS. **1** and **2** include one or more network links **51**, for which there are many possible implementations, including local area network (LAN) links and wide area network (WAN) links. Such networking links are commonplace in offices, enterprise-wide computer networks, intranets and the Internet. It will be appreciated that the network connections shown are exemplary and other means of establishing a data path between the computers may be used. When used in a LAN, the computer **20** may be connected to the network **63** through a network interface or adapter **53**. When used in a WAN, the computer **20** typically includes a modem **54** or other means for establishing communications over the network link **51**, as shown by the dashed line in FIG. **1**. The network link **51** may also be created over public networks, using technologies such as dial-up networking, the Internet, Digital Subscriber Line (DSL), Asynchronous Transfer Mode (ATM), Virtual Private Network (VPN) or any other conventional communication method. The modem **54** may be connected to the system bus **23** via the serial port interface **46**, and may be external or internal. In a networked environment, programs depicted relative to the computer **20**, or portions thereof, may be stored on other devices within the network **63**.

In the description that follows, the invention will be described with reference to acts and symbolic representations of operations that are performed by one or more computers, unless indicated otherwise. As such, it will be understood that such acts and operations, which are at times referred to as being executed, include the manipulation by the processing unit of the computer of electrical signals representing data in a structured form. This manipulation transforms the data or maintains it at locations in the memory system of the computer, which reconfigures or otherwise alters the operation of the computer in a manner well understood by those skilled in the art. The data structures where data is maintained are physical locations of the memory that have particular properties defined by the format of the data. However, while the invention is being described in the foregoing context, it is not meant to be limiting as those of skill in the art will appreciate that various of the acts and operation described hereinafter may also be implemented in hardware.

Turning now to FIG. **2**, a schematic diagram depicts components of an exemplary communications network envi-

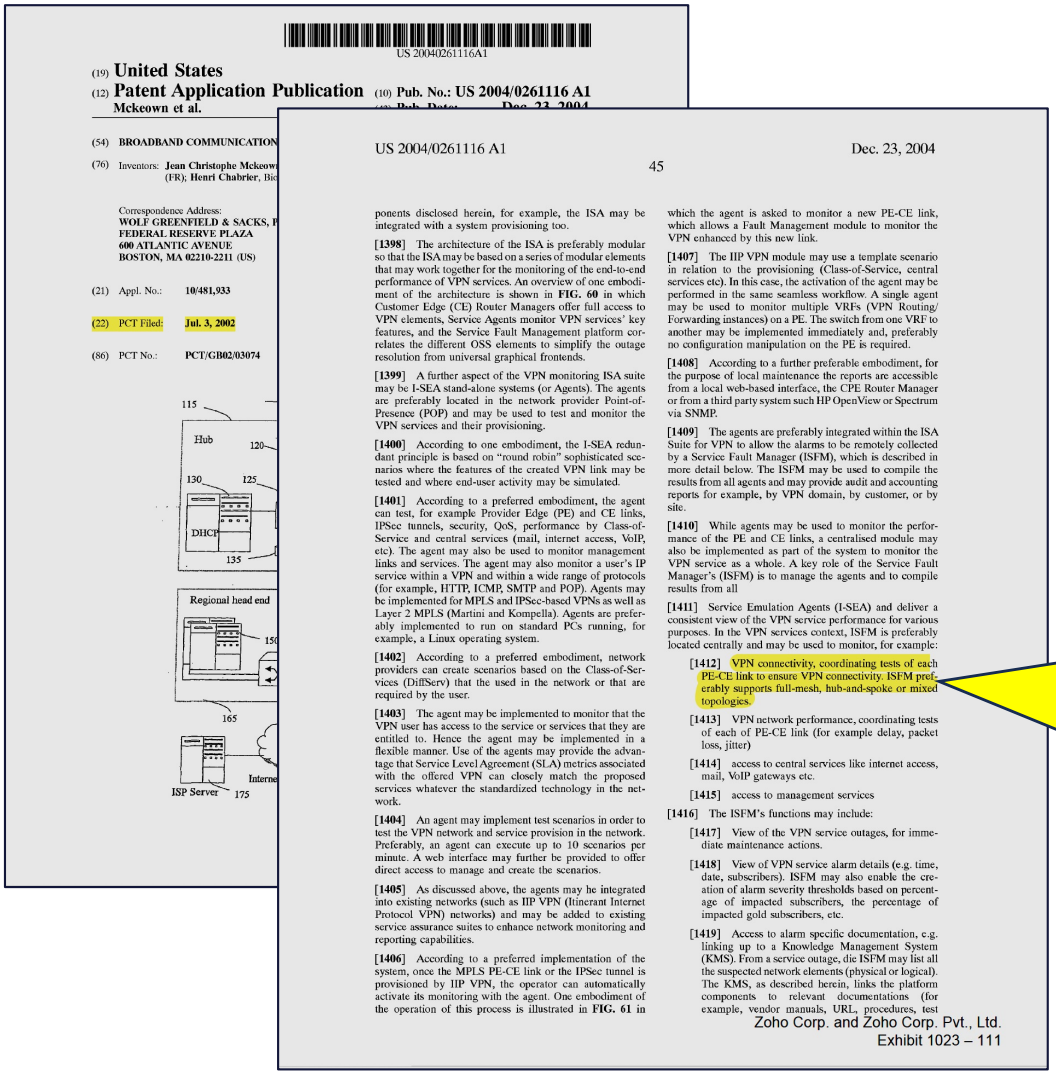
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Exhibit 1019 – 013

Conferencing bridge application with "network link 51 [that] may be created over public networks using technologies such as dial-up networking, the Internet, Digital Subscriber Line (DSL), Asynchronous Transfer Mode (ATM), **Virtual Private Network (VPN)**, **or any other conventional communication method.**

Ex. 1019 at 6:36-41.

See also IPR2023-00380, -382, Ex. 1003 ¶ 96; IPR2023-00377, Ex. 1003 ¶ 101; IPR2023-00378, -379, Ex. 1003 ¶ 112 (cited at IPR2023-00380, Pet. (Paper No. 1) at 9, IPR2023-00382, Pet. (Paper No. 1) at 9, IPR2023-00377, Pet. (Paper No. 1) at 11, IPR2023-00378, Pet. (Paper No. 1) at 11, IPR2023-00379, Pet. (Paper No. 1) at 10).

USING VPN IN CONFERENCING SYSTEMS WAS CONVENTIONAL PRACTICE



US 20040261116A1

(19) United States
(12) Patent Application Publication (10) Pub. No.: US 2004/0261116 A1
Mckeown et al.

(54) BROADBAND COMMUNICATION

(76) Inventors: Jean Christophe Mckeow (FR); Henri Chabrier, Bi

Correspondence Address:
WOLF GREENFIELD & SACKS, P
FEDERAL RESERVE PLAZA
600 ATLANTIC AVENUE
BOSTON, MA 02210-2211 (US)

(21) Appl. No.: 10/481,933

(22) PCT Filed: Jul. 3, 2002

(86) PCT No.: PCT/GB02/03074

ponents disclosed herein, for example, the ISA may be integrated with a system provisioning too.

[1398] The architecture of the ISA is preferably modular so that the ISA may be based on a series of modular elements that may work together for the monitoring of the end-to-end performance of VPN services. An overview of one embodiment of the architecture is shown in **FIG. 60** in which Customer Edge (CE) Router Managers offer full access to VPN elements, Service Agents monitor VPN services' key features, and the Service Fault Management platform correlates the different OSS elements to simplify the outage resolution from universal graphical frontends.

[1399] A further aspect of the VPN monitoring ISA suite may be I-SEA stand-alone systems (or Agents). The agents are preferably located in the network provider Point-of-Presence (POP) and may be used to test and monitor the VPN services and their provisioning.

[1400] According to one embodiment, the I-SEA redundant principle is based on "round robin" sophisticated scenarios where the features of the created VPN link may be tested and where end-user activity may be simulated.

[1401] According to a preferred embodiment, the agent can test, for example Provider Edge (PE) and CE links, IPSec tunnels, security, QoS, performance by Class-of-Service and central services (mail, internet access, VoIP, etc). The agent may also be used to monitor management links and services. The agent may also monitor a user's IP service within a VPN and within a wide range of protocols (for example, HTTP, ICMP, SMTP and POP). Agents may be implemented for MPLS and IPSec-based VPNs as well as Layer 2 MPLS (Martini and Kompella). Agents are preferably implemented to run on standard PCs running, for example, a Linux operating system.

[1402] According to a preferred embodiment, network providers can create scenarios based on the Class-of-Services (DiffServ) that the used in the network or that are required by the user.

[1403] The agent may be implemented to monitor that the VPN user has access to the service or services that they are entitled to. Hence the agent may be implemented in a flexible manner. Use of the agents may provide the advantage that Service Level Agreement (SLA) metrics associated with the offered VPN can closely match the proposed services whatever the standardized technology in the network.

[1404] An agent may implement test scenarios in order to test the VPN network and service provision in the network. Preferably, an agent can execute up to 10 scenarios per minute. A web interface may further be provided to offer direct access to manage and create the scenarios.

[1405] As discussed above, the agents may he integrated into existing networks (such as IIP VPN (Itinerant Internet Protocol VPN) networks) and may be added to existing service assurance suites to enhance network monitoring and reporting capabilities.

[1406] According to a preferred implementation of the system, once the MPLS PE-CE link or the IPSec tunnel is provisioned by IIP VPN, the operator can automatically activate its monitoring with the agent. One embodiment of the operation of this process is illustrated in **FIG. 61** in which the agent is asked to monitor a new PE-CE link, which allows a Fault Management module to monitor the VPN enhanced by this new link.

[1407] The IIP VPN module may use a template scenario in relation to the provisioning (Class-of-Service, central services etc). In this case, the activation of the agent may be performed in the same seamless workflow. A single agent may be used to monitor multiple VRFs (VPN Routing/Forwarding instances) on a PE. The switch from one VRF to another may be implemented immediately and, preferably no configuration manipulation on the PE is required.

[1408] According to a further preferable embodiment, for the purpose of local maintenance the reports are accessible from a local web-based interface, the CPE Router Manager or from a third party system such HP OpenView or Spectrum via SNMP.

[1409] The agents are preferably integrated within the ISA Suite for VPN to allow the alarms to be remotely collected by a Service Fault Manager (ISFM), which is described in more detail below. The ISFM may be used to compile the results from all agents and may provide audit and accounting reports for example, by VPN domain, by customer, or by site.

[1410] While agents may be used to monitor the performance of the PE and CE links, a centralised module may also be implemented as part of the system to monitor the VPN service as a whole. A key role of the Service Fault Manager's (ISFM) is to manage the agents and to compile results from all

[1411] Service Emulation Agents (I-SEA) and deliver a consistent view of the VPN service performance for various purposes. In the VPN services context, ISFM is preferably located centrally and may be used to monitor, for example:

[1412] VPN connectivity, coordinating tests of each PE-CE link to ensure VPN connectivity. ISFM preferably supports full-mesh, hub-and-spoke or mixed topologies.

[1413] VPN network performance, coordinating tests of each of PE-CE link (for example delay, packet loss, jitter)

[1414] access to central services like internet access, mail, VoIP gateways etc.

[1415] access to management services

[1416] The ISFM's functions may include:

[1417] View of the VPN service outages, for immediate maintenance actions.

[1418] View of VPN service alarm details (e.g. time, date, subscribers). ISFM may also enable the creation of alarm severity thresholds based on percentage of impacted subscribers, the percentage of impacted gold subscribers, etc.

[1419] Access to alarm specific documentation, e.g. linking up to a Knowledge Management System (KMS). From a service outage, die ISFM may list all the suspected network elements (physical or logical). The KMS, as described herein, links the platform components to relevant documentations (for example, vendor manuals, URL, procedures, test

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Exhibit 1023 – 111

VPN connectivity, coordinating tests of each PE-CE link to ensure VPN connectivity. ISFM preferably **supports full-mesh, hub-and-spoke or mixed topologies**.

Ex. 1023 at [1412]; *see also id.* at [1452]-[1453].

See also IPR2023-00380, -382, Ex. 1003 ¶ 96; IPR2023-00377, Ex. 1003 ¶ 101; IPR2023-00378, -379, Ex. 1003 ¶ 112 (cited at IPR2023-00380, Pet. (Paper No. 1) at 9, IPR2023-00382, Pet. (Paper No. 1) at 9, IPR2023-00377, Pet. (Paper No. 1) at 11, IPR2023-00378, Pet. (Paper No. 1) at 11, IPR2023-00379, Pet. (Paper No. 1) at 10).

IMPLEMENTING VPN TUNNELS WAS WELL-KNOWN IN THE ART

6. Virtual Private Network

91. A Virtual Private Network (VPN) is "defined loosely as a network in which customer connectivity amongst multiple sites is deployed on a shared infrastructure with the same access or security policies as a private network." *See* VPN Textbook (Ex. 1010) at 115; *see also* Yee Decl. (Ex. 1011) ¶¶ 16, 23, 95-111. VPNs were known to be used to securely exchange data between different locations over the Internet. VPN Textbook at 118-119; *see also* U.S. Patent No. 6,980,526 (the "'526 patent") (Ex. 1020) at 2:65-3:3, 5:52-55. It was also known that VPNs may be implemented over the public Internet using IPsec tunnels between sites.

IPR2023-00380, -382, Ex. 1003 ¶ 91; *see also* IPR2023-00377, Ex. 1003 ¶ 96; IPR2023-00378, -379, Ex. 1003 ¶ 107 (cited at IPR2023-00380, Pet. (Paper No. 1) at 9, IPR2023-00382, Pet. (Paper No. 1) at 9, IPR2023-00377, Pet. (Paper No. 1) at 11, IPR2023-00378, Pet. (Paper No. 1) at 11, IPR2023-00379, Pet. (Paper No. 1) at 10).

IMPLEMENTING VPN TUNNELS WAS WELL-KNOWN IN THE ART

3. Tunneling

83. Tunneling refers to a networking technique whereby a packet is encapsulated in another network packet, typically so that it can traverse an underlying network that does not support the capability required by the encapsulated packet. IP in IP tunneling is described in RFC 1853 which was published in the mid-1990s. Tunneling can be performed in both directions. At the input end of the tunnel, packets are encapsulated, or wrapped, in a way where the source address of the encapsulating header is that of the encapsulating device and the destination address of the encapsulating header is the far end of the tunnel. At the far end of the tunnel, the packets are stripped of the encapsulating header, or unwrapped, and the original packet emerges for further actions such as routing to its destination using the original packet's header.

IPR2023-00380, -382, Ex. 1003 ¶ 83; *see also* IPR2023-00377, Ex. 1003 ¶ 88; IPR2023-00378, -379, Ex. 1003 ¶ 99 (cited at IPR2023-00380, Pet. (Paper No. 1) at 9, IPR2023-00382, Pet. (Paper No. 1) at 9, IPR2023-00377, Pet. (Paper No. 1) at 11, IPR2023-00378, Pet. (Paper No. 1) at 11, IPR2023-00379, Pet. (Paper No. 1) at 10).

IMPLEMENTING VPN TUNNELS WAS WELL-KNOWN IN THE ART

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IPR2023-00380, -382, Ex. 1003 ¶ 83; *see also* IPR2023-00377, Ex. 1003 ¶ 88; IPR2023-00378, -379, Ex. 1003 ¶ 99; *see also* IPR2023-00380, -382, Ex. 1003 ¶¶ 84-90; *see also* IPR2023-00377, Ex. 1003 ¶¶ 89-95; IPR2023-00378, -379, Ex. 1003 ¶¶ 100-106 (discussing implementation of tunneling in common network security approaches in the 1990s) (cited at IPR2023-00380, Pet. (Paper No. 1) at 9, IPR2023-00382, Pet. (Paper No. 1) at 9, IPR2023-00377, Pet. (Paper No. 1) at 11, IPR2023-00378, Pet. (Paper No. 1) at 11, IPR2023-00379, Pet. (Paper No. 1) at 10).

THE COMBINATION OF KNAPPE/DRELL WITH ELLIOTT AND THE VPN TEXTBOOK RENDERS OBVIOUS THE "FIRST" AND "SECOND VPN TUNNEL" LIMITATIONS

1) **The '612 patent, independent claims 19 and 25: (IPR2023-00382, Grounds 1 and 2)**

- "receiving first . . . data . . . through a **first Virtual Private Network (VPN) tunnel**"
- "receiving second . . . data . . . through a **second Virtual Private Network (VPN) tunnel**"

IPR2023-00382, Ex. 1001 at 11:25-29, 12:16-21.

2) **The '332 patent, independent claims 1 and 8, dependent claim 5: (IPR2023-00377, Grounds 1-3)**

- Claims 1/8: "receiving first . . . data . . . through **a first Virtual Private Network (VPN) tunnel**"
- Claim 5: "wherein the data . . . is transported through **a second VPN Tunnel**"

IPR2023-00377, Ex. 1001 at 9:57-59, 10:20-22, 10:36-39.

3) **The '525 patent, dependent claims 5 and 7: (IPR2023-00378, Grounds 2, 4)**

- "wherein the . . . data stream . . . are transported across a **Virtual Private Network (VPN) tunnel**"

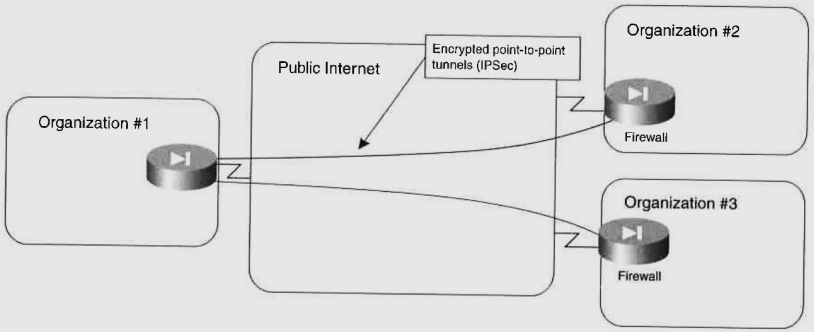
IPR2023-00378, Ex. 1001 at 10:30-32, 10:37-39.

VPN TEXTBOOK

These are the two major reasons why we don't see many organizations using Internet, which cannot offer end-to-end quality of service, isolation, or security, as the infrastructure for their intra-organizational communications. Intranet VPNs were thus usually implemented with traditional technologies like X.25, Frame Relay, or ATM.

Inter-organizational communications frequently take place between central sites of the organizations—usually using dedicated security devices, such as firewalls or encryption gear similar to the setup demonstrated in Figure 7-3. These communications also might have less stringent quality of service requirements. This set of requirements makes the Internet more and more suitable for inter-organizational communications; therefore, it's no surprise that more and more business-to-business traffic takes place over the Internet.

Figure 7-3 *Typical Extranet Setup*



Remote user access into a corporate network, typically from changing or unknown locations, is always riddled with security issues, which [illegible] end basis using such technologies as encryption or one-time passwords. Thus, the security requirements for VPDN services were never as high as the requirements for Intranet communications. It's no surprise that most of the VPDN services today are implemented on top of Internet Protocol (IP), either over the Internet or using the private backbone of a service provider, as illustrated in Figure 7-4. The protocols used to implement VPDN service over IP include Layer 2 Forwarding (L2F) or Layer 2 Transport Protocol (L2TP).

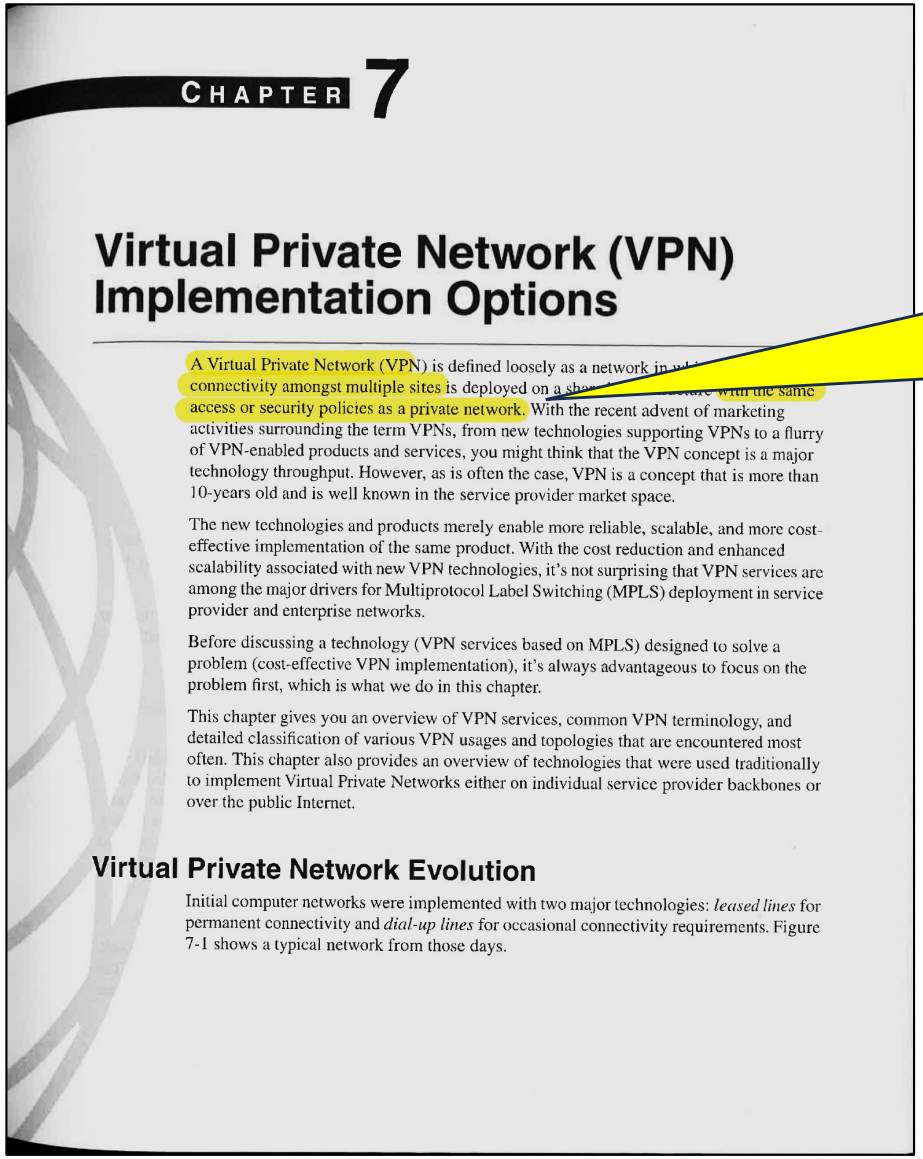
This set of requirements makes the Internet more and more suitable for inter-organizational communications; therefore, it's no surprise that **more and more business-to-business traffic takes place over the Internet**.

Remote user access into a corporate **network**, typically from changing or unknown locations, **is always riddled with security issues**.

Ex. 1010 at p. 119; *see also* Ex. 1003 ¶ 146.

See IPR2023-00377, Pet. (Paper No. 1) at 20;
IPR2023-00382, Pet. (Paper No. 1) at 15-16;
IPR2023-00378, Pet. (Paper No. 1) at 54;
IPR2023-00371, Pet. (Paper No. 1) at 20-21.

VPN TEXTBOOK



CHAPTER 7

Virtual Private Network (VPN) Implementation Options

A Virtual Private Network (VPN) is defined loosely as a network in [illegible] connectivity amongst multiple sites is deployed on a [illegible] access or security policies as a private network. With the recent advent of marketing activities surrounding the term VPNs, from new technologies supporting VPNs to a flurry of VPN-enabled products and services, you might think that the VPN concept is a major technology throughput. However, as is often the case, VPN is a concept that is more than 10-years old and is well known in the service provider market space.

The new technologies and products merely enable more reliable, scalable, and more cost-effective implementation of the same product. With the cost reduction and enhanced scalability associated with new VPN technologies, it's not surprising that VPN services are among the major drivers for Multiprotocol Label Switching (MPLS) deployment in service provider and enterprise networks.

Before discussing a technology (VPN services based on MPLS) designed to solve a problem (cost-effective VPN implementation), it's always advantageous to focus on the problem first, which is what we do in this chapter.

This chapter gives you an overview of VPN services, common VPN terminology, and detailed classification of various VPN usages and topologies that are encountered most often. This chapter also provides an overview of technologies that were used traditionally to implement Virtual Private Networks either on individual service provider backbones or over the public Internet.

Virtual Private Network Evolution

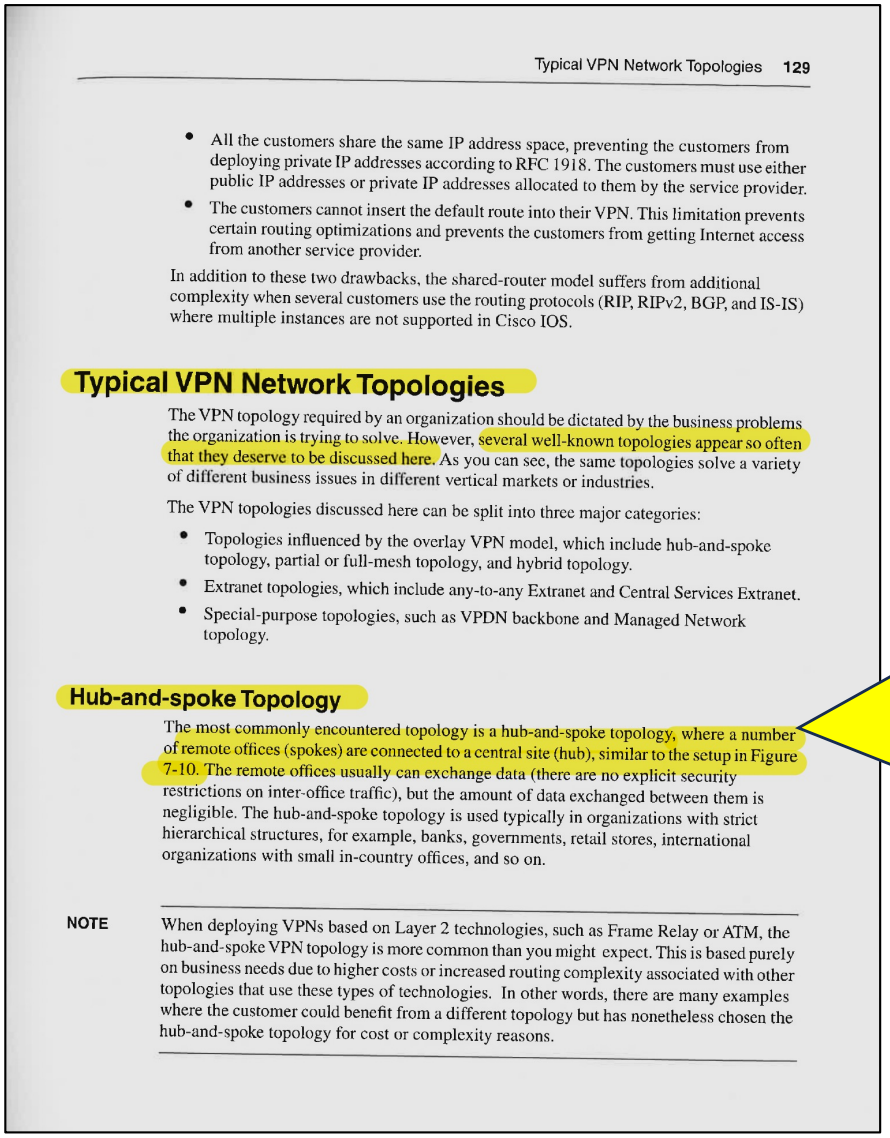
Initial computer networks were implemented with two major technologies: *leased lines* for permanent connectivity and *dial-up lines* for occasional connectivity requirements. Figure 7-1 shows a typical network from those days.

A Virtual Private Network (VPN) . . . [provides] connectivity amongst multiple sites . . . with the same access or security policies as a private network.

Ex. 1010 at p. 119; *see also* Ex. 1003 ¶ 147.

See IPR2023-00377, Pet. (Paper No. 1) at 20;
IPR2023-00382, Pet. (Paper No. 1) at 15-16;
IPR2023-00378, Pet. (Paper No. 1) at 54;
IPR2023-00371, Pet. (Paper No. 1) at 20-21.

VPN TEXTBOOK



- All the customers share the same IP address space, preventing the customers from deploying private IP addresses according to RFC 1918. The customers must use either public IP addresses or private IP addresses allocated to them by the service provider.
- The customers cannot insert the default route into their VPN. This limitation prevents certain routing optimizations and prevents the customers from getting Internet access from another service provider.

In addition to these two drawbacks, the shared-router model suffers from additional complexity when several customers use the routing protocols (RIP, RIPv2, BGP, and IS-IS) where multiple instances are not supported in Cisco IOS.

Typical VPN Network Topologies

The VPN topology required by an organization should be dictated by the business problems the organization is trying to solve. However, several well-known topologies appear so often that they deserve to be discussed here. As you can see, the same topologies solve a variety of different business issues in different vertical markets or industries.

The VPN topologies discussed here can be split into three major categories:

- Topologies influenced by the overlay VPN model, which include hub-and-spoke topology, partial or full-mesh topology, and hybrid topology.
- Extranet topologies, which include any-to-any Extranet and Central Services Extranet.
- Special-purpose topologies, such as VPDN backbone and Managed Network topology.

Hub-and-spoke Topology

The most commonly encountered topology is a hub-and-spoke topology, where a number of remote offices (spokes) are connected to a central site (hub), similar to the setup in Figure 7-10. The remote offices usually can exchange data (there are no explicit security restrictions on inter-office traffic), but the amount of data exchanged between them is negligible. The hub-and-spoke topology is used typically in organizations with strict hierarchical structures, for example, banks, governments, retail stores, international organizations with small in-country offices, and so on.

NOTE When deploying VPNs based on Layer 2 technologies, such as Frame Relay or ATM, the hub-and-spoke VPN topology is more common than you might expect. This is based purely on business needs due to higher costs or increased routing complexity associated with other topologies that use these types of technologies. In other words, there are many examples where the customer could benefit from a different topology but has nonetheless chosen the hub-and-spoke topology for cost or complexity reasons.

Typical VPN Network Topologies

The VPN Topology required by an organization should be dictated by the business problem that organization is trying to solve. However, **several well-known topologies appear so often** they deserve to be discussed here.

…

The most commonly encountered topology is a hub-and spoke topology, where a number of remote offices (spokes) are connected to a central site (hub), similar to the setup in Figure 7-10.

Ex. 1010 at p. 129.

See IPR2023-00377, Pet. (Paper No. 1) at 20;
IPR2023-00382, Pet. (Paper No. 1) at 15-16;
IPR2023-00378, Pet. (Paper No. 1) at 54;
IPR2023-00371, Pet. (Paper No. 1) at 20-21.

VPN TEXTBOOK

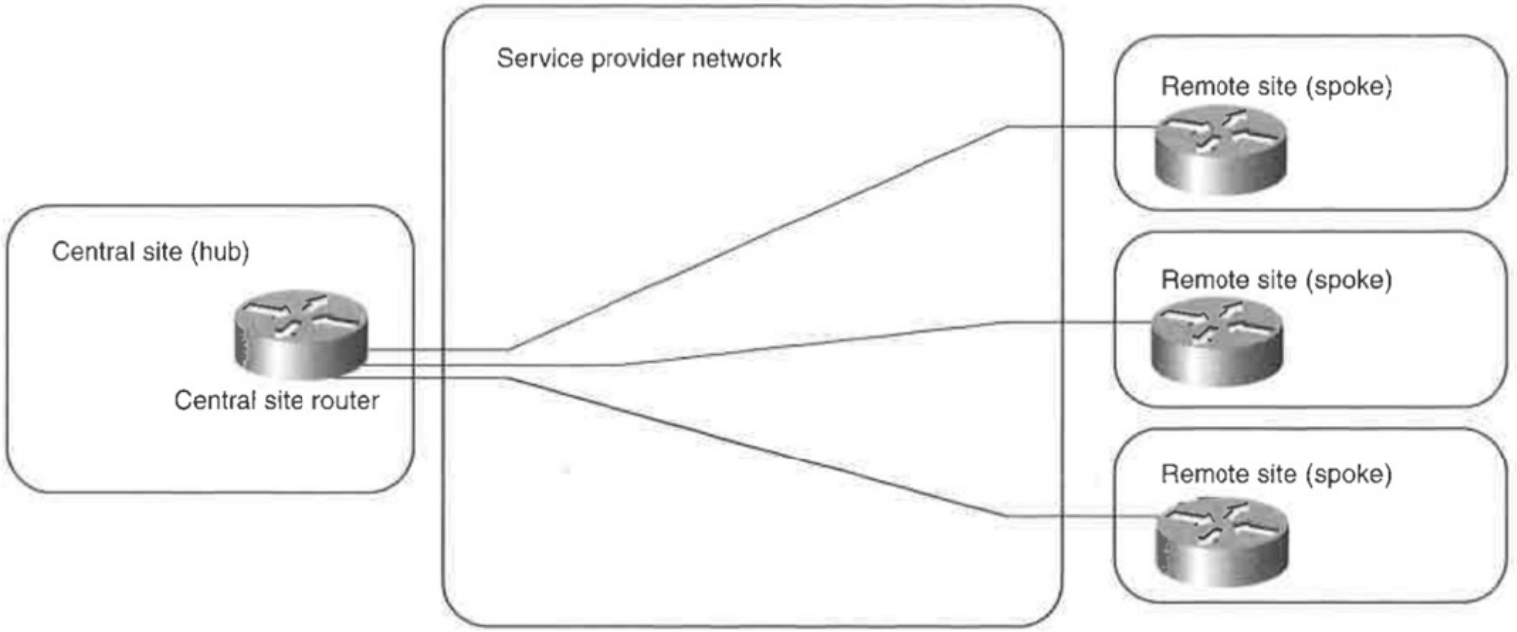
D. VPN Textbook (Ex. 1010)

The VPN textbook describes VPN topologies, the most common being the hub-and-spoke topology, also referred to as a "star" topology. Pet. 20–21 (citing Ex. 1010, 129–130; Ex. 1003, Houh Decl. ¶ 112).

IPR2023-00371, Institution Decision (Paper 6) at 14; *see also* IPR2023-00377, Institution Decision at 23; IPR2023-00382, Institution Decision at 15-16.

VPN TEXTBOOK

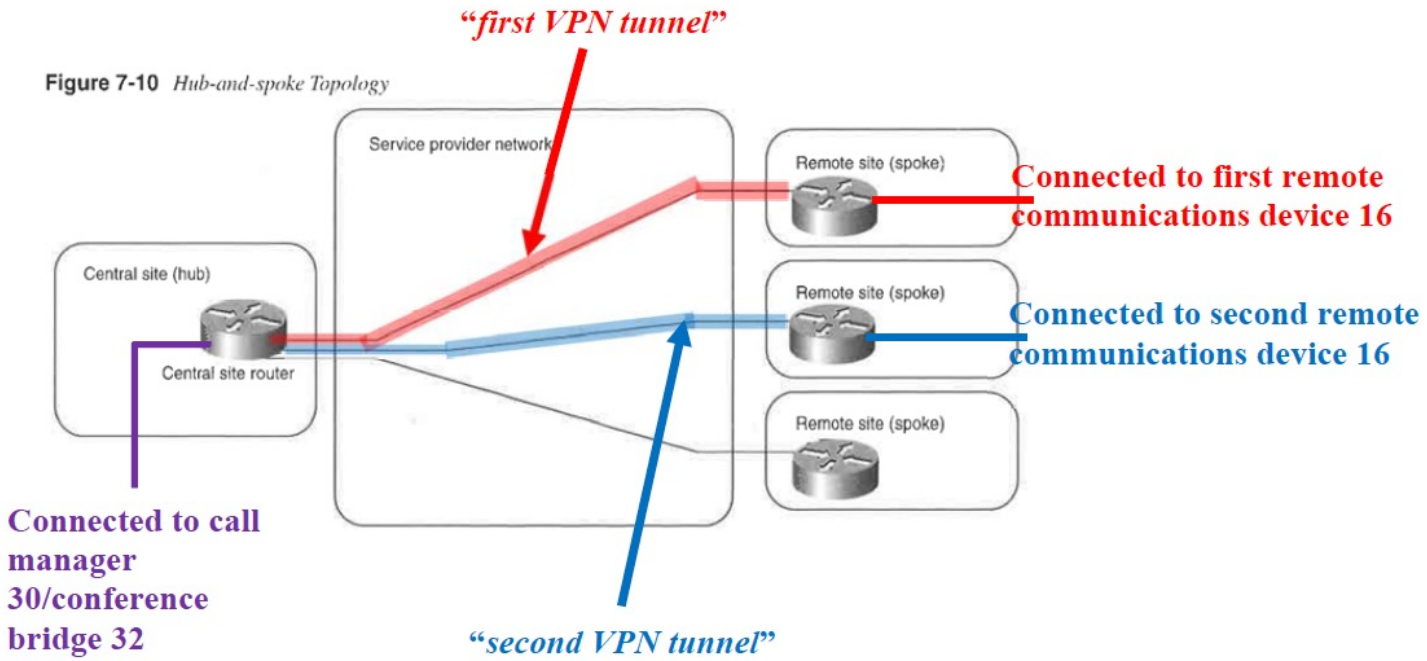
Figure 7-10 *Hub-and-spoke Topology*



Ex. 1010 at FIG. 7-10 (p. 130).

See IPR2023-00377, Pet. (Paper No. 1) at 20;
IPR2023-00382, Pet. (Paper No. 1) at 15-16;
IPR2023-00378, Pet. (Paper No. 1) at 54;
IPR2023-00371, Pet. (Paper No. 1) at 20-21.

VPN TEXTBOOK



Ex. 1010 at FIG. 7-10 (annotated); IPR2023-00382, Ex. 1003 ¶ 431 (cited at Pet. (Paper No. 1) at 43); *see also* IPR2023-00377, Ex. 1003 ¶ 234 (cited at Pet. (Paper No. 1) at 48).

The Prior Art Renders Obvious the Encoding Data into Compressed Data Limitations

IPR2023-0377, Ground 2
(U.S. Patent No. 9,253,332)

THE "COMPRESSED" DATA LIMITATIONS

Ground	Claims	Statutory Basis	Reference(s)
1	1-2, 4-5 and 7	§ 103	Knappe, Elliott, and VPN Textbook
2	3, 6, and 8-11	§ 103	Knappe, Elliott, VPN Textbook, and Drell
3	12	§ 103	Knappe, Elliott, VPN Textbook, Drell, and Hoke

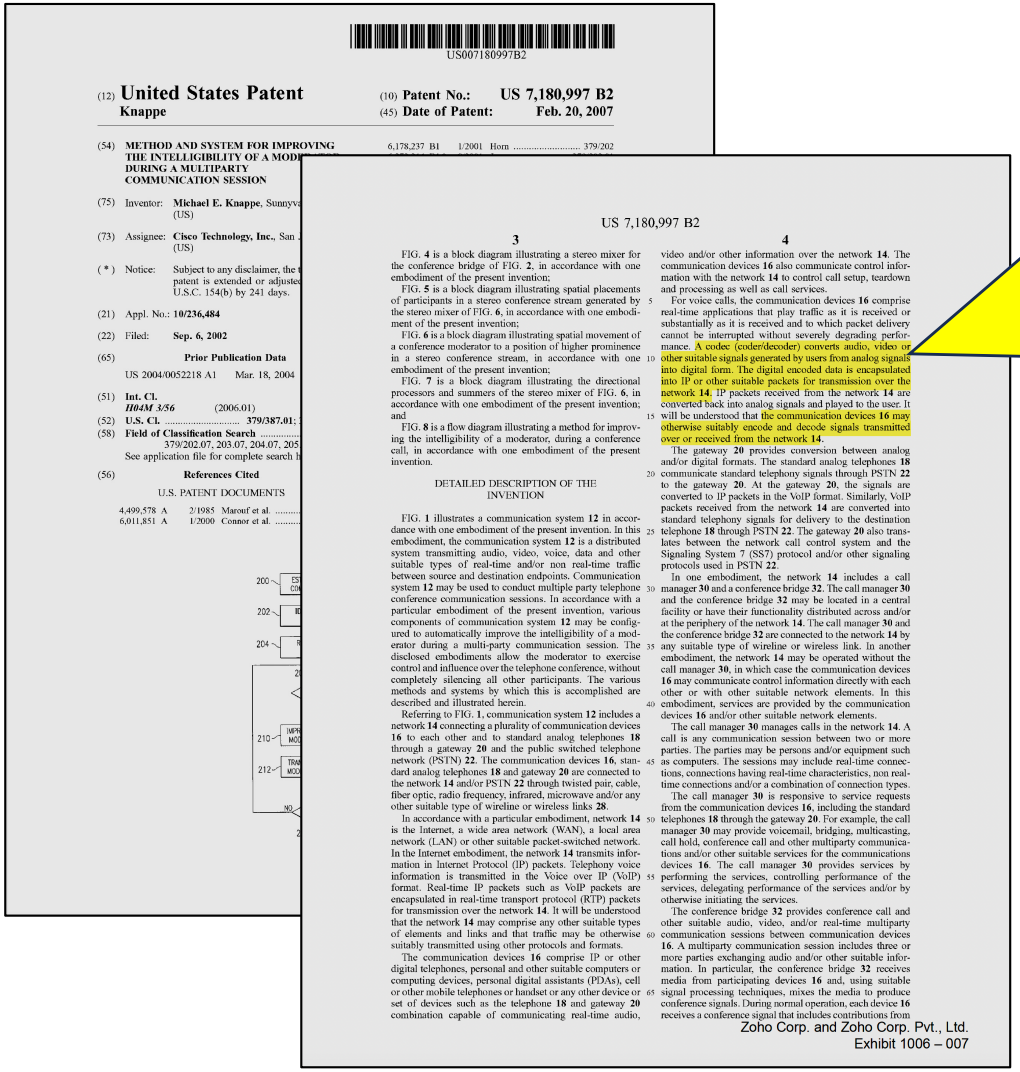
1) The '332 patent, dependent claim 6: (IPR2023-00377, Ground 2)

- **"encoding the first mixed audio data into a first <u>compressed audio data</u>** before transmitting the first mixed audio data to the remote client."

IPR2023-00377, Ex. 1001 at 10:23-26.

A POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE WITH DRELL

KNAPPE



US007180997B2

(12) **United States Patent**
Knappe

(10) **Patent No.: US 7,180,997 B2**
(45) **Date of Patent: Feb. 20, 2007**

(54) **METHOD AND SYSTEM FOR IMPROVING THE INTELLIGIBILITY OF A MODERATOR DURING A MULTIPARTY COMMUNICATION SESSION**

(75) Inventor: **Michael E. Knappe**, Sunnyvale (US)

(73) Assignee: **Cisco Technology, Inc.**, San Jose (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 241 days.

(21) Appl. No.: **10/236,484**

(22) Filed: **Sep. 6, 2002**

(65) **Prior Publication Data**

US 2004/0052218 A1 Mar. 18, 2004

(51) **Int. Cl.**
H04M 3/56 (2006.01)

(52) **U.S. Cl.** **379/387.01**;

(58) **Field of Classification Search** .....
379/202.07, 203.07, 204.07, 205
See application file for complete search h

(56) **References Cited**

U.S. PATENT DOCUMENTS

4,499,578 A 2/1985 Maronf et al. .....
6,011,851 A 1/2000 Connor et al. .....

6,178,237 B1 1/2001 Horn 379/202

US 7,180,997 B2

3

FIG. **4** is a block diagram illustrating a stereo mixer for the conference bridge of FIG. **2**, in accordance with one embodiment of the present invention;

FIG. **5** is a block diagram illustrating spatial placements of participants in a stereo conference stream generated by the stereo mixer of FIG. **6**, in accordance with one embodiment of the present invention;

FIG. **6** is a block diagram illustrating spatial movement of a conference moderator to a position of higher prominence in a stereo conference stream, in accordance with one embodiment of the present invention;

FIG. **7** is a block diagram illustrating the directional processors and summers of the stereo mixer of FIG. **6**, in accordance with one embodiment of the present invention; and

FIG. **8** is a flow diagram illustrating a method for improving the intelligibility of a moderator, during a conference call, in accordance with one embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

FIG. **1** illustrates a communication system **12** in accordance with one embodiment of the present invention. In this embodiment, the communication system **12** is a distributed system transmitting audio, video, voice, data and other suitable types of real-time and/or non real-time traffic between source and destination endpoints. Communication system **12** may be used to conduct multiple party telephone conference communication sessions. In accordance with a particular embodiment of the present invention, various components of communication system **12** may be configured to automatically improve the intelligibility of a moderator during a multi-party communication session. The disclosed embodiments allow the moderator to exercise control and influence over the telephone conference, without completely silencing all other participants. The various methods and systems by which this is accomplished are described and illustrated herein.

Referring to FIG. **1**, communication system **12** includes a network **14** connecting a plurality of communication devices **16** to each other and to standard analog telephones **18** through a gateway **20** and the public switched telephone network (PSTN) **22**. The communication devices **16**, standard analog telephones **18** and gateway **20** are connected to the network **14** and/or PSTN **22** through twisted pair, cable, fiber optic, radio frequency, infrared, microwave and/or any other suitable type of wireline or wireless links **28**.

In accordance with a particular embodiment, network **14** is the Internet, a wide area network (WAN), a local area network (LAN) or other suitable packet-switched network. In the Internet embodiment, the network **14** transmits information in Internet Protocol (IP) packets. Telephony voice information is transmitted in the Voice over IP (VoIP) format. Real-time IP packets such as VoIP packets are encapsulated in real-time transport protocol (RTP) packets for transmission over the network **14**. It will be understood that the network **14** may comprise any other suitable types of elements and links and that traffic may be otherwise suitably transmitted using other protocols and formats.

The communication devices **16** comprise IP or other digital telephones, personal and other suitable computers or computing devices, personal digital assistants (PDAs), cell or other mobile telephones or handset or any other device or set of devices such as the telephone **18** and gateway **20** combination capable of communicating real-time audio,

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video and/or other information over the network **14**. The communication devices **16** also communicate control information with the network **14** to control call setup, teardown and processing as well as call services.

For voice calls, the communication devices **16** comprise real-time applications that play traffic as it is received or substantially as it is received and to which packet delivery cannot be interrupted without severely degrading performance. A codec (coder/decoder) converts audio, video or other suitable signals generated by users from analog signals into digital form. The digital encoded data is encapsulated into IP or other suitable packets for transmission over the network **14**. IP packets received from the network **14** are converted back into analog signals and played to the user. It will be understood that the communication devices **16** may otherwise suitably encode and decode signals transmitted over or received from the network **14**.

The gateway **20** provides conversion between analog and/or digital formats. The standard analog telephones **18** communicate standard telephony signals through PSTN **22** to the gateway **20**. At the gateway **20**, the signals are converted to IP packets in the VoIP format. Similarly, VoIP packets received from the network **14** are converted into standard telephony signals for delivery to the destination telephone **18** through PSTN **22**. The gateway **20** also translates between the network call control system and the Signaling System 7 (SS7) protocol and/or other signaling protocols used in PSTN **22**.

In one embodiment, the network **14** includes a call manager **30** and a conference bridge **32**. The call manager **30** and the conference bridge **32** may be located in a central facility or have their functionality distributed across and/or at the periphery of the network **14**. The call manager **30** and the conference bridge **32** are connected to the network **14** by any suitable type of wireline or wireless link. In another embodiment, the network **14** may be operated without the call manager **30**, in which case the communication devices **16** may communicate control information directly with each other or with other suitable network elements. In this embodiment, services are provided by the communication devices **16** and/or other suitable network elements.

The call manager **30** manages calls in the network **14**. A call is any communication session between two or more parties. The parties may be persons and/or equipment such as computers. The sessions may include real-time connections, connections having real-time characteristics, non real-time connections and/or a combination of connection types.

The call manager **30** is responsive to service requests from the communication devices **16**, including the standard telephones **18** through the gateway **20**. For example, the call manager **30** may provide voicemail, bridging, multicasting, call hold, conference call and other multiparty communications and/or other suitable services for the communications devices **16**. The call manager **30** provides services by performing the services, controlling performance of the services, delegating performance of the services and/or by otherwise initiating the services.

The conference bridge **32** provides conference call and other suitable audio, video, and/or real-time multiparty communication sessions between communication devices **16**. A multiparty communication session includes three or more parties exchanging audio and/or other suitable information. In particular, the conference bridge **32** receives media from participating devices **16** and, using suitable signal processing techniques, mixes the media to produce conference signals. During normal operation, each device **16** receives a conference signal that includes contributions from

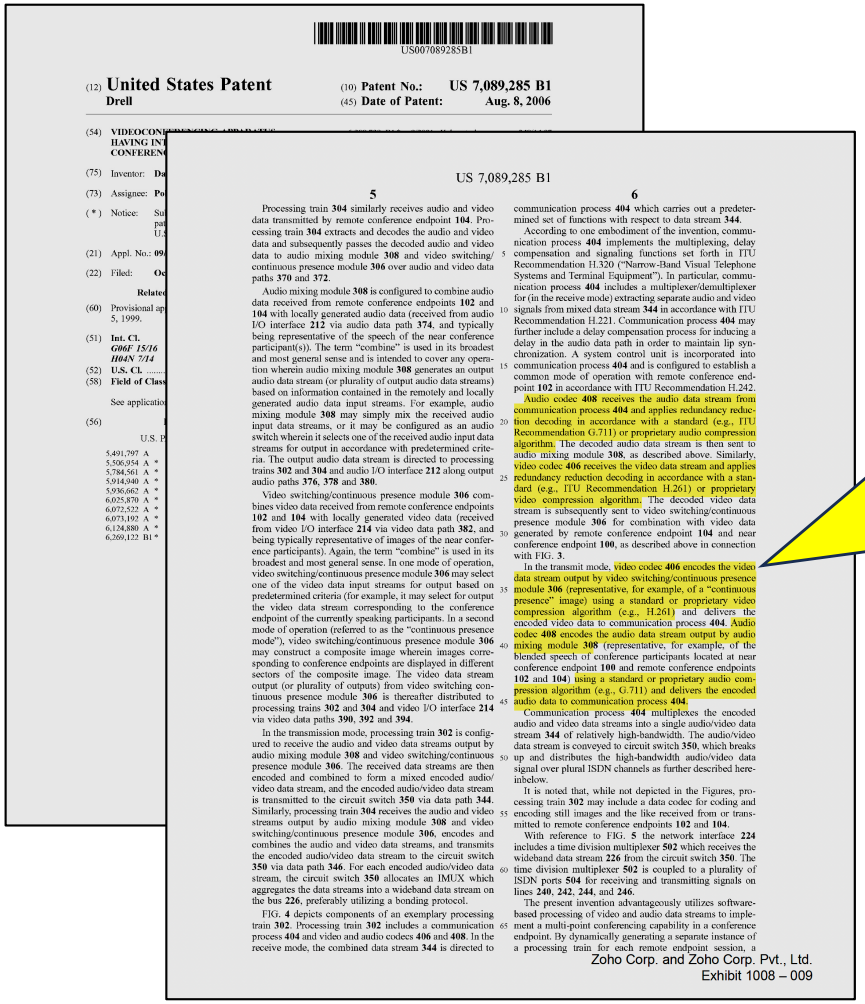
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Exhibit 1006 – 007

A **codec (coder/decoder) converts audio, video** or other suitable **signals** generated by users from analog signals to digital form. **The digital encoded data** is encapsulated into IP or other suitable packets from transmission over the network 14. . . . the **communication devices 16 may otherwise suitably encode and decode signals** transmitted from the network 14.

Ex. 1006 at 4:9-17; *see also* IPR2023-0377, Ex. 1003 ¶¶ 258 -261.

A POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE WITH DRELL

DRELL



US007089285B1

(12) **United States Patent**
Drell

(10) **Patent No.: US 7,089,285 B1**
(45) **Date of Patent: Aug. 8, 2006**

US 7,089,285 B1

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Processing train **304** similarly receives audio and video data transmitted by remote conference endpoint **104**. Processing train **304** extracts and decodes the audio and video data and subsequently passes the decoded audio and video data to audio mixing module **308** and video switching/continuous presence module **306** over audio and video data paths **370** and **372**.

Audio mixing module **308** is configured to combine audio data received from remote conference endpoints **102** and **104** with locally generated audio data (received from audio I/O interface **212** via audio data path **374**, and typically being representative of the speech of the near conference participant(s)). The term "combine" is used in its broadest and most general sense and is intended to cover any operation wherein audio mixing module **308** generates an output audio data stream (or plurality of output audio data streams) based on information contained in the remotely and locally generated audio data input streams. For example, audio mixing module **308** may simply mix the received audio input data streams, or it may be configured as an audio switch wherein it selects one of the received audio input data streams for output in accordance with predetermined criteria. The output audio data stream is directed to processing trains **302** and **304** and audio I/O interface **212** along output audio paths **376**, **378** and **380**.

Video switching/continuous presence module **306** combines video data received from remote conference endpoints **102** and **104** with locally generated video data (received from video I/O interface **214** via video data path **382**, and being typically representative of images of the near conference participants). Again, the term "combine" is used in its broadest and most general sense. In one mode of operation, video switching/continuous presence module **306** may select one of the video data input streams for output based on predetermined criteria (for example, it may select for output the video data stream corresponding to the conference endpoint of the currently speaking participants. In a second mode of operation (referred to as the "continuous presence mode"), video switching/continuous presence module **306** may construct a composite image wherein images corresponding to conference endpoints are displayed in different sectors of the composite image. The video data stream output (or plurality of outputs) from video switching continuous presence module **306** is thereafter distributed to processing trains **302** and **304** and video I/O interface **214** via video data paths **390**, **392** and **394**.

In the transmission mode, processing train **302** is configured to receive the audio and video data streams output by audio mixing module **308** and video switching/continuous presence module **306**. The received data streams are then encoded and combined to form a mixed encoded audio/video data stream, and the encoded audio/video data stream is transmitted to the circuit switch **350** via data path **344**. Similarly, processing train **304** receives the audio and video streams output by audio mixing module **308** and video switching/continuous presence module **306**, encodes and combines the audio and video data streams, and transmits the encoded audio/video data stream to the circuit switch **350** via data path **346**. For each encoded audio/video data stream, the circuit switch **350** allocates an IMUX which aggregates the data streams into a wideband data stream on the bus **226**, preferably utilizing a bonding protocol.

FIG. **4** depicts components of an exemplary processing train **302**. Processing train **302** includes a communication process **404** and video and audio codecs **406** and **408**. In the receive mode, the combined data stream **344** is directed to communication process **404** which carries out a predetermined set of functions with respect to data stream **344**.

According to one embodiment of the invention, communication process **404** implements the multiplexing, delay compensation and signaling functions set forth in ITU Recommendation H.320 ("Narrow-Band Visual Telephone Systems and Terminal Equipment"). In particular, communication process **404** includes a multiplexer/demultiplexer for (in the receive mode) extracting separate audio and video signals from mixed data stream **344** in accordance with ITU Recommendation H.221. Communication process **404** may further include a delay compensation process for inducing a delay in the audio data path in order to maintain lip synchronization. A system control unit is incorporated into communication process **404** and is configured to establish a common mode of operation with remote conference endpoint **102** in accordance with ITU Recommendation H.242.

Audio codec **408** receives the audio data stream from communication process **404** and applies redundancy reduction decoding in accordance with a standard (e.g., ITU Recommendation G.711) or proprietary audio compression algorithm. The decoded audio data stream is then sent to audio mixing module **308**, as described above. Similarly, video codec **406** receives the video data stream and applies redundancy reduction decoding in accordance with a standard (e.g., ITU Recommendation H.261) or proprietary video compression algorithm. The decoded video data stream is subsequently sent to video switching/continuous presence module **306** for combination with video data generated by remote conference endpoint **104** and near conference endpoint **100**, as described above in connection with FIG. **3**.

In the transmit mode, video codec **406** encodes the video data stream output by video switching/continuous presence module **306** (representative, for example, of a "continuous presence" image) using a standard or proprietary video compression algorithm (e.g., H.261) and delivers the encoded video data to communication process **404**. Audio codec **408** encodes the audio data stream output by audio mixing module **308** (representative, for example, of the blended speech of conference participants located at near conference endpoint **100** and remote conference endpoints **102** and **104**) using a standard or proprietary audio compression algorithm (e.g., G.711) and delivers the encoded audio data to communication process **404**.

Communication process **404** multiplexes the encoded audio and video data streams into a single audio/video data stream **344** of relatively high-bandwidth. The audio/video data stream is conveyed to circuit switch **350**, which breaks up and distributes the high-bandwidth audio/video data signal over plural ISDN channels as further described hereinbelow.

It is noted that, while not depicted in the Figures, processing train **302** may include a data codec for coding and encoding still images and the like received from or transmitted to remote conference endpoints **102** and **104**.

With reference to FIG. **5** the network interface **224** includes a time division multiplexer **502** which receives the wideband data stream **226** from the circuit switch **350**. The time division multiplexer **502** is coupled to a plurality of ISDN ports **504** for receiving and transmitting signals on lines **240**, **242**, **244**, and **246**.

The present invention advantageously utilizes software-based processing of video and audio data streams to implement a multi-point conferencing capability in a conference endpoint. By dynamically generating a separate instance of a processing train for each remote endpoint session, a

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video codec 406 encodes the video data stream output by video switching/continuous presence module 306 . . . **using a standard or proprietary video compression algorithm** (e.g. H.261) . . . .

Audio codec 408 encodes the audio data stream output by audio mixing module 308 . . . **using a standard or proprietary audio compression algorithm** (e.g. G.711)[.]

Ex. 1008 at 6:18-45; *see also* IPR2023-0377, Ex. 1003 ¶¶ 258 -261.

The Prior Art Renders Obvious the Transmitting Collaboration Data Limitations

IPR2023-0382, Ground 1
(U.S. Patent No. 9,843,612)

THE COLLABORATION DATA LIMITATIONS

Ground	Claims	Statutory Basis	Reference(s)
1	19-21 and 25-27	§ 103	Knappe, Elliott, VPN Textbook, and Hendricks
2	22-24 and 28-29	§ 103	Knappe, Elliott, VPN Textbook, Hendricks, and Drell

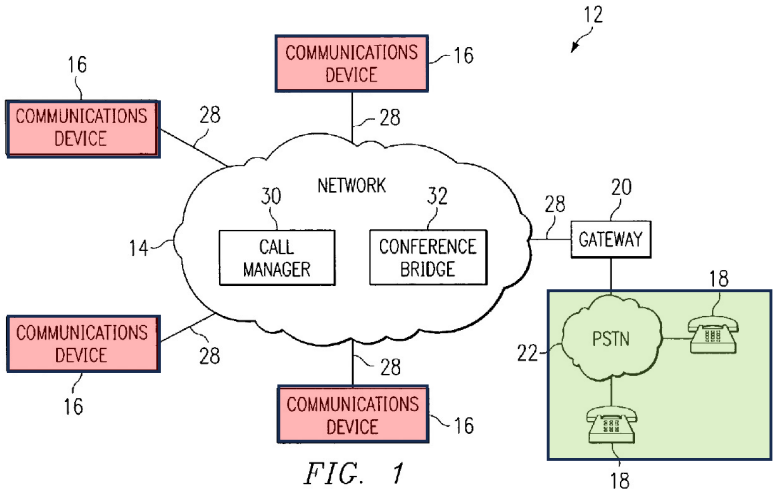
1) The '612 patent, claims 19 and 25: (IPR2023-00382, Ground 1)

- "**receiving** first audio data , first video data, and first **collaboration data**"
- "**transmitting** the first video and the first **collaboration data**"

IPR2023-00382, Ex. 1001 at 11:25-27, 11:46-47, 12:16-18, 12:37-38.

A POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE KNAPPE WITH ELLIOTT

KNAPPE



Ex. 1006 at FIG. 1.

ELLIOTT

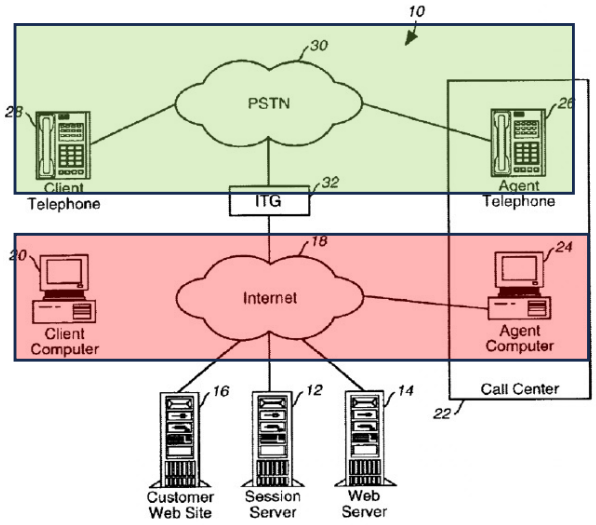
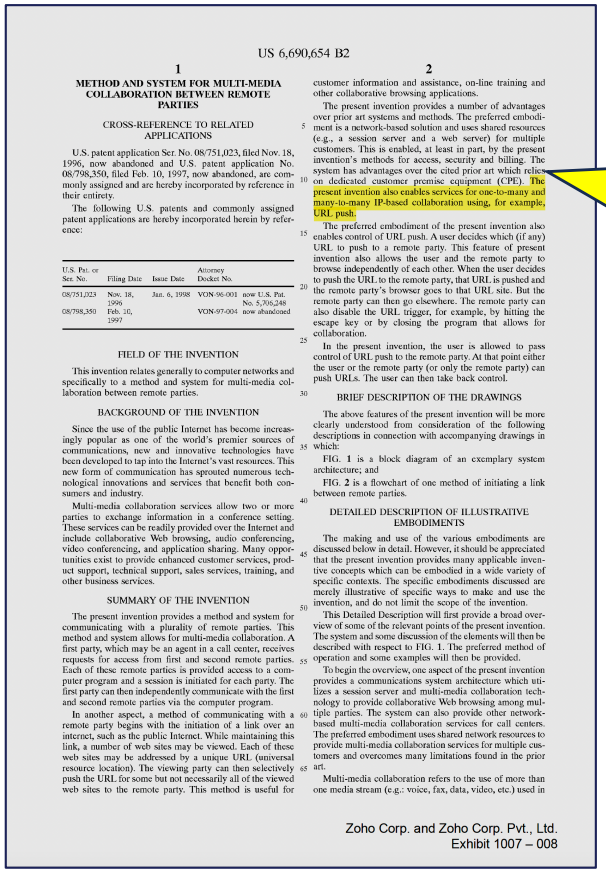


FIG. 1

Ex. 1007 at FIG. 1.

ELLIOTT DISCLOSES MULTI-PARTY CONFERENCING



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METHOD AND SYSTEM FOR MULTI-MEDIA COLLABORATION BETWEEN REMOTE PARTIES

CROSS-REFERENCE TO RELATED APPLICATIONS

U.S. patent application Ser. No. 08/751,023, filed Nov. 18, 1996, now abandoned and U.S. patent application No. 08/798,350, filed Feb. 10, 1997, now abandoned, are commonly assigned and are hereby incorporated by reference in their entirety.

The following U.S. patents and commonly assigned patent applications are hereby incorporated herein by reference:

U.S. Pat. or Ser. No.	Filing Date	Issue Date	Attorney Docket No.	
08/751,023	Nov. 18, 1996	Jan. 6, 1998	VON-96-001	now U.S. Pat. No. 5,706,248
08/798,350	Feb. 10, 1997		VON-97-004	now abandoned

FIELD OF THE INVENTION

This invention relates generally to computer networks and specifically to a method and system for multi-media collaboration between remote parties.

BACKGROUND OF THE INVENTION

Since the use of the public Internet has become increasingly popular as one of the world's premier sources of communications, new and innovative technologies have been developed to tap into the Internet's vast resources. This new form of communication has sprouted numerous technological innovations and services that benefit both consumers and industry.

Multi-media collaboration services allow two or more parties to exchange information in a conference setting. These services can be readily provided over the Internet and include collaborative Web browsing, audio conferencing, video conferencing, and application sharing. Many opportunities exist to provide enhanced customer services, product support, technical support, sales services, training, and other business services.

SUMMARY OF THE INVENTION

The present invention provides a method and system for communicating with a plurality of remote parties. This method and system allows for multi-media collaboration. A first party, which may be an agent in a call center, receives requests for access from first and second remote parties. Each of these remote parties is provided access to a computer program and a session is initiated for each party. The first party can then independently communicate with the first and second remote parties via the computer program.

In another aspect, a method of communicating with a remote party begins with the initiation of a link over an internet, such as the public Internet. While maintaining this link, a number of web sites may be viewed. Each of these web sites may be addressed by a unique URL (universal resource location). The viewing party can then selectively push the URL for some but not necessarily all of the viewed web sites to the remote party. This method is useful for

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customer information and assistance, on-line training and other collaborative browsing applications.

The present invention provides a number of advantages over prior art systems and methods. The preferred embodiment is a network-based solution and uses shared resources (e.g., a session server and a web server) for multiple customers. This is enabled, at least in part, by the present invention's methods for access, security and billing. The system has advantages over the cited prior art which relies on dedicated customer premise equipment (CPE). The present invention also enables services for one-to-many and many-to-many IP-based collaboration using, for example, URL push.

The preferred embodiment of the present invention also enables control of URL push. A user decides which (if any) URL to push to a remote party. This feature of present invention also allows the user and the remote party to browse independently of each other. When the user decides to push the URL to the remote party, that URL is pushed and the remote party's browser goes to that URL site. But the remote party can then go elsewhere. The remote party can also disable the URL trigger, for example, by hitting the escape key or by closing the program that allows for collaboration.

In the present invention, the user is allowed to pass control of URL push to the remote party. At that point either the user or the remote party (or only the remote party) can push URLs. The user can then take back control.

BRIEF DESCRIPTION OF THE DRAWINGS

The above features of the present invention will be more clearly understood from consideration of the following descriptions in connection with accompanying drawings in which:

FIG. **1** is a block diagram of an exemplary system architecture; and

FIG. **2** is a flowchart of one method of initiating a link between remote parties.

DETAILED DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

The making and use of the various embodiments are discussed below in detail. However, it should be appreciated that the present invention provides many applicable inventive concepts which can be embodied in a wide variety of specific contexts. The specific embodiments discussed are merely illustrative of specific ways to make and use the invention, and do not limit the scope of the invention.

This Detailed Description will first provide a broad overview of some of the relevant points of the present invention. The system and some discussion of the elements will then be described with respect to FIG. 1. The preferred method of operation and some examples will then be provided.

To begin the overview, one aspect of the present invention provides a communications system architecture which utilizes a session server and multi-media collaboration technology to provide collaborative Web browsing among multiple parties. The system can also provide other network-based multi-media collaboration services for call centers. The preferred embodiment uses shared network resources to provide multi-media collaboration services for multiple customers and overcomes many limitations found in the prior art.

Multi-media collaboration refers to the use of more than one media stream (e.g.: voice, fax, data, video, etc.) used in

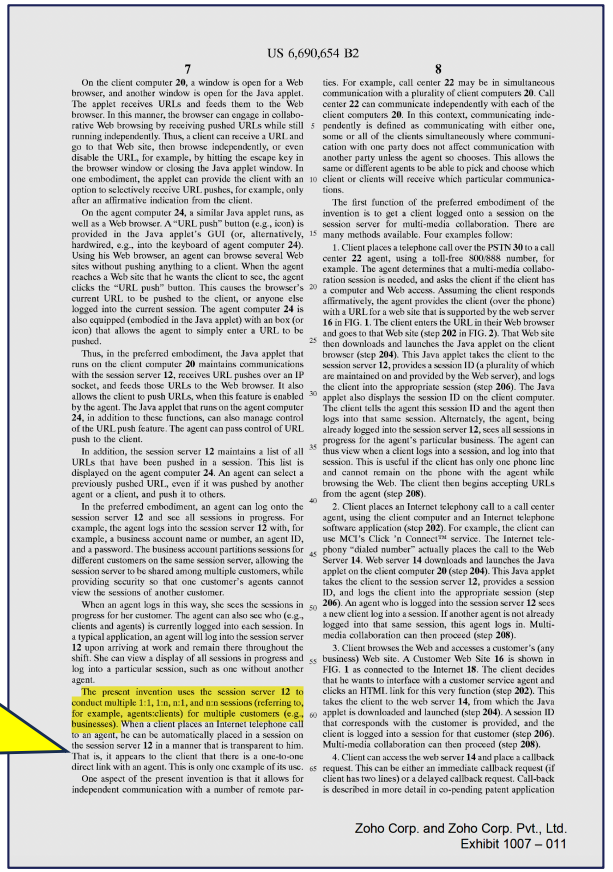
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Exhibit 1007 – 008

The present invention also enables services for one-to-one and **many-to-many IP-based collaboration**[.]

Ex. 1007 at 2:10-13.

The present invention uses the session server 12 to conduct multiple 1:1, **1:n, n:1, and n:n sessions** . . . for multiple customers[.]

Ex. 1007 at 7:58-61.



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On the client computer **20**, a window is open for a Web browser, and another window is open for the Java applet. The applet receives URLs and feeds them to the Web browser. In this manner, the browser can engage in collaborative Web browsing by receiving pushed URLs while still running independently. Thus, a client can receive a URL and go to that Web site, then browse independently, or even disable the URL, for example, by hitting the escape key in the browser window or closing the Java applet window. In one embodiment, the applet can provide the client with an option to selectively receive URL pushes, for example, only after an affirmative indication from the client.

On the agent computer **24**, a similar Java applet runs, as well as a Web browser. A "URL push" button (e.g., icon) is provided in the Java applet's GUI (or, alternatively, hardwired, e.g., into the keyboard of agent computer **24**). Using his Web browser, an agent can browse several Web sites without pushing anything to a client. When the agent reaches a Web site that he wants the client to see, the agent clicks the "URL push" button. This causes the browser's current URL to be pushed to the client, or anyone else logged into the current session. The agent computer **24** is also equipped (embodied in the Java applet) with an box (or icon) that allows the agent to simply enter a URL to be pushed.

Thus, in the preferred embodiment, the Java applet that runs on the client computer **20** maintains communications with the session server **12**, receives URL pushes over an IP socket, and feeds those URLs to the Web browser. It also allows the client to push URLs, when this feature is enabled by the agent. The Java applet that runs on the agent computer **24**, in addition to these functions, can also manage control of the URL push feature. The agent can pass control of URL push to the client.

In addition, the session server **12** maintains a list of all URLs that have been pushed in a session. This list is displayed on the agent computer **24**. An agent can select a previously pushed URL, even if it was pushed by another agent or a client, and push it to others.

In the preferred embodiment, an agent can log onto the session server **12** and see all sessions in progress. For example, the agent logs into the session server **12** with, for example, a business account name or number, an agent ID, and a password. The business account partitions sessions for different customers on the same session server, allowing the session server to be shared among multiple customers, while providing security so that one customer's agents cannot view the sessions of another customer.

When an agent logs in this way, she sees the sessions in progress for her customer. The agent can also see who (e.g., clients and agents) is currently logged into each session. In a typical application, an agent will log into the session server **12** upon arriving at work and remain there throughout the shift. She can view a display of all sessions in progress and log into a particular session, such as one without another agent.

The present invention uses the session server **12** to conduct multiple 1:1, 1:n, n:1, and n:n sessions (referring to, for example, agents:clients) for multiple customers (e.g., businesses). When a client places an Internet telephone call to an agent, he can be automatically placed in a session on the session server **12** in a manner that is transparent to him. That is, it appears to the client that there is a one-to-one direct link with an agent. This is only one example of its use.

One aspect of the present invention is that it allows for independent communication with a number of remote parties. For example, call center **22** may be in simultaneous communication with a plurality of client computers **20**. Call center **22** can communicate independently with each of the client computers **20**. In this context, communicating independently is defined as communicating with either one, some or all of the clients simultaneously where communication with one party does not affect communication with another party unless the agent so chooses. This allows the same or different agents to be able to pick and choose which client or clients will receive which particular communications.

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The first function of the preferred embodiment of the invention is to get a client logged onto a session on the session server for multi-media collaboration. There are many methods available. Four examples follow:

1. Client places a telephone call over the PSTN **30** to a call center **22** agent, using a toll-free 800/888 number, for example. The agent determines that a multi-media collaboration session is needed, and asks the client if the client has a computer and Web access. Assuming the client responds affirmatively, the agent provides the client (over the phone) with a URL for a web site that is supported by the web server **16** in FIG. **1**. The client enters the URL in their Web browser and goes to that Web site (step **202** in FIG. **2**). That Web site then downloads and launches the Java applet on the client browser (step **204**). This Java applet takes the client to the session server **12**, provides a session ID (a plurality of which are maintained on and provided by the Web server), and logs the client into the appropriate session (step **206**). The Java applet also displays the session ID on the client computer. The client tells the agent this session ID and the agent then logs into that same session. Alternately, the agent, being already logged into the session server **12**, sees all sessions in progress for the agent's particular business. The agent can thus view when a client logs into a session, and log into that session. This is useful if the client has only one phone line and cannot remain on the phone with the agent while browsing the Web. The client then begins accepting URLs from the agent (step **208**).

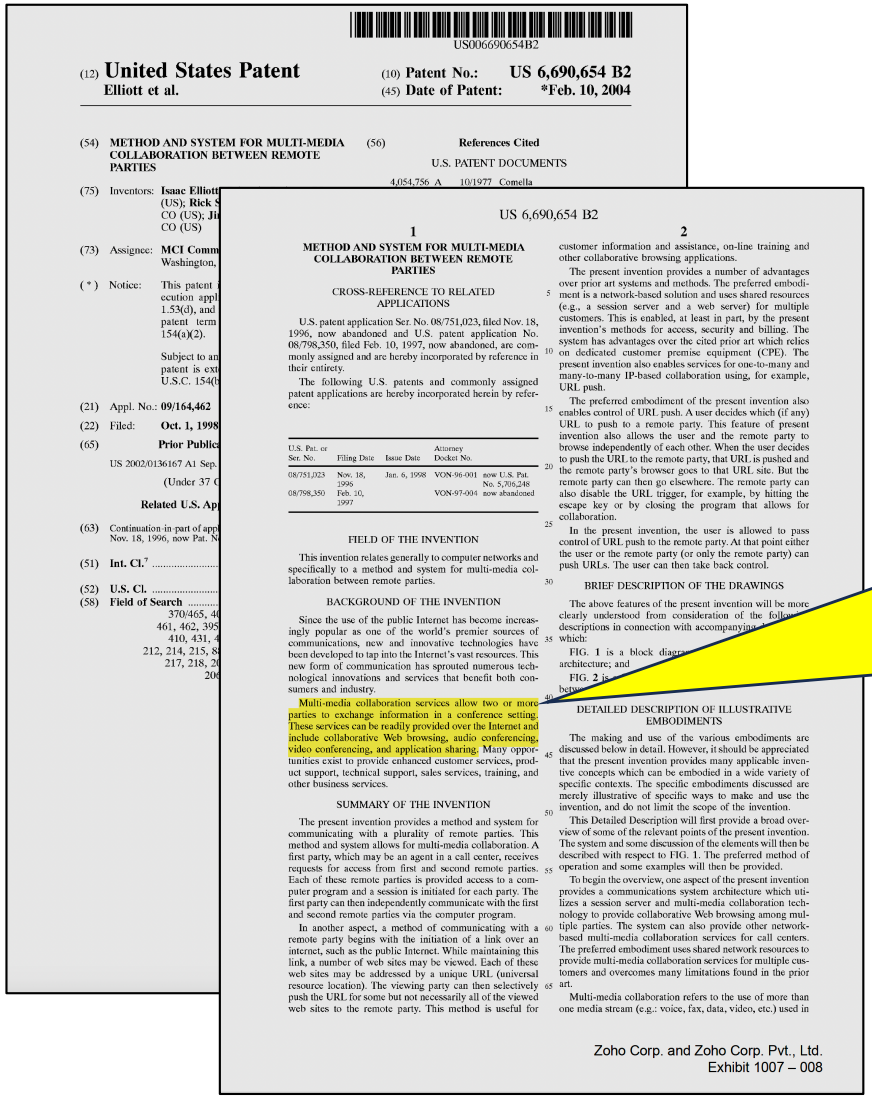
2. Client places an Internet telephony call to a call center agent, using the client computer and an Internet telephone software application (step **202**). For example, the client can use MCI's Click 'n Connect™ service. The Internet telephony "dialed number" actually places the call to the Web Server **14**. Web server **14** downloads and launches the Java applet on the client computer **20** (step **204**). This Java applet takes the client to the session server **12**, provides a session ID, and logs the client into the appropriate session (step **206**). An agent who is logged into the session server **12** sees a new client log into a session. If another agent is not already logged into that same session, this agent logs in. Multi-media collaboration can then proceed (step **208**).

3. Client browses the Web and accesses a customer's (any business) Web site. A Customer Web Site **16** is shown in FIG. **1** as connected to the Internet **18**. The client decides that he wants to interface with a customer service agent and clicks an HTML link for this very function (step **202**). This takes the client to the web server **14**, from which the Java applet is downloaded and launched (step **204**). A session ID that corresponds with the customer is provided, and the client is logged into a session for that customer (step **206**). Multi-media collaboration can then proceed (step **208**).

4. Client can access the web server **14** and place a callback request. This can be either an immediate callback request (if client has two lines) or a delayed callback request. Call-back is described in more detail in co-pending patent application

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Exhibit 1007 – 011

ELLIOTT DISCLOSES SHARING COLLABORATION DATA



US00669065 4B2

(12) **United States Patent**
Elliott et al.

(10) **Patent No.: US 6,690,654 B2**
(45) **Date of Patent: *Feb. 10, 2004**

(54) **METHOD AND SYSTEM FOR MULTI-MEDIA COLLABORATION BETWEEN REMOTE PARTIES**

(56) **References Cited**

U.S. PATENT DOCUMENTS

4,054,756 A 10/1977 Comella

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1

METHOD AND SYSTEM FOR MULTI-MEDIA COLLABORATION BETWEEN REMOTE PARTIES

CROSS-REFERENCE TO RELATED APPLICATIONS

U.S. patent application Ser. No. 08/751,023, filed Nov. 18, 1996, now abandoned and U.S. patent application No. 08/798,350, filed Feb. 10, 1997, now abandoned, are commonly assigned and are hereby incorporated by reference in their entirety.

The following U.S. patents and commonly assigned patent applications are hereby incorporated herein by reference:

U.S. Pat. or Ser. No.	Filing Date	Issue Date	Attorney Docket No.	
08/751,023	Nov. 18, 1996	Jan. 6, 1998	VON-96-001	now U.S. Pat. No. 5,706,248
08/798,350	Feb. 10, 1997		VON-97-004	now abandoned

FIELD OF THE INVENTION

This invention relates generally to computer networks and specifically to a method and system for multi-media collaboration between remote parties.

BACKGROUND OF THE INVENTION

Since the use of the public Internet has become increasingly popular as one of the world's premier sources of communications, new and innovative technologies have been developed to tap into the Internet's vast resources. This new form of communication has sprouted numerous technological innovations and services that benefit both consumers and industry.

Multi-media collaboration services allow two or more parties to exchange information in a conference setting. These services can be readily provided over the Internet and include collaborative Web browsing, audio conferencing, video conferencing, and application sharing. Many opportunities exist to provide enhanced customer services, product support, technical support, sales services, training, and other business services.

SUMMARY OF THE INVENTION

The present invention provides a method and system for communicating with a plurality of remote parties. This method and system allows for multi-media collaboration. A first party, which may be an agent in a call center, receives requests for access from first and second remote parties. Each of these remote parties is provided access to a computer program and a session is initiated for each party. The first party can then independently communicate with the first and second remote parties via the computer program.

In another aspect, a method of communicating with a remote party begins with the initiation of a link over an internet, such as the public Internet. While maintaining this link, a number of web sites may be viewed. Each of these web sites may be addressed by a unique URL (universal resource location). The viewing party can then selectively push the URL for some but not necessarily all of the viewed web sites to the remote party. This method is useful for

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customer information and assistance, on-line training and other collaborative browsing applications.

The present invention provides a number of advantages over prior art systems and methods. The preferred embodiment is a network-based solution and uses shared resources (e.g., a session server and a web server) for multiple customers. This is enabled, at least in part, by the present invention's methods for access, security and billing. The system has advantages over the cited prior art which relies on dedicated customer premise equipment (CPE). The present invention also enables services for one-to-many and many-to-many IP-based collaboration using, for example, URL push.

The preferred embodiment of the present invention also enables control of URL push. A user decides which (if any) URL to push to a remote party. This feature of present invention also allows the user and the remote party to browse independently of each other. When the user decides to push the URL to the remote party, that URL is pushed and the remote party's browser goes to that URL site. But the remote party can then go elsewhere. The remote party can also disable the URL trigger, for example, by hitting the escape key or by closing the program that allows for collaboration.

In the present invention, the user is allowed to pass control of URL push to the remote party. At that point either the user or the remote party (or only the remote party) can push URLs. The user can then take back control.

BRIEF DESCRIPTION OF THE DRAWINGS

The above features of the present invention will be more clearly understood from consideration of the following descriptions in connection with accompanying [illegible] which:

FIG. 1 is a block diagram [illegible] architecture; and

FIG. 2 is [illegible] between

DETAILED DESCRIPTION OF ILLUSTRATIVE EMBODIMENTS

The making and use of the various embodiments are discussed below in detail. However, it should be appreciated that the present invention provides many applicable inventive concepts which can be embodied in a wide variety of specific contexts. The specific embodiments discussed are merely illustrative of specific ways to make and use the invention, and do not limit the scope of the invention.

This Detailed Description will first provide a broad overview of some of the relevant points of the present invention. The system and some discussion of the elements will then be described with respect to FIG. 1. The preferred method of operation and some examples will then be provided.

To begin the overview, one aspect of the present invention provides a communications system architecture which utilizes a session server and multi-media collaboration technology to provide collaborative Web browsing among multiple parties. The system can also provide other network-based multi-media collaboration services for call centers. The preferred embodiment uses shared network resources to provide multi-media collaboration services for multiple customers and overcomes many limitations found in the prior art.

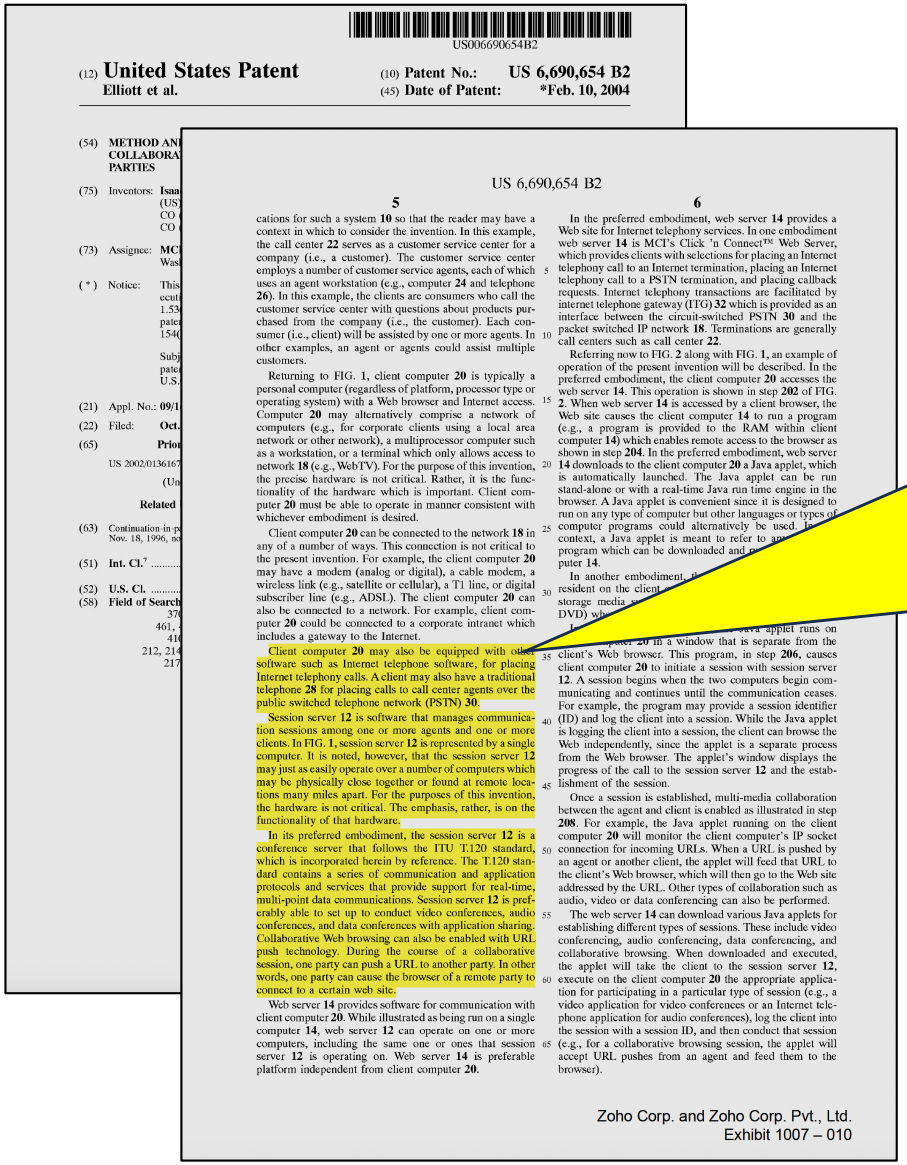
Multi-media collaboration refers to the use of more than one media stream (e.g.: voice, fax, data, video, etc.) used in

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Exhibit 1007 – 008

Multimedia **collaboration services** allow two or more parties to exchange information **in a conference setting**. These services can be readily provided over the Internet and include **collaborative Web browsing, audio conferencing, video conferencing, and application sharing.**

Ex. 1007 at 1:40-45; *see also* IPR2023-00382, Ex. 1003 ¶ 346.

ELLIOTT DISCLOSES SHARING COLLABORATION DATA



US00690654B2

(12) **United States Patent**
Elliott et al.

(10) **Patent No.: US 6,690,654 B2**
(45) **Date of Patent: *Feb. 10, 2004**

US 6,690,654 B2

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cations for such a system **10** so that the reader may have a context in which to consider the invention. In this example, the call center **22** serves as a customer service center for a company (i.e., a customer). The customer service center employs a number of customer service agents, each of which uses an agent workstation (e.g., computer **24** and telephone **26**). In this example, the clients are consumers who call the customer service center with questions about products purchased from the company (i.e., the customer). Each consumer (i.e., client) will be assisted by one or more agents. In other examples, an agent or agents could assist multiple customers.

Returning to FIG. **1**, client computer **20** is typically a personal computer (regardless of platform, processor type or operating system) with a Web browser and Internet access. Computer **20** may alternatively comprise a network of computers (e.g., for corporate clients using a local area network or other network), a multiprocessor computer such as a workstation, or a terminal which only allows access to network **18** (e.g., WebTV). For the purpose of this invention, the precise hardware is not critical. Rather, it is the functionality of the hardware which is important. Client computer **20** must be able to operate in manner consistent with whichever embodiment is desired.

Client computer **20** can be connected to the network **18** in any of a number of ways. This connection is not critical to the present invention. For example, the client computer **20** may have a modem (analog or digital), a cable modem, a wireless link (e.g., satellite or cellular), a T1 line, or digital subscriber line (e.g., ADSL). The client computer **20** can also be connected to a network. For example, client computer **20** could be connected to a corporate intranet which includes a gateway to the Internet.

Client computer **20** may also be equipped with other software such as Internet telephone software, for placing Internet telephony calls. A client may also have a traditional telephone **28** for placing calls to call center agents over the public switched telephone network (PSTN) **30**.

Session server **12** is software that manages communication sessions among one or more agents and one or more clients. In FIG. **1**, session server **12** is represented by a single computer. It is noted, however, that the session server **12** may just as easily operate over a number of computers which may be physically close together or found at remote locations many miles apart. For the purposes of this invention, the hardware is not critical. The emphasis, rather, is on the functionality of that hardware.

In its preferred embodiment, the session server **12** is a conference server that follows the ITU T.120 standard, which is incorporated herein by reference. The T.120 standard contains a series of communication and application protocols and services that provide support for real-time, multi-point data communications. Session server **12** is preferably able to set up to conduct video conferences, audio conferences, and data conferences with application sharing. Collaborative Web browsing can also be enabled with URL push technology. During the course of a collaborative session, one party can push a URL to another party. In other words, one party can cause the browser of a remote party to connect to a certain web site.

Web server **14** provides software for communication with client computer **20**. While illustrated as being run on a single computer **14**, web server **12** can operate on one or more computers, including the same one or ones that session server **12** is operating on. Web server **14** is preferable platform independent from client computer **20**.

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In the preferred embodiment, web server **14** provides a Web site for Internet telephony services. In one embodiment web server **14** is MCI's Click 'n Connect™ Web Server, which provides clients with selections for placing an Internet telephony call to an Internet termination, placing an Internet telephony call to a PSTN termination, and placing callback requests. Internet telephony transactions are facilitated by internet telephone gateway (ITG) **32** which is provided as an interface between the circuit-switched PSTN **30** and the packet switched IP network **18**. Terminations are generally call centers such as call center **22**.

Referring now to FIG. **2** along with FIG. **1**, an example of operation of the present invention will be described. In the preferred embodiment, the client computer **20** accesses the web server **14**. This operation is shown in step **202** of FIG. **2**. When web server **14** is accessed by a client browser, the Web site causes the client computer **14** to run a program (e.g., a program is provided to the RAM within client computer **14**) which enables remote access to the browser as shown in step **204**. In the preferred embodiment, web server **14** downloads to the client computer **20** a Java applet, which is automatically launched. The Java applet can be run stand-alone or with a real-time Java run time engine in the browser. A Java applet is convenient since it is designed to run on any type of computer but other languages or types of computer programs could alternatively be used. In this context, a Java applet is meant to refer to any such program which can be downloaded and executed on computer **14**.

In another embodiment, the [illegible] resident on the client computer's storage media [illegible] DVD) which [illegible] Java applet runs on computer **20** in a window that is separate from the client's Web browser. This program, in step **206**, causes client computer **20** to initiate a session with session server **12**. A session begins when the two computers begin communicating and continues until the communication ceases. For example, the program may provide a session identifier (ID) and log the client into a session. While the Java applet is logging the client into a session, the client can browse the Web independently, since the applet is a separate process from the Web browser. The applet's window displays the progress of the call to the session server **12** and the establishment of the session.

Once a session is established, multi-media collaboration between the agent and client is enabled as illustrated in step **208**. For example, the Java applet running on the client computer **20** will monitor the client computer's IP socket connection for incoming URLs. When a URL is pushed by an agent or another client, the applet will feed that URL to the client's Web browser, which will then go to the Web site addressed by the URL. Other types of collaboration such as audio, video or data conferencing can also be performed.

The web server **14** can download various Java applets for establishing different types of sessions. These include video conferencing, audio conferencing, data conferencing, and collaborative browsing. When downloaded and executed, the applet will take the client to the session server **12**, execute on the client computer **20** the appropriate application for participating in a particular type of session (e.g., a video application for video conferences or an Internet telephone application for audio conferences), log the client into the session with a session ID, and then conduct that session (e.g., for a collaborative browsing session, the applet will accept URL pushes from an agent and feed them to the browser).

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Exhibit 1007 – 010

Elliott discloses connecting calls "**over the public switched telephone network (PSTN 30)**" using "session server 12" that "follows the ITU T.120 standard . . . that **provides support for real-time multi-point data communications**. Session server 12 is preferably able to set up to conduct **video conferences with application sharing. Collaborative Web browsing can also be enabled**[.]"

Ex. 1007 at 5:35-61; *see also* IPR2023-00382, Ex. 1003 ¶¶ 346-347.

The Prior Art Renders Obvious the Mixing/Mixer Limitations

IPR2023-0377, Grounds 1-3
(U.S. Patent No. 9,253,332)

IPR2023-00382, Grounds 1-2
(U.S. Patent No. 9,843,612)

IPR2023-00378, Grounds 1-4
(U.S. Patent No. 9,094,525)

IPR2023-00379, Ground 1
(U.S. Patent No. 9,094,525)

KNAPPE RENDERS OBVIOUS THE MIXING LIMITATIONS

1) The ’612 patent, claims 19 and 25: (IPR2023-00382, Grounds 1 and 2)

- “**mixing the first audio data** [first remote client] **with the second audio data** [second remote client]”
- “**mixing the second audio data with the third audio data** [PSTN client]”
- **“mixing the first audio data with the third audio data”**

IPR2023-00382, Ex. 1001 at 11:33-38, 12:24-30.

2) The ’332 patent, claims 1-2 and 8-9: (IPR2023-00377, Grounds 1-3)

- “**mixing first audio data** from the PSTN **with the second audio data** from the moderator”
- **“mixing the second audio data . . . with the third audio data from the remote client**.”
- “**mixing the first audio data . . . with the third audi0 data”**

IPR2023-00377, Ex. 1001 at 9:57-59, 10:20-22, 10:36-39.

3) The ’525 patent, claims 1-2, 9-10, 17-18: (IPR2023-00378, Grounds 1-4, IPR2023-00379, Ground 1)

- “**first mixer** that mixes a PSTN client audio . . . with a moderator audio-video data”
- **“second mixer** that mixes the moderator audio-video data . .. with the remote client audio-video data”
- **“third mixer** that mixes audio data from the PSTN with the audio-video data stream from the remote client

IPR2023-00378, -379, Ex. 1001 at 10:4-14, 10:18-21, 10:60-11:2, 11:7-10, 12:4-19, 12:24-27.

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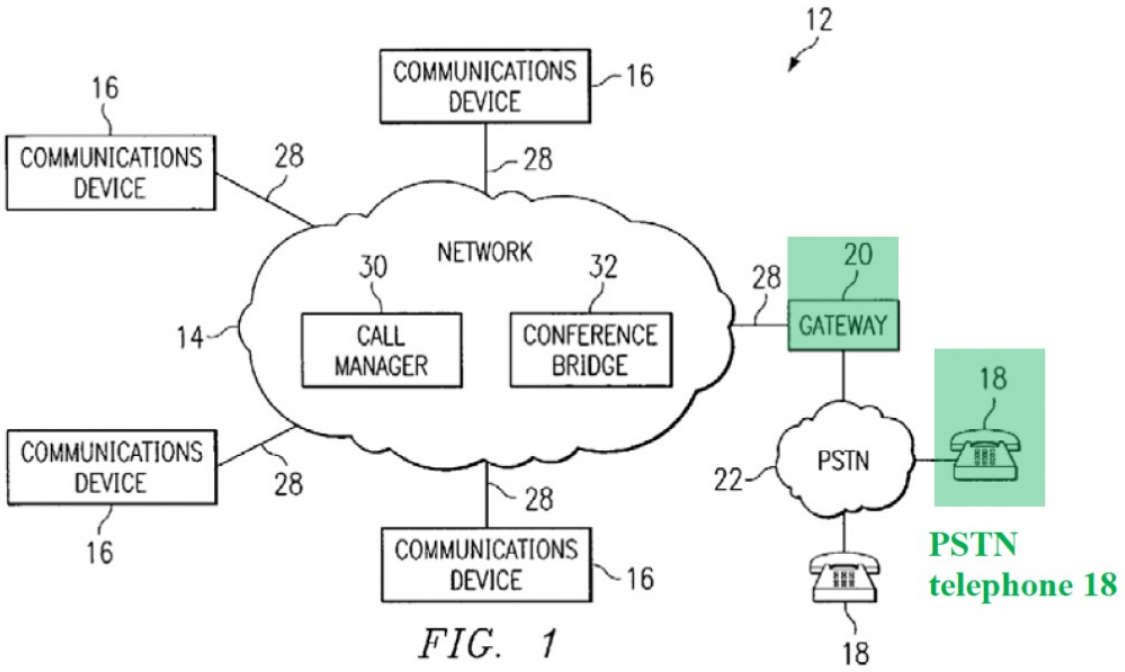
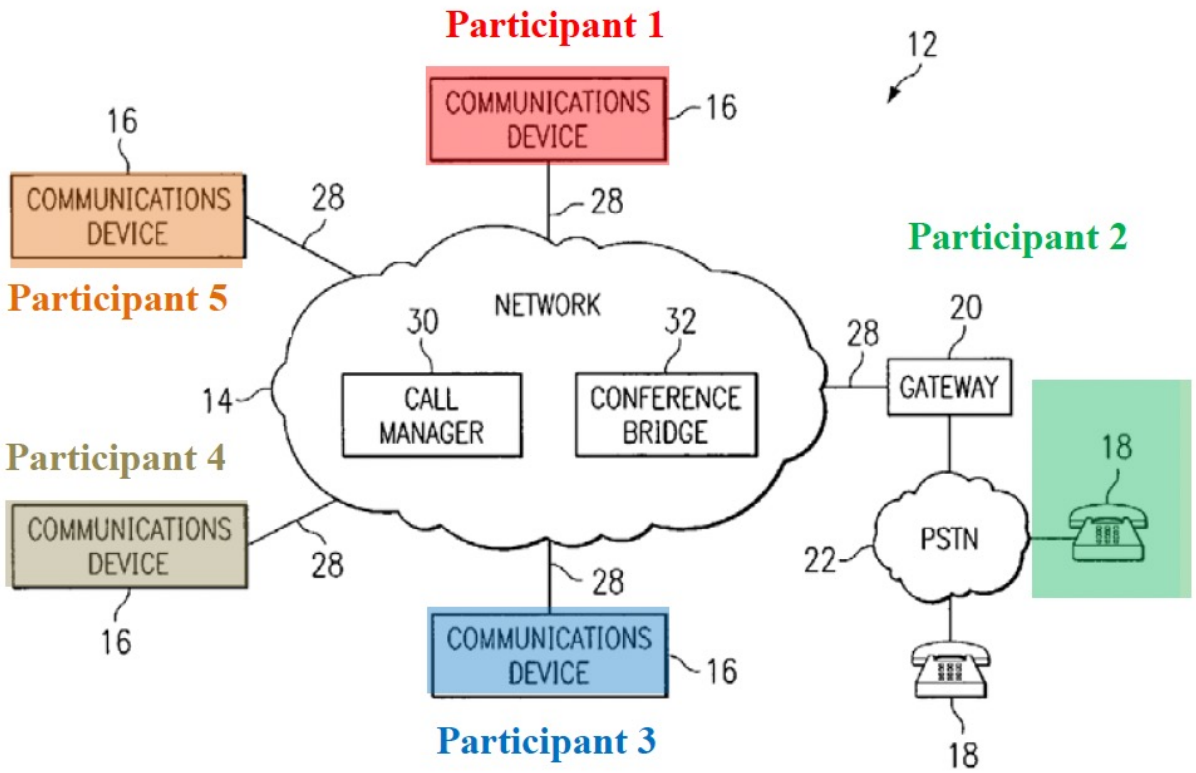


FIG. 1

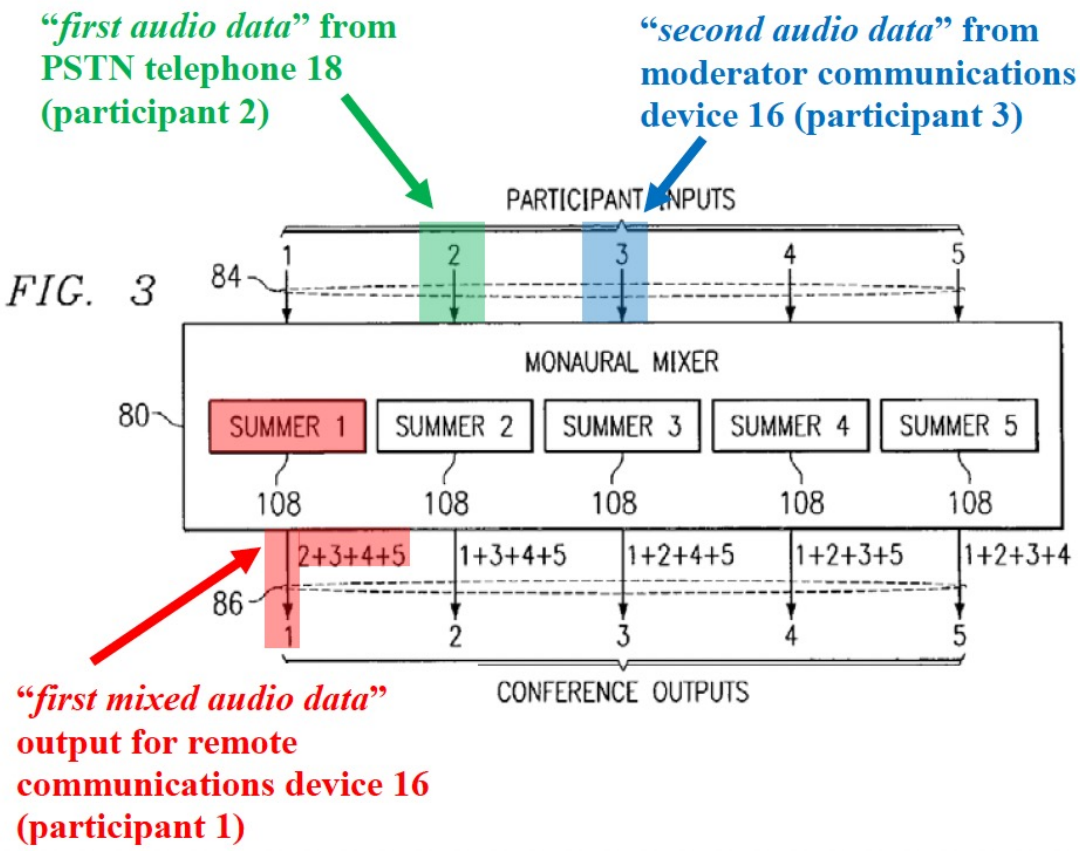
Ex. 1006 at FIG. 1 (annotations added), cited by IPR2023-00377, Pet. at 30; *see also* IPR2023-382, Pet. at 45.

KNAPPE



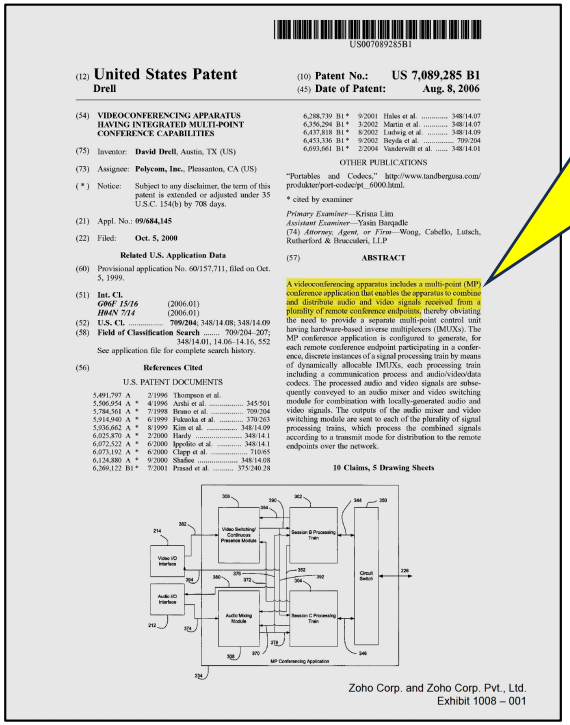
Ex. 1006 at FIG. 1 (annotations added), cited by IPR2023-00377, Pet. at 36; *see also* IPR2023-382, Pet. at 34.

KNAPPE



Ex. 1006 at FIG. 3 (annotations added), cited by IPR2023-00377, Pet. at 40; *see also* IPR2023-00382, Pet. at 48 & IPR2023-00378, Pet. at 34.

DRELL



US007089285B1

(12) **United States Patent**
Drell

(10) **Patent No.: US 7,089,285 B1**
(45) **Date of Patent: Aug. 8, 2006**

(54) **VIDEOCONFERENCING APPARATUS HAVING INTEGRATED MULTI-POINT CONFERENCE CAPABILITIES**

(75) Inventor: **David Drell**, Austin, TX (US)

(73) Assignee: **Polycom, Inc.**, Pleasanton, CA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 708 days.

(21) Appl. No.: **09/684,145**

(22) Filed: **Oct. 5, 2000**

Related U.S. Application Data

(60) Provisional application No. 60/157,711, filed on Oct. 5, 1999.

(51) **Int. Cl.**
G06F 15/16 (2006.01)
H04N 7/14 (2006.01)

(52) **U.S. Cl.** **709/204**; 348/14.08; 348/14.09

(58) **Field of Classification Search** 709/204–207; 348/14.01, 14.06–14.16, 552
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,491,797	A	2/1996	Thompson et al.	
5,566,954	A *	4/1996	Arihi et al.	345/501
5,784,561	A *	7/1998	Bruno et al.	709/204
5,914,940	A *	6/1999	Fukuoka et al.	370/263
5,936,662	A *	8/1999	Kim et al.	348/14.09
6,025,870	A *	2/2000	Hardy	348/14.1
6,072,522	A *	6/2000	Ippolito et al.	348/14.1
6,073,192	A *	6/2000	Clapp et al.	710/65
6,126,880	A *	9/2000	Shallow	348/14.08
6,269,122	B1 *	7/2001	Prasad et al.	375/240.28
6,288,739	B1 *	9/2001	Hales et al.	348/14.07
6,356,294	B1 *	3/2002	Martin et al.	348/14.07
6,437,818	B1 *	8/2002	Ludwig et al.	348/14.09
6,453,336	B1 *	9/2002	Beyda et al.	709/204
6,693,661	B1 *	2/2004	Vanderwilt et al.	348/14.01

OTHER PUBLICATIONS

"Portables and Codecs," http://www.tandbergusa.com/produkter/port-codec/pt_6000.html.

* cited by examiner

Primary Examiner—Kristua Lim
Assistant Examiner—Yisin Barqadle
(74) *Attorney, Agent, or Firm*—Wong, Cabello, Lutsch, Rutherford & Brucculeri, LLP

(57) **ABSTRACT**

A videoconferencing apparatus includes a multi-point (MP) conference application that enables the apparatus to combine and distribute audio and video signals received from a plurality of remote conference endpoints, thereby obviating the need to provide a separate multi-point control unit having hardware-based inverse multiplexers (IMUXs). The MP conference application is configured to generate, for each remote conference endpoint participating in a conference, discrete instances of a signal processing train by means of dynamically allocable IMUXs, each processing train including a communication process and audio/video/data codecs. The processed audio and video signals are subsequently conveyed to an audio mixer and video switching module for combination with locally-generated audio and video signals. The outputs of the audio mixer and video switching module are sent to each of the plurality of signal processing trains, which process the combined signals according to a transmit mode for distribution to the remote endpoints over the network.

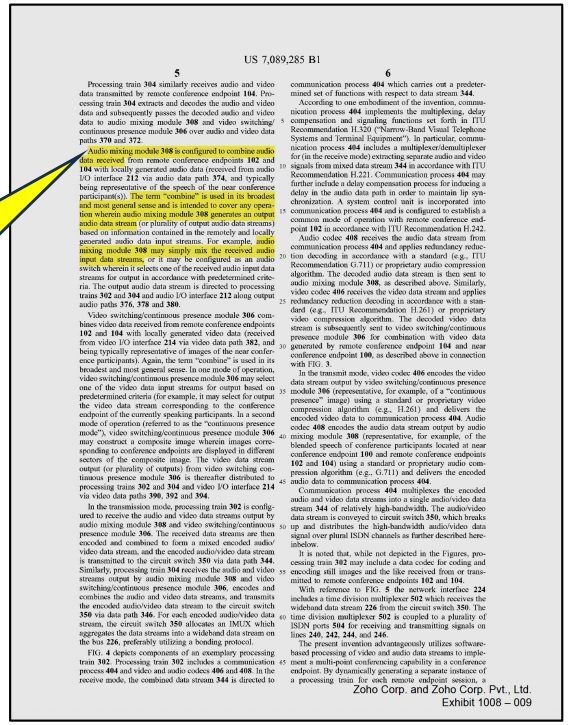
10 Claims, 5 Drawing Sheets

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Exhibit 1008 – 001

A video conferencing apparatus includes a **multi-point (MP) conferencing application** that enables the apparatus **to combine and distribute audio and video signals received from a plurality of remote conference endpoints.**

Ex. 1008 at Abstract.

Audio mixing module 308 is configured **to combine audio data** . . . . . The term "combine" is used in the most general sense and is intended to cover any operation wherein audio mixing module 308 generates an output data stream . . . **audio mixing module 308 may simply mix the received audio input data streams.**



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Processing train **304** similarly receives audio and video data transmitted by remote conference endpoint **104**. Processing train **304** extracts and decodes the audio and video data and subsequently passes the decoded audio and video data to audio mixing module **308** and video switching/continuous presence module **306** over audio and video data paths **370** and **372**.

Audio mixing module **308** is configured to combine audio data received from remote conference endpoints **102** and **104** with locally generated audio data (received from audio I/O interface **212** via audio data path **374**, and typically being representative of the speech of the near conference participant(s)). The term "combine" is used in its broadest and most general sense and is intended to cover any operation wherein audio mixing module **308** generates an output audio data stream (or plurality of output audio data streams) based on information contained in the remotely and locally generated audio data input streams. For example, audio mixing module **308** may simply mix the received audio input data streams, or it may be configured as an audio switch wherein it selects one of the received audio input data streams for output in accordance with predetermined criteria. The output audio data stream is directed to processing trains **302** and **304** and audio I/O interface **212** along output audio paths **376**, **378** and **380**.

Video switching/continuous presence module **306** combines video data received from remote conference endpoints **102** and **104** with locally generated video data (received from video I/O interface **214** via video data path **382**, and being typically representative of images of the near conference participants). Again, the term "combine" is used in its broadest and most general sense. In one mode of operation, video switching/continuous presence module **306** may select one of the video data input streams for output based on predetermined criteria (for example, it may select for output the video data stream corresponding to the conference endpoint of the currently speaking participants. In a second mode of operation (referred to as the "continuous presence mode"), video switching/continuous presence module **306** may construct a composite image wherein images corresponding to conference endpoints are displayed in different sectors of the composite image. The video data stream output (or plurality of outputs) from video switching/continuous presence module **306** is thereafter distributed to processing trains **302** and **304** and video I/O interface **214** via video data paths **390**, **392** and **394**.

In the transmission mode, processing train **302** is configured to receive the audio and video data streams output by audio mixing module **308** and video switching/continuous presence module **306**. The received data streams are then encoded and combined to form a mixed encoded audio/video data stream, and the encoded audio/video data stream is transmitted to the circuit switch **350** via data path **344**. Similarly, processing train **304** receives the audio and video streams output by audio mixing module **308** and video switching/continuous presence module **306**, encodes and combines the audio and video data streams, and transmits the encoded audio/video data stream to the circuit switch **350** via data path **346**. For each encoded audio/video data stream, the circuit switch **350** allocates an IMUX which aggregates the data streams into a wideband data stream on the bus **226**, preferably utilizing a bonding protocol.

FIG. **4** depicts components of an exemplary processing train **302**. Processing train **302** includes a communication process **404** and video and audio codecs **406** and **408**. In the receive mode, the combined data stream **344** is directed to communication process **404** which carries out a predetermined set of functions with respect to data stream **344**.

According to one embodiment of the invention, communication process **404** implements the multiplexing, delay compensation and signaling functions set forth in ITU Recommendation H.320 ("Narrow-Band Visual Telephone Systems and Terminal Equipment"). In particular, communication process **404** includes a multiplexer/demultiplexer for (in the receive mode) extracting separate audio and video signals from mixed data stream **344** in accordance with ITU Recommendation H.221. Communication process **404** may further include a delay compensation process for inducing a delay in the audio data path in order to maintain lip synchronization. A system control unit is incorporated into communication process **404** and is configured to establish a common mode of operation with remote conference endpoint **102** in accordance with ITU Recommendation H.242.

Audio codec **408** receives the audio data stream from communication process **404** and applies redundancy reduction decoding in accordance with a standard (e.g., ITU Recommendation G.711) or proprietary audio compression algorithm. The decoded audio data stream is then sent to audio mixing module **308**, as described above. Similarly, video codec **406** receives the video data stream and applies redundancy reduction decoding in accordance with a standard (e.g., ITU Recommendation H.261) or proprietary video compression algorithm. The decoded video data stream is subsequently sent to video switching/continuous presence module **306** for combination with video data generated by remote conference endpoint **104** and near conference endpoint **100**, as described above in connection with FIG. **3**.

In the transmit mode, video codec **406** encodes the video data stream output by video switching/continuous presence module **306** (representative, for example, of a "continuous presence" image) using a standard or proprietary video compression algorithm (e.g., H.261) and delivers the encoded video data to communication process **404**. Audio codec **408** encodes the audio data stream output by audio mixing module **308** (representative, for example, of the blended speech of conference participants located at near conference endpoint **100** and remote conference endpoints **102** and **104**) using a standard or proprietary audio compression algorithm (e.g., G.711) and delivers the encoded audio data to communication process **404**.

Communication process **404** multiplexes the encoded audio and video data streams into a single audio/video data stream **344** of relatively high-bandwidth. The audio/video data stream is conveyed to circuit switch **350**, which breaks up and distributes the high-bandwidth audio/video data signal over plural ISDN channels as further described hereinbelow.

It is noted that, while not depicted in the Figures, processing train **302** may include a data codec for coding and encoding still images and the like received from or transmitted to remote conference endpoints **102** and **104**.

With reference to FIG. **5** the network interface **224** includes a time division multiplexer **502** which receives the wideband data stream **226** from the circuit switch **350**. The time division multiplexer **502** is coupled to a plurality of ISDN ports **504** for receiving and transmitting signals on lines **240**, **242**, **244**, and **246**.

The present invention advantageously utilizes software-based processing of video and audio data streams to implement a multi-point conferencing capability in a conference endpoint. By dynamically generating a separate instance of a processing train for each remote endpoint session, a

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Exhibit 1008 – 009

Ex. 1008 at 5:8-21; *see also* IPR2023-00378, -379, Ex. 1003 ¶ 154 (cited at IPR2023-00378, Pet. (Paper No. 1) at 18-20, 30-35, 41-43, 48-49 & IPR2023-00379, Pet. (Paper No. 1) at 17-19, 34-42, 47).

KNAPPE

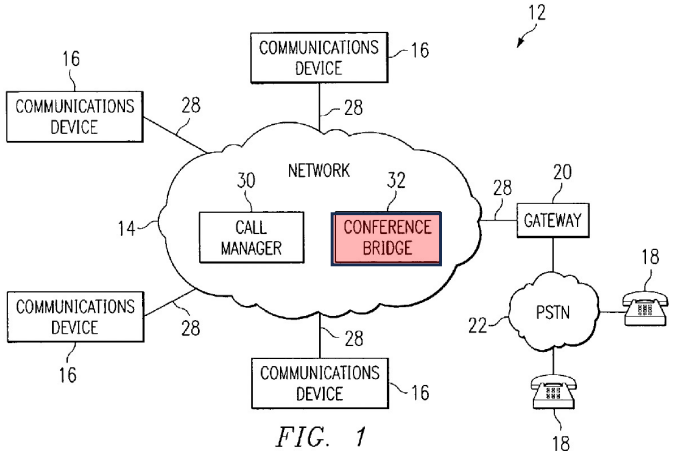


FIG. 1

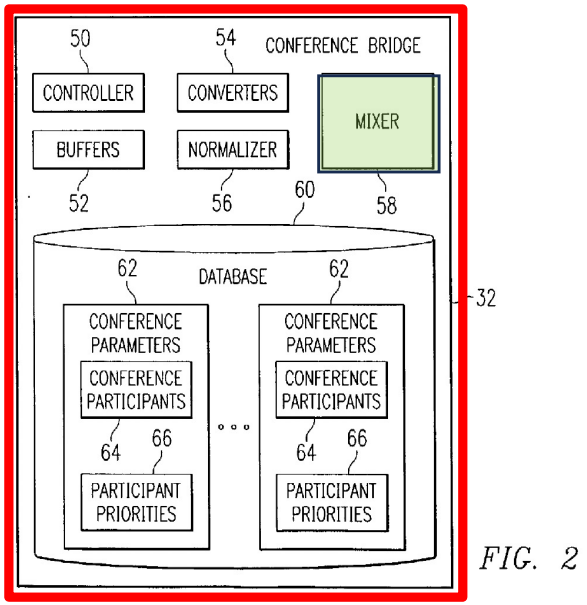


FIG. 2

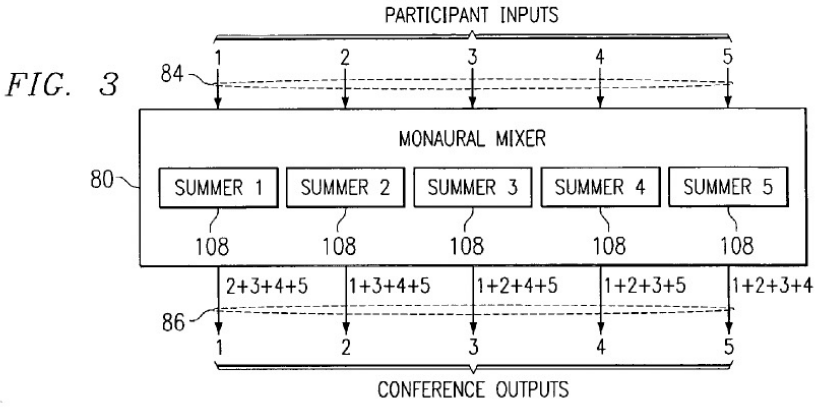


FIG. 3

Ex. 1006 at FIGS. 1-3 (highlighting added).

IPR2023-00378, Pet. (Paper No. 1) at 18-20, 30-35, 41-43, 48-49 & IPR2023-00379, Pet. (Paper No. 1) at 17-19, 34-42, 47).

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE DRELL WITH KNAPPE [FOR 525 PATENT, IPR 2023-00378, -0379 ONLY]

159. In summary, a POSITA would have been motivated to apply Knappe's multiple-summer teachings when implementing a mixer in Drell because doing so would beneficially (1) further Drell's goal of generating separate instances for each conference endpoint, (2) increase intelligibility of the moderator (3) enhance the moderator's control over the conference, and (4) exclude audio of the participant receiving the mixed audio thereby avoiding undesirable audio effects, such as

feedback or echo. Each of these reasons separately and together would have motivated a POSITA.

IPR2023-00378, -00379, Ex. 1003 ¶ 159 *(see also* ¶¶ 154-160).

IPR2023-00378, Pet. (Paper No. 1) at 18-20, 30-35, 41-43, 48-49 & IPR2023-00379, Pet. (Paper No. 1) at 17-19, 34-42, 47.

POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE DRELL WITH KNAPPE [FOR 525 PATENT, IPR 2023-00378, -0379 ONLY]

156. It would have been obvious to a POSITA, when implementing a mixer in Drell, to use a plurality of summers (one for each participant) as Knappe teaches, because this would allow for generating separate audio instances for each conference endpoint. Such an implementation would complement Drell's teaching of generating separate instances for each endpoint. *See* Drell (Ex. 1008) at 2:19-21 ("Generally, the MP conferencing application generates separate processing trains for signal streams from/to each of the remote conference endpoints."); *id*. at 6:66-7:4 ("By dynamically generating a separate instance of a processing train for each remote endpoint session, a videoconferencing system embodying the invention may easily and flexibly accommodate endpoint sessions comprising a range of connection bandwidths and communication protocols.").

157. Further, a POSITA would have applied Knappe's mixing teachings to Drell because using separate summers (one for each participant) would facilitate providing higher prominence to the moderator and "increasing the intelligibility of the moderator relative to the remaining participants." *See, e.g.*, Knappe (Ex. 1006) at 2:30-34, 6:59-7:17; *see also id*. at 2:40-46, 8:10-12. The increased intelligibility of the moderator enhances the moderator's control over the conference. *See, e.g.*, *id*. at 1:59-67.

IPR2023-00378, -00379, Ex. 1003 ¶¶ 156-157.

IPR2023-00378, Pet. (Paper No. 1) at 18-20, 30-35, 41-43, 48-49 & IPR2023-00379, Pet. (Paper No. 1) at 17-19, 34-42, 47.

The Prior Art Renders Obvious the "Moderator" Limitations

IPR2023-0378, Grounds 1-4
(U.S. Patent No. 9,094,525)

IPR2023-0379, Ground 1
(U.S. Patent No. 9,094,525)

THE MODERATOR LIMITATIONS

1) The ’525 patent, independent claims 1, 9, and 17: (IPR2023-00378, Grounds 1-4; IPR2023-00377, Ground 1)

1. A system for supporting a multi-participant conference call comprising:

a first mixer that mixes a Public Switched Telephone Network (PSTN) client audio data stream with a moderator audio-video data stream into a first mixed data stream;

a first transport output that transmits the first mixed data stream to at least one remote client that receives the first mixed data stream, the at least one remote client communicatively coupled to the Internet, which generates a remote client audio-video data stream;

a second mixer that mixes the moderator audio-video data stream with the remote client audio-video data stream into a second mixed data stream; and

a second transport output that transmits a mixed audio data stream, corresponding to the second mixed data stream, to the PSTN client.

17. A computer-implemented multi-participant conference system utilizing a local moderator computer, the system comprising:

a voice over IP decoder configured to decode audio data received by the local moderator computer from a telephone through a public switched telephone network (PSTN);

a voice over IP encoder configured to compress audio data transported from the local moderator computer to the telephone through the PSTN;

a first mixer configured to mix decoded audio data received by the local moderator computer from the telephone through the PSTN with audio data received by the local moderator computer from a local audio input device and video data received by the local moderator computer from a local video input device into an audio-video stream for transmission through an Internet to a first remote computer and a second remote computer;

a second mixer configured to mix audio data received by the local moderator computer from the first remote computer and the second remote computer through the Internet with audio data received by the local moderator computer from the local audio input device into an audio stream for transmission to the telephone through the PSTN; and

a data collaboration unit configured to transmit collaboration data from the local moderator through the internet to the first remote computer and the second remote computer.

IPR2023-00378, -379, Ex. 1001 at 10:4-14, 12:4-18.

A POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE DRELL WITH KNAPPE

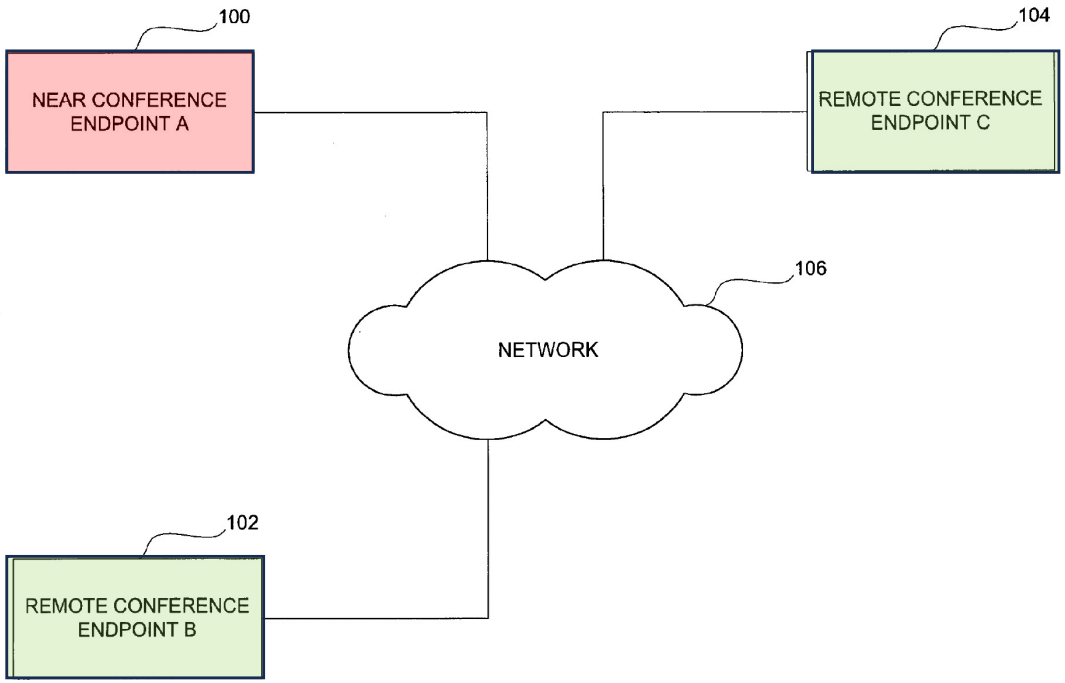
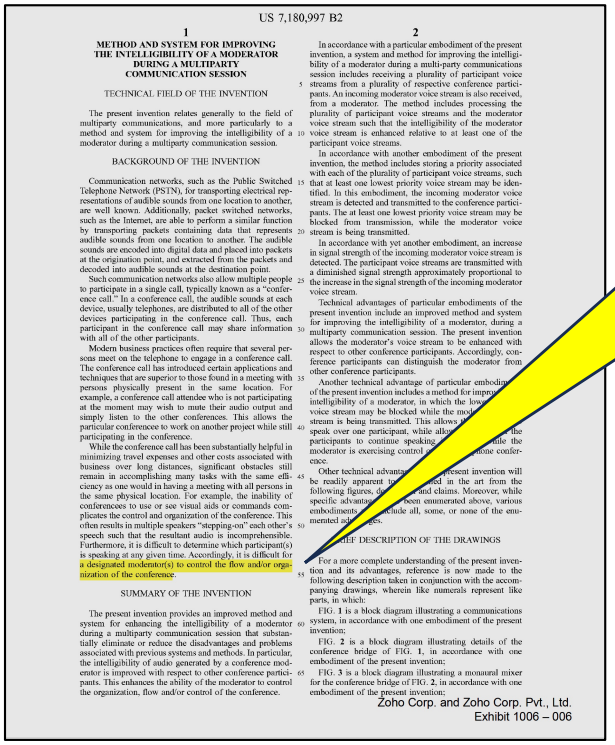


FIG. 1

Ex. 1008 at FIG. 1 (annotated); *see* IPR2023-00378, Pet. at 18, 29-30 & Reply at 7; IPR2023-00379, Pet. at 17, 23-24, 58-59 & Reply at 13-14.

KNAPPE TEACHES USE OF A MODERATOR IN MULTI-PARTY CONFERENCING



US 7,180,997 B2

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METHOD AND SYSTEM FOR IMPROVING THE INTELLIGIBILITY OF A MODERATOR DURING A MULTIPARTY COMMUNICATION SESSION

TECHNICAL FIELD OF THE INVENTION

The present invention relates generally to the field of multiparty communications, and more particularly to a method and system for improving the intelligibility of a moderator during a multiparty communication session.

BACKGROUND OF THE INVENTION

Communication networks, such as the Public Switched Telephone Network (PSTN), for transporting electrical representations of audible sounds from one location to another, are well known. Additionally, packet switched networks, such as the Internet, are able to perform a similar function by transporting packets containing data that represents audible sounds from one location to another. The audible sounds are encoded into digital data and placed into packets at the origination point, and extracted from the packets and decoded into audible sounds at the destination point.

Such communication networks also allow multiple people to participate in a single call, typically known as a "conference call." In a conference call, the audible sounds at each device, usually telephones, are distributed to all of the other devices participating in the conference call. Thus, each participant in the conference call may share information with all of the other participants.

Modern business practices often require that several persons meet on the telephone to engage in a conference call. The conference call has introduced certain applications and techniques that are superior to those found in a meeting with persons physically present in the same location. For example, a conference call attendee who is not participating at the moment may wish to mute their audio output and simply listen to the other conferences. This allows the particular conferencees to work on another project while still participating in the conference.

While the conference call has been substantially helpful in minimizing travel expenses and other costs associated with business over long distances, significant obstacles still remain in accomplishing many tasks with the same efficiency as one would in having a meeting with all persons in the same physical location. For example, the inability of conferencees to use or see visual aids or commands complicates the control and organization of the conference. This often results in multiple speakers "stepping-on" each other's speech such that the resultant audio is incomprehensible. Furthermore, it is difficult to determine which participant(s) is speaking at any given time. Accordingly, it is difficult for a designated moderator(s) to control the flow and/or organization of the conference.

SUMMARY OF THE INVENTION

The present invention provides an improved method and system for enhancing the intelligibility of a moderator during a multiparty communication session that substantially eliminate or reduce the disadvantages and problems associated with previous systems and methods. In particular, the intelligibility of audio generated by a conference moderator is improved with respect to other conference participants. This enhances the ability of the moderator to control the organization, flow and/or control of the conference.

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In accordance with a particular embodiment of the present invention, a system and method for improving the intelligibility of a moderator during a multi-party communications session includes receiving a plurality of participant voice streams from a plurality of respective conference participants. An incoming moderator voice stream is also received, from a moderator. The method includes processing the plurality of participant voice streams and the moderator voice stream such that the intelligibility of the moderator voice stream is enhanced relative to at least one of the participant voice streams.

In accordance with another embodiment of the present invention, the method includes storing a priority associated with each of the plurality of participant voice streams, such that at least one lowest priority voice stream may be identified. In this embodiment, the incoming moderator voice stream is detected and transmitted to the conference participants. The at least one lowest priority voice stream may be blocked from transmission, while the moderator voice stream is being transmitted.

In accordance with yet another embodiment, an increase in signal strength of the incoming moderator voice stream is detected. The participant voice streams are transmitted with a diminished signal strength approximately proportional to the increase in the signal strength of the incoming moderator voice stream.

Technical advantages of particular embodiments of the present invention include an improved method and system for improving the intelligibility of a moderator, during a multiparty communication session. The present invention allows the moderator's voice stream to be enhanced with respect to other conference participants. Accordingly, conference participants can distinguish the moderator from other conference participants.

Another technical advantage of particular embodi[ments] of the present invention includes a method for improv[ing] intelligibility of a moderator, in which the low[est] voice stream may be blocked while the mod[erator] stream is being transmitted. This allows [the] speak over one participant, while allo[wing] participants to continue speaking. [Thus] the moderator is exercising control [over] conference.

Other technical advantag[es] present invention will be readily apparent [to] in the art from the following figures, [descriptions] and claims. Moreover, while specific advantag[es] been enumerated above, various embodiments [may] include all, some, or none of the enumerated [advan]tages.

BRIEF DESCRIPTION OF THE DRAWINGS

For a more complete understanding of the present invention and its advantages, reference is now made to the following description taken in conjunction with the accompanying drawings, wherein like numerals represent like parts, in which:

FIG. **1** is a block diagram illustrating a communications system, in accordance with one embodiment of the present invention;

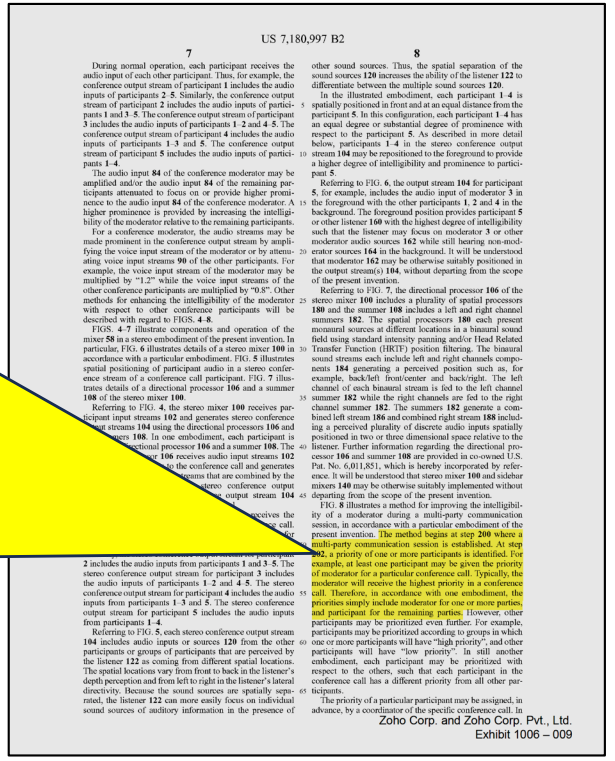
FIG. **2** is a block diagram illustrating details of the conference bridge of FIG. **1**, in accordance with one embodiment of the present invention;

FIG. **3** is a block diagram illustrating a monaural mixer for the conference bridge of FIG. **2**, in accordance with one embodiment of the present invention;

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Exhibit 1006 – 006

"**. . . a designated moderator(s) to control the flow and/or organization of the conference.**"

"The method begins at step 200 where a multi-party communication session is established. At step 202 a priority of one or more participants is identified. For example, at least **one participant may be given the priority of a moderator for a particular conference call.** … [T]he priorities simply include moderator for one or more parties and participant for the remaining parties."



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During normal operation, each participant receives the audio input of each other participant. Thus, for example, the conference output stream of participant **1** includes the audio inputs of participants **2**-**5**. Similarly, the conference output stream of participant **2** includes the audio inputs of participants **1** and **3**-**5**. The conference output stream of participant **3** includes the audio inputs of participants **1**-**2** and **4**-**5**. The conference output stream of participant **4** includes the audio inputs of participants **1**-**3** and **5**. The conference output stream of participant **5** includes the audio inputs of participants **1**-**4**.

The audio input **84** of the conference moderator may be amplified and/or the audio input **84** of the remaining participants attenuated to focus on or provide higher prominence to the audio input **84** of the conference moderator. A higher prominence is provided by increasing the intelligibility of the moderator relative to the remaining participants.

For a conference moderator, the audio streams may be made prominent in the conference output stream by amplifying the voice input stream of the moderator or by attenuating voice input streams **90** of the other participants. For example, the voice input stream of the moderator may be multiplied by "1.2" while the voice input streams of the other conference participants are multiplied by "0.8". Other methods for enhancing the intelligibility of the moderator with respect to other conference participants will be described with regard to FIGS. **4**-**8**.

FIGS. **4**-**7** illustrate components and operation of the mixer **58** in a stereo embodiment of the present invention. In particular, FIG. **6** illustrates details of a stereo mixer **100** in accordance with a particular embodiment. FIG. **5** illustrates spatial positioning of participant audio in a stereo conference stream of a conference call participant. FIG. **7** illustrates details of a directional processor **106** and a summer **108** of the stereo mixer **100**.

Referring to FIG. **4**, the stereo mixer **100** receives participant input streams **102** and generates stereo conference [output streams **104** using the directional processors **106** and] summers **108**. In one embodiment, each participant is [provided] a directional processor **106** and a summer **108**. The [mixer] **106** receives audio input streams **102** [from] to the conference call and generates [stereo] streams that are combined by the [summers] stereo conference output [streams] output stream **104** [for] receives the [participant] call.

2 includes the audio inputs from participants **1** and **3**-**5**. The stereo conference output stream for participant **3** includes the audio inputs of participants **1**-**2** and **4**-**5**. The stereo conference output stream for participant **4** includes the audio inputs from participants **1**-**3** and **5**. The stereo conference output stream for participant **5** includes the audio inputs from participants **1**-**4**.

Referring to FIG. **5**, each stereo conference output stream **104** includes audio inputs or sources **120** from the other participants or groups of participants that are perceived by the listener **122** as coming from different spatial locations. The spatial locations vary from front to back in the listener's depth perception and from left to right in the listener's lateral directivity. Because the sound sources are spatially separated, the listener **122** can more easily focus on individual sound sources of auditory information in the presence of

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other sound sources. Thus, the spatial separation of the sound sources **120** increases the ability of the listener **122** to differentiate between the multiple sound sources **120**.

In the illustrated embodiment, each participant **1**-**4** is spatially positioned in front and at an equal distance from the participant **5**. In this configuration, each participant **1**-**4** has an equal degree or substantial degree of prominence with respect to the participant **5**. As described in more detail below, participants **1**-**4** in the stereo conference output stream **104** may be repositioned to the foreground to provide a higher degree of intelligibility and prominence to participant **5**.

Referring to FIG. **6**, the output stream **104** for participant **5**, for example, includes the audio input of moderator **3** in the foreground with the other participants **1**, **2** and **4** in the background. The foreground position provides participant **5** or other listener **160** with the highest degree of intelligibility such that the listener may focus on moderator **3** or other moderator audio sources **162** while still hearing non-moderator sources **164** in the background. It will be understood that moderator **162** may be otherwise suitably positioned in the output stream(s) **104**, without departing from the scope of the present invention.

Referring to FIG. **7**, the directional processor **106** of the stereo mixer **100** includes a plurality of spatial processors **180** and the summer **108** includes a left and right channel summers **182**. The spatial processors **180** each present monaural sources at different locations in a binaural sound field using standard intensity panning and/or Head Related Transfer Function (HRTF) position filtering. The binaural sound streams each include left and right channels components **184** generating a perceived position such as, for example, back/left front/center and back/right. The left channel of each binaural stream is fed to the left channel summer **182** while the right channels are fed to the right channel summer **182**. The summers **182** generate a combined left stream **186** and combined right stream **188** including a perceived plurality of discrete audio inputs spatially positioned in two or three dimensional space relative to the listener. Further information regarding the directional processor **106** and summer **108** are provided in co-owned U.S. Pat. No. 6,011,851, which is hereby incorporated by reference. It will be understood that stereo mixer **100** and sidebar mixers **140** may be otherwise suitably implemented without departing from the scope of the present invention.

FIG. **8** illustrates a method for improving the intelligibility of a moderator during a multi-party communication session, in accordance with a particular embodiment of the present invention. The method begins at step **200** where a multi-party communication session is established. At step **202**, a priority of one or more participants is identified. For example, at least one participant may be given the priority of moderator for a particular conference call. Typically, the moderator will receive the highest priority in a conference call. Therefore, in accordance with one embodiment, the priorities simply include moderator for one or more parties, and participant for the remaining parties. However, other participants may be prioritized even further. For example, participants may be prioritized according to groups in which one or more participants will have "high priority", and other participants will have "low priority". In still another embodiment, each participant may be prioritized with respect to the others, such that each participant in the conference call has a different priority from all other participants.

The priority of a particular participant may be assigned, in advance, by a coordinator of the specific conference call. In

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Exhibit 1006 – 009

Ex. 1006 at 1:54-55, 8:49-57; *see also* IPR2023-0378, -379, Ex. 1003 ¶¶ 150-153; *see also* IPR2023-00378, Pet. at 18, 29 & Reply at 7; IPR2023-00379, Pet. at 17, 23, 58 & Reply at 13-14.

A POSITA WOULD HAVE BEEN MOTIVATED TO COMBINE DRELL WITH KNAPPE

> 152. Accordingly, it would have been obvious to one of ordinary skill in the art for at least one of Drell's conference participants, such as the user of the remote conference endpoint or near conference endpoint, to be a moderator per Knappe's teaching, because doing so would beneficially allow the user (acting as the moderator) to initiate, organize, control, and influence the conference. Furthermore, as there are a finite number of participants in a conference, it would have been obvious to try designating one of Drell's conference endpoint users as the moderator. The combination of Drell and Knappe is merely the ordinary use of a common technique (identifying the user of the conference endpoint as a moderator per Knappe) in the similar conferencing system of Drell to yield predictable results of initiating, organizing, controlling, and influencing the conference. Each of these reasons separately and together would have motivated a POSITA.

IPR2023-0378, -379, Ex. 1003 ¶ 152; *see also id.* ¶¶ 150-153.

IPR2023-00378, Pet. at 18, 29-30 & Reply at 7; IPR2023-00379, Pet. at 17, 23-24, 58-59 & Reply at 13-14.

The Prior Art Renders Obvious the "Encoder" and "VoIP Encoder/Decoder" Limitations

IPR2023-0378, Grounds 1-4
(U.S. Patent No. 9,094,525)

IPR2023-0379, Ground 1
(U.S. Patent No. 9,094,525)

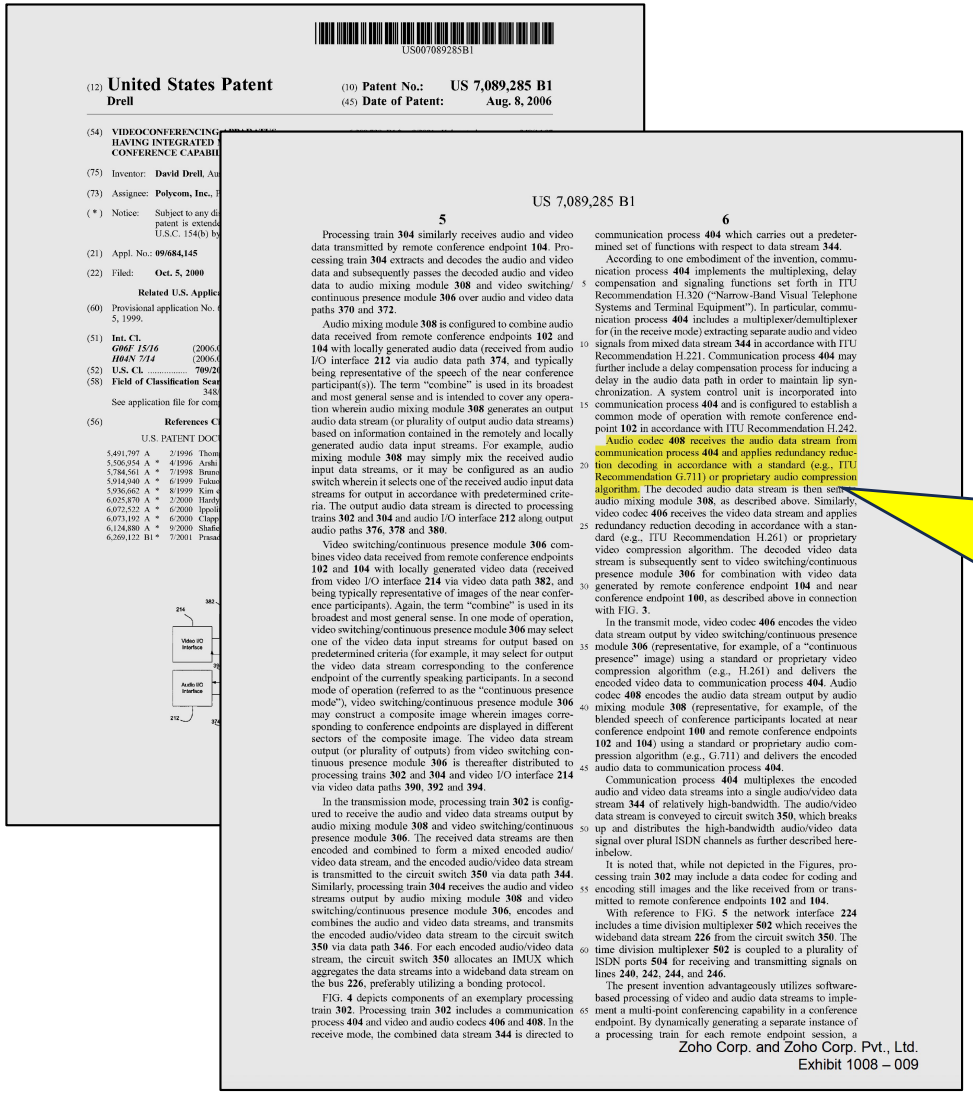
THE "ENCODER" AND "VOIP ENCODER/DECODER" LIMITATIONS

1) The '525 patent, dependent claims 3, 6, 15-16, 17: (IPR2023-00378, Grounds 1-4; IPR2023-00379, Ground 1)

- Claim 3: "**a VoIP decoder** that receives IP packets . . . **the VoIP decoder decodes the IP packets** including the audio data"
- Claim 6: "**a VoIP encoder that encodes** the second mixed data stream and encodes the second mixed voice data before transmitting"
- Claim 15: "**encoding** the mixed audio-video data before the transmitting"
- Claim 16: "encoding the compressed mixed audio data before the transmitting"
- Claim 17: "**a voice over IP decoder** configured to decode audio data received by the local moderator computer . . . **a voice over IP encoder** configured to compress audio data transported from the local moderator computer"

IPR2023-00378, -379, Ex. 1001 at 10:23-26, 10:33-36, 11:24-30, 11:33-12:3.

DRELL TEACHES USE OF CODECS



US007089285B1

(12) **United States Patent**
Drell

(10) **Patent No.: US 7,089,285 B1**
(45) **Date of Patent: Aug. 8, 2006**

(54) VIDEOCONFERENCING ... HAVING INTEGRATED ... CONFERENCE CAPABI...

(75) Inventor: **David Drell**, Au...

(73) Assignee: **Polycom, Inc.**, ...

(*) Notice: Subject to any di... patent is extend... U.S.C. 154(b) by...

(21) Appl. No.: **09/684,145**

(22) Filed: **Oct. 5, 2000**

Related U.S. Applic...

(60) Provisional application No. ... 5, 1999.

(51) **Int. Cl.**
G06F 15/16 (2006.0...
H04N 7/14 (2006.0...

(52) **U.S. Cl.** ..... **709/20...**

(58) **Field of Classification Sear...** 348/...
See application file for com...

(56) **References Cl...**

U.S. PATENT DOC...

5,491,797	A		2/1996	Thom...
5,506,954	A	*	4/1996	Arshi...
5,784,561	A	*	7/1998	Bruno...
5,914,940	A	*	6/1999	Fukuo...
5,936,662	A	*	8/1999	Kim e...
6,025,870	A	*	2/2000	Hardy...
6,072,522	A	*	6/2000	Ippoli...
6,073,192	A	*	6/2000	Clapp...
6,124,880	A	*	9/2000	Shafe...
6,269,122	B1	*	7/2001	Prasad...

Processing train **304** similarly receives audio and video data transmitted by remote conference endpoint **104**. Processing train **304** extracts and decodes the audio and video data and subsequently passes the decoded audio and video data to audio mixing module **308** and video switching/continuous presence module **306** over audio and video data paths **370** and **372**.

Audio mixing module **308** is configured to combine audio data received from remote conference endpoints **102** and **104** with locally generated audio data (received from audio I/O interface **212** via audio data path **374**, and typically being representative of the speech of the near conference participant(s)). The term "combine" is used in its broadest and most general sense and is intended to cover any operation wherein audio mixing module **308** generates an output audio data stream (or plurality of output audio data streams) based on information contained in the remotely and locally generated audio data input streams. For example, audio mixing module **308** may simply mix the received audio input data streams, or it may be configured as an audio switch wherein it selects one of the received audio input data streams for output in accordance with predetermined criteria. The output audio data stream is directed to processing trains **302** and **304** and audio I/O interface **212** along output audio paths **376**, **378** and **380**.

Video switching/continuous presence module **306** combines video data received from remote conference endpoints **102** and **104** with locally generated video data (received from video I/O interface **214** via video data path **382**, and being typically representative of images of the near conference participants). Again, the term "combine" is used in its broadest and most general sense. In one mode of operation, video switching/continuous presence module **306** may select one of the video data input streams for output based on predetermined criteria (for example, it may select for output the video data stream corresponding to the conference endpoint of the currently speaking participants. In a second mode of operation (referred to as the "continuous presence mode"), video switching/continuous presence module **306** may construct a composite image wherein images corresponding to conference endpoints are displayed in different sectors of the composite image. The video data stream output (or plurality of outputs) from video switching continuous presence module **306** is thereafter distributed to processing trains **302** and **304** and video I/O interface **214** via video data paths **390**, **392** and **394**.

In the transmission mode, processing train **302** is configured to receive the audio and video data streams output by audio mixing module **308** and video switching/continuous presence module **306**. The received data streams are then encoded and combined to form a mixed encoded audio/video data stream, and the encoded audio/video data stream is transmitted to the circuit switch **350** via data path **344**. Similarly, processing train **304** receives the audio and video streams output by audio mixing module **308** and video switching/continuous presence module **306**, encodes and combines the audio and video data streams, and transmits the encoded audio/video data stream to the circuit switch **350** via data path **346**. For each encoded audio/video data stream, the circuit switch **350** allocates an IMUX which aggregates the data streams into a wideband data stream on the bus **226**, preferably utilizing a bonding protocol.

FIG. **4** depicts components of an exemplary processing train **302**. Processing train **302** includes a communication process **404** and video and audio codecs **406** and **408**. In the receive mode, the combined data stream **344** is directed to

communication process **404** which carries out a predetermined set of functions with respect to data stream **344**.

According to one embodiment of the invention, communication process **404** implements the multiplexing, delay compensation and signaling functions set forth in ITU Recommendation H.320 ("Narrow-Band Visual Telephone Systems and Terminal Equipment"). In particular, communication process **404** includes a multiplexer/demultiplexer for (in the receive mode) extracting separate audio and video signals from mixed data stream **344** in accordance with ITU Recommendation H.221. Communication process **404** may further include a delay compensation process for inducing a delay in the audio data path in order to maintain lip synchronization. A system control unit is incorporated into communication process **404** and is configured to establish a common mode of operation with remote conference endpoint **102** in accordance with ITU Recommendation H.242.

Audio codec **408** receives the audio data stream from communication process **404** and applies redundancy reduction decoding in accordance with a standard (e.g., ITU Recommendation G.711) or proprietary audio compression algorithm. The decoded audio data stream is then sent to audio mixing module **308**, as described above. Similarly, video codec **406** receives the video data stream and applies redundancy reduction decoding in accordance with a standard (e.g., ITU Recommendation H.261) or proprietary video compression algorithm. The decoded video data stream is subsequently sent to video switching/continuous presence module **306** for combination with video data generated by remote conference endpoint **104** and near conference endpoint **100**, as described above in connection with FIG. **3**.

In the transmit mode, video codec **406** encodes the video data stream output by video switching/continuous presence module **306** (representative, for example, of a "continuous presence" image) using a standard or proprietary video compression algorithm (e.g., H.261) and delivers the encoded video data to communication process **404**. Audio codec **408** encodes the audio data stream output by audio mixing module **308** (representative, for example, of the blended speech of conference participants located at near conference endpoint **100** and remote conference endpoints **102** and **104**) using a standard or proprietary audio compression algorithm (e.g., G.711) and delivers the encoded audio data to communication process **404**.

Communication process **404** multiplexes the encoded audio and video data streams into a single audio/video data stream **344** of relatively high-bandwidth. The audio/video data stream is conveyed to circuit switch **350**, which breaks up and distributes the high-bandwidth audio/video data signal over plural ISDN channels as further described hereinbelow.

It is noted that, while not depicted in the Figures, processing train **302** may include a data codec for coding and encoding still images and the like received from or transmitted to remote conference endpoints **102** and **104**.

With reference to FIG. **5** the network interface **224** includes a time division multiplexer **502** which receives the wideband data stream **226** from the circuit switch **350**. The time division multiplexer **502** is coupled to a plurality of ISDN ports **504** for receiving and transmitting signals on lines **240**, **242**, **244**, and **246**.

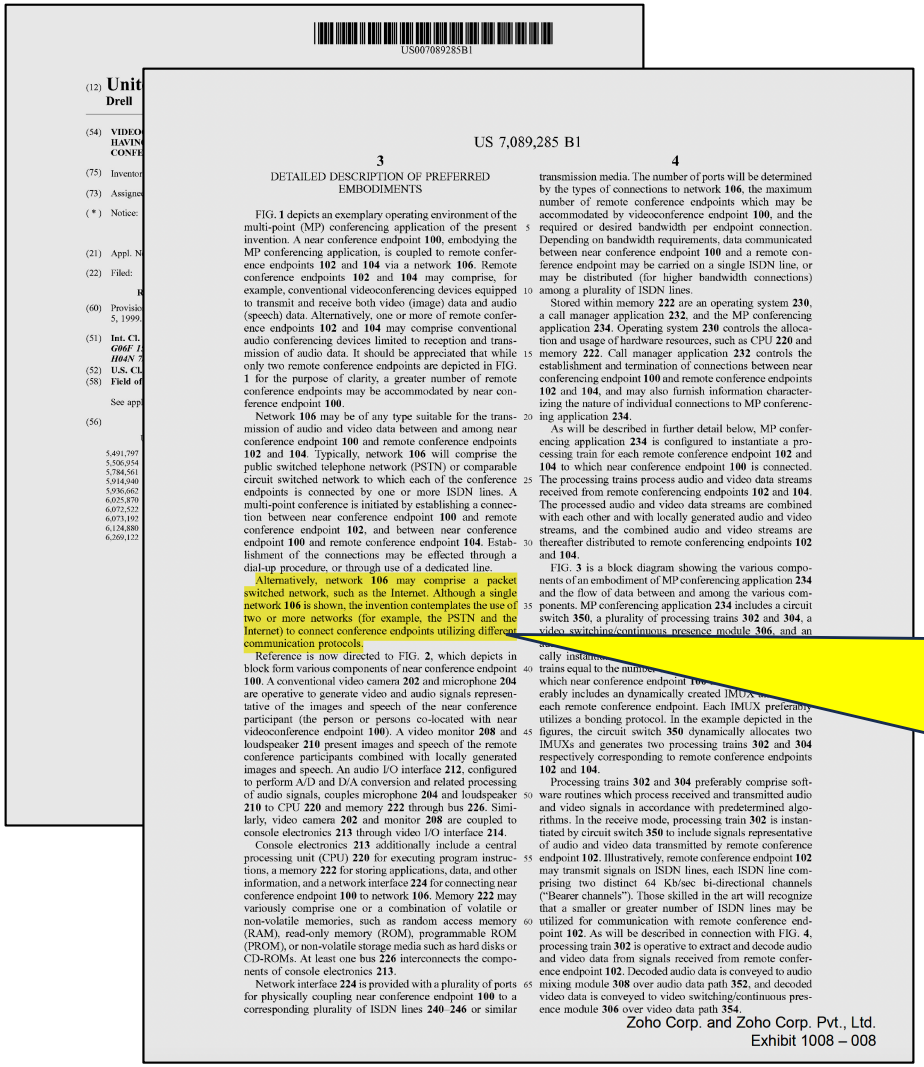
The present invention advantageously utilizes software-based processing of video and audio data streams to implement a multi-point conferencing capability in a conference endpoint. By dynamically generating a separate instance of a processing train for each remote endpoint session, a

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Exhibit 1008 – 009

Audio codec 408 receives the audio data stream from communication process 404 and applies redundancy reduction **decoding in accordance with a standard** (e.g. ITU Recommendation G.711) or proprietary audio compression algorithm.

Ex. 1008 at 6:18-45; *see also* IPR2023-0378, -379, Ex. 1003 ¶¶ 133-134, 277-279; IPR2023-00378, Pet. at 51-52, 63-64 & Reply at 8; IPR2023-00379, Pet. at 59-61 & Reply at 11.

DRELL TEACHES USE OF CODECS IN CONJUNCTION WITH TRANSMISSION OF DATA OVER THE INTERNET



US007089285B1

(12) Unit
Drell

(54) VIDEO
HAVIN
CONFE

(75) Inventor

(73) Assigne

(*) Notice:

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G06F 1
H04N 7

(52) U.S. Cl.

(58) Field of

See app

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5,491,797
5,506,954
5,784,561
5,914,940
5,936,662
6,025,870
6,072,522
6,073,192
6,124,880
6,269,122

US 7,089,285 B1

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DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

FIG. **1** depicts an exemplary operating environment of the multi-point (MP) conferencing application of the present invention. A near conference endpoint **100**, embodying the MP conferencing application, is coupled to remote conference endpoints **102** and **104** via a network **106**. Remote conference endpoints **102** and **104** may comprise, for example, conventional videoconferencing devices equipped to transmit and receive both video (image) data and audio (speech) data. Alternatively, one or more of remote conference endpoints **102** and **104** may comprise conventional audio conferencing devices limited to reception and transmission of audio data. It should be appreciated that while only two remote conference endpoints are depicted in FIG. **1** for the purpose of clarity, a greater number of remote conference endpoints may be accommodated by near conference endpoint **100**.

Network **106** may be of any type suitable for the transmission of audio and video data between and among near conference endpoint **100** and remote conference endpoints **102** and **104**. Typically, network **106** will comprise the public switched telephone network (PSTN) or comparable circuit switched network to which each of the conference endpoints is connected by one or more ISDN lines. A multi-point conference is initiated by establishing a connection between near conference endpoint **100** and remote conference endpoint **102**, and between near conference endpoint **100** and remote conference endpoint **104**. Establishment of the connections may be effected through a dial-up procedure, or through use of a dedicated line.

Alternatively, network **106** may comprise a packet switched network, such as the Internet. Although a single network **106** is shown, the invention contemplates the use of two or more networks (for example, the PSTN and the Internet) to connect conference endpoints utilizing different communication protocols.

Reference is now directed to FIG. **2**, which depicts in block form various components of near conference endpoint **100**. A conventional video camera **202** and microphone **204** are operative to generate video and audio signals representative of the images and speech of the near conference participant (the person or persons co-located with near videoconference endpoint **100**). A video monitor **208** and loudspeaker **210** present images and speech of the remote conference participants combined with locally generated images and speech. An audio I/O interface **212**, configured to perform A/D and D/A conversion and related processing of audio signals, couples microphone **204** and loudspeaker **210** to CPU **220** and memory **222** through bus **226**. Similarly, video camera **202** and monitor **208** are coupled to console electronics **213** through video I/O interface **214**.

Console electronics **213** additionally include a central processing unit (CPU) **220** for executing program instructions, a memory **222** for storing applications, data, and other information, and a network interface **224** for connecting near conference endpoint **100** to network **106**. Memory **222** may variously comprise one or a combination of volatile or non-volatile memories, such as random access memory (RAM), read-only memory (ROM), programmable ROM (PROM), or non-volatile storage media such as hard disks or CD-ROMs. At least one bus **226** interconnects the components of console electronics **213**.

Network interface **224** is provided with a plurality of ports for physically coupling near conference endpoint **100** to a corresponding plurality of ISDN lines **240**–**246** or similar

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transmission media. The number of ports will be determined by the types of connections to network **106**, the maximum number of remote conference endpoints which may be accommodated by videoconference endpoint **100**, and the required or desired bandwidth per endpoint connection. Depending on bandwidth requirements, data communicated between near conference endpoint **100** and a remote conference endpoint may be carried on a single ISDN line, or may be distributed (for higher bandwidth connections) among a plurality of ISDN lines.

Stored within memory **222** are an operating system **230**, a call manager application **232**, and the MP conferencing application **234**. Operating system **230** controls the allocation and usage of hardware resources, such as CPU **220** and memory **222**. Call manager application **232** controls the establishment and termination of connections between near conferencing endpoint **100** and remote conference endpoints **102** and **104**, and may also furnish information characterizing the nature of individual connections to MP conferencing application **234**.

As will be described in further detail below, MP conferencing application **234** is configured to instantiate a processing train for each remote conference endpoint **102** and **104** to which near conference endpoint **100** is connected. The processing trains process audio and video data streams received from remote conferencing endpoints **102** and **104**. The processed audio and video data streams are combined with each other and with locally generated audio and video streams, and the combined audio and video streams are thereafter distributed to remote conferencing endpoints **102** and **104**.

FIG. **3** is a block diagram showing the various components of an embodiment of MP conferencing application **234** and the flow of data between and among the various components. MP conferencing application **234** includes a circuit switch **350**, a plurality of processing trains **302** and **304**, a video switching/continuous presence module **306**, and an

cally instan
trains equal to the numb
which near conference endpoint
erably includes an dynamically created IMUX
each remote conference endpoint. Each IMUX preferably
utilizes a bonding protocol. In the example depicted in the
figures, the circuit switch **350** dynamically allocates two
IMUXs and generates two processing trains **302** and **304**
respectively corresponding to remote conference endpoints
102 and **104**.

Processing trains **302** and **304** preferably comprise software routines which process received and transmitted audio and video signals in accordance with predetermined algorithms. In the receive mode, processing train **302** is instantiated by circuit switch **350** to include signals representative of audio and video data transmitted by remote conference endpoint **102**. Illustratively, remote conference endpoint **102** may transmit signals on ISDN lines, each ISDN line comprising two distinct 64 Kb/sec bi-directional channels ("Bearer channels"). Those skilled in the art will recognize that a smaller or greater number of ISDN lines may be utilized for communication with remote conference endpoint **102**. As will be described in connection with FIG. **4**, processing train **302** is operative to extract and decode audio and video data from signals received from remote conference endpoint **102**. Decoded audio data is conveyed to audio mixing module **308** over audio data path **352**, and decoded video data is conveyed to video switching/continuous presence module **306** over video data path **354**.

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Exhibit 1008 – 008

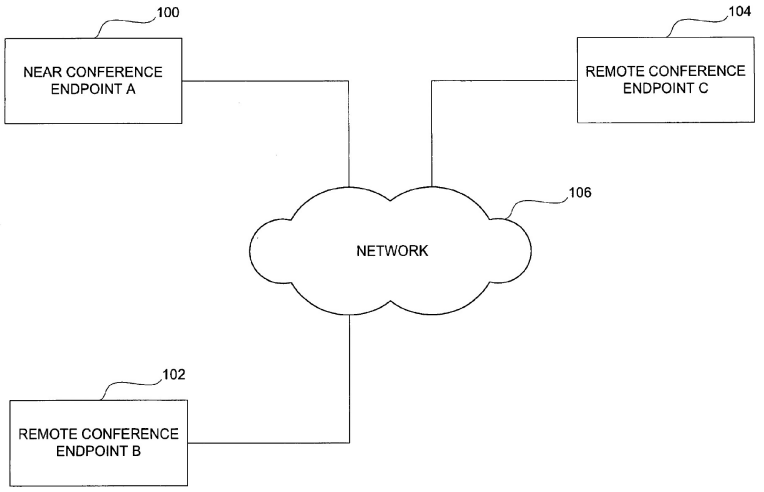


FIG. 1

[N]etwork 106 may comprise a **packet switched network, such as the Internet**. Although a single network 106 is shown, the invention contemplates the use of two or more networks (for example, the PSN and the Internet) to connect conference endpoints utilizing different communication protocols.

Ex. 1008 at 3:33-38, FIG. 1; *see also* IPR2023-0378, -379 Ex. 1003 ¶ 278; IPR2023-00378, Pet. at 51-52, 63-64 & Reply at 8-9; IPR2023-00379, Pet. at 59-61 & Reply at 11.

DRELL TEACHES USE OF CODECS IN CONJUNCTION WITH TRANSMISSION OF DATA OVER THE INTERNET

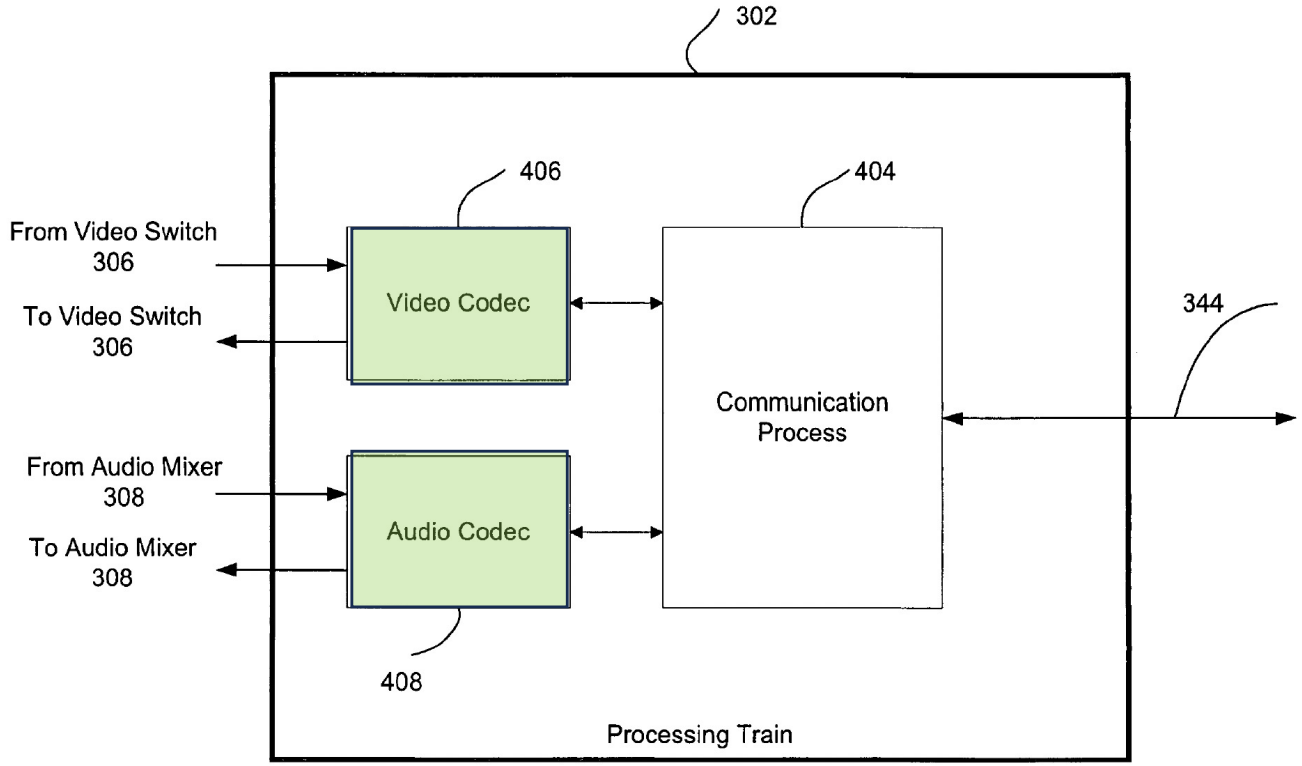
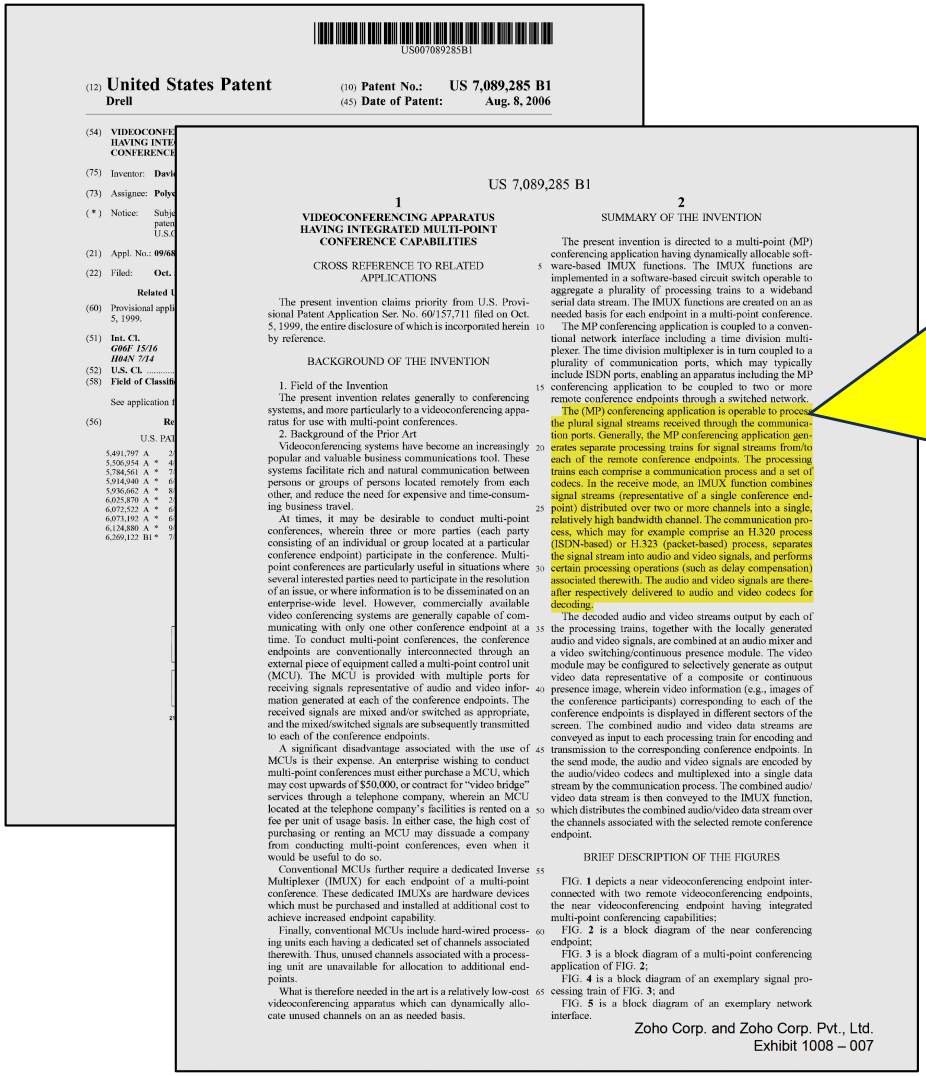


FIG. 4

Ex. 1008 at FIG. 4 (annotated); *see also* IPR2023-0378, -379 Ex. 1003 ¶¶ 133, 278; IPR2023-00378, Pet. at 51-52, 63-64 & Reply at 8; IPR2023-00379, Pet. at 59-61 & Reply at 11.

DRELL TEACHES USE OF CODECS IN CONJUNCTION WITH TRANSMISSION OF DATA OVER THE INTERNET



US007089285B1

(12) **United States Patent**
Drell

(10) **Patent No.: US 7,089,285 B1**
(45) **Date of Patent: Aug. 8, 2006**

US 7,089,285 B1

1

VIDEOCONFERENCING APPARATUS HAVING INTEGRATED MULTI-POINT CONFERENCE CAPABILITIES

CROSS REFERENCE TO RELATED APPLICATIONS

The present invention claims priority from U.S. Provisional Patent Application Ser. No. 60/157,711 filed on Oct. 5, 1999, the entire disclosure of which is incorporated herein by reference.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates generally to conferencing systems, and more particularly to a videoconferencing apparatus for use with multi-point conferences.

2. Background of the Prior Art

Videoconferencing systems have become an increasingly popular and valuable business communications tool. These systems facilitate rich and natural communication between persons or groups of persons located remotely from each other, and reduce the need for expensive and time-consuming business travel.

At times, it may be desirable to conduct multi-point conferences, wherein three or more parties (each party consisting of an individual or group located at a particular conference endpoint) participate in the conference. Multi-point conferences are particularly useful in situations where several interested parties need to participate in the resolution of an issue, or where information is to be disseminated on an enterprise-wide level. However, commercially available video conferencing systems are generally capable of communicating with only one other conference endpoint at a time. To conduct multi-point conferences, the conference endpoints are conventionally interconnected through an external piece of equipment called a multi-point control unit (MCU). The MCU is provided with multiple ports for receiving signals representative of audio and video information generated at each of the conference endpoints. The received signals are mixed and/or switched as appropriate, and the mixed/switched signals are subsequently transmitted to each of the conference endpoints.

A significant disadvantage associated with the use of MCUs is their expense. An enterprise wishing to conduct multi-point conferences must either purchase a MCU, which may cost upwards of $50,000, or contract for "video bridge" services through a telephone company, wherein an MCU located at the telephone company's facilities is rented on a fee per unit of usage basis. In either case, the high cost of purchasing or renting an MCU may dissuade a company from conducting multi-point conferences, even when it would be useful to do so.

Conventional MCUs further require a dedicated Inverse Multiplexer (IMUX) for each endpoint of a multi-point conference. These dedicated IMUXs are hardware devices which must be purchased and installed at additional cost to achieve increased endpoint capability.

Finally, conventional MCUs include hard-wired processing units each having a dedicated set of channels associated therewith. Thus, unused channels associated with a processing unit are unavailable for allocation to additional endpoints.

What is therefore needed in the art is a relatively low-cost videoconferencing apparatus which can dynamically allocate unused channels on an as needed basis.

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SUMMARY OF THE INVENTION

The present invention is directed to a multi-point (MP) conferencing application having dynamically allocable software-based IMUX functions. The IMUX functions are implemented in a software-based circuit switch operable to aggregate a plurality of processing trains to a wideband serial data stream. The IMUX functions are created on an as needed basis for each endpoint in a multi-point conference.

The MP conferencing application is coupled to a conventional network interface including a time division multiplexer. The time division multiplexer is in turn coupled to a plurality of communication ports, which may typically include ISDN ports, enabling an apparatus including the MP conferencing application to be coupled to two or more remote conference endpoints through a switched network.

The (MP) conferencing application is operable to process the plural signal streams received through the communication ports. Generally, the MP conferencing application generates separate processing trains for signal streams from/to each of the remote conference endpoints. The processing trains each comprise a communication process and a set of codecs. In the receive mode, an IMUX function combines signal streams (representative of a single conference endpoint) distributed over two or more channels into a single, relatively high bandwidth channel. The communication process, which may for example comprise an H.320 process (ISDN-based) or H.323 (packet-based) process, separates the signal stream into audio and video signals, and performs certain processing operations (such as delay compensation) associated therewith. The audio and video signals are thereafter respectively delivered to audio and video codecs for decoding.

The decoded audio and video streams output by each of the processing trains, together with the locally generated audio and video signals, are combined at an audio mixer and a video switching/continuous presence module. The video module may be configured to selectively generate as output video data representative of a composite or continuous presence image, wherein video information (e.g., images of the conference participants) corresponding to each of the conference endpoints is displayed in different sectors of the screen. The combined audio and video data streams are conveyed as input to each processing train for encoding and transmission to the corresponding conference endpoints. In the send mode, the audio and video signals are encoded by the audio/video codecs and multiplexed into a single data stream by the communication process. The combined audio/video data stream is then conveyed to the IMUX function, which distributes the combined audio/video data stream over the channels associated with the selected remote conference endpoint.

BRIEF DESCRIPTION OF THE FIGURES

FIG. **1** depicts a near videoconferencing endpoint interconnected with two remote videoconferencing endpoints, the near videoconferencing endpoint having integrated multi-point conferencing capabilities;

FIG. **2** is a block diagram of the near conferencing endpoint;

FIG. **3** is a block diagram of a multi-point conferencing application of FIG. **2**;

FIG. **4** is a block diagram of an exemplary signal processing train of FIG. **3**; and

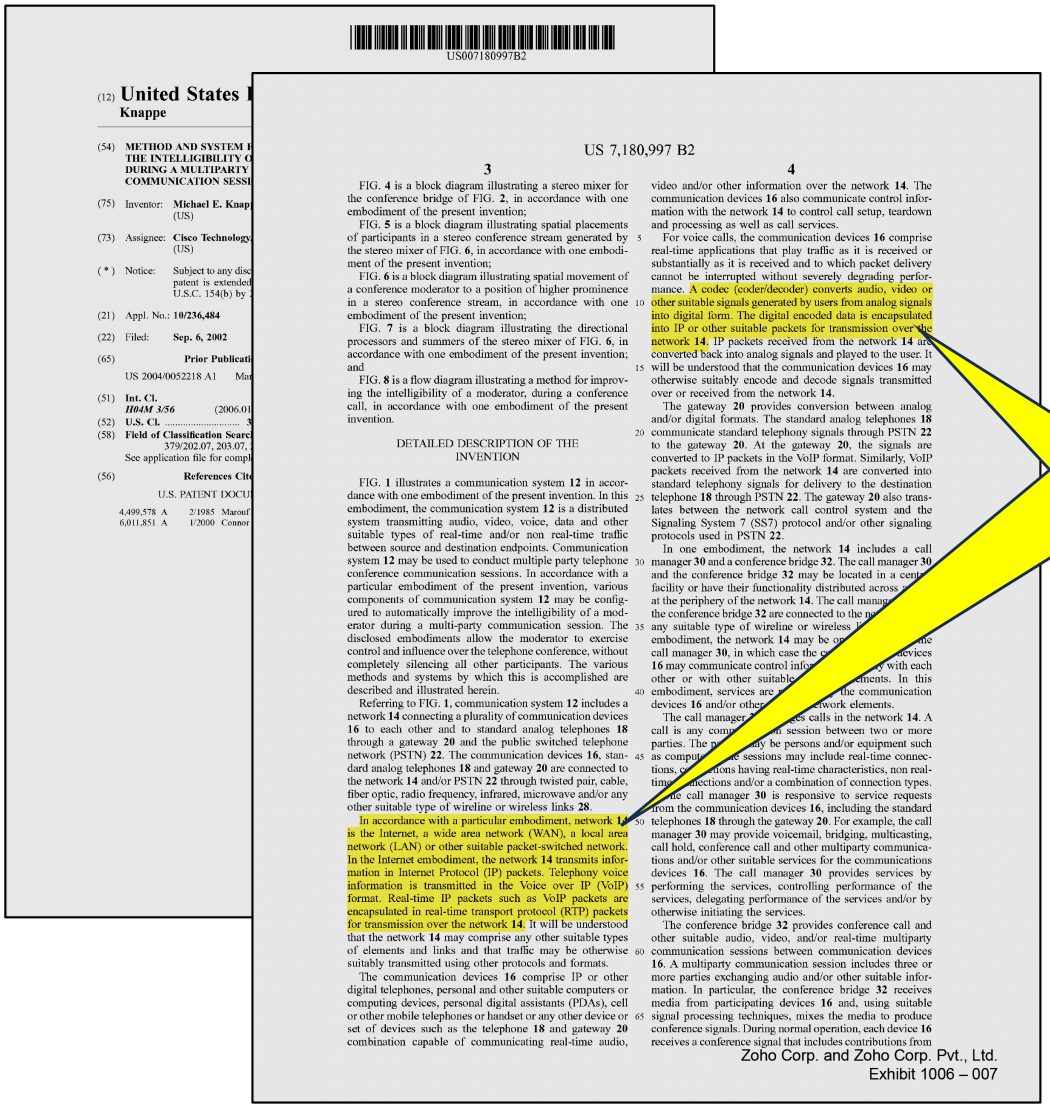
FIG. **5** is a block diagram of an exemplary network interface.

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Exhibit 1008 – 007

The (MP) conferencing application is operable to process the plural signal streams received through the communication ports. Generally, the MP conferencing application generates separate processing trains for **signal streams from/to each of the remote conference endpoints**. The processing trains each comprise a communication **process and a set of codecs** . . . The communication process . . . may for example comprise an **H.323 (packet-based) process** . . . The audio and video signals are . . . respectively delivered to audio and video codecs for decoding.

Ex. 1008 at 2:17-33; *see also* IPR2023-0378, -379 Ex. 1003 ¶¶ 280-281; IPR2023-00378, Pet. at 51-52, 63-64 & Reply at 8; IPR2023-00379, Pet. at 59-61 & Reply at 11.

KNAPPE TEACHES USE OF CODEC (CODER/DECODER) FOR VOIP PACKETS



US007180997B2

(12) **United States**
Knappe

(54) METHOD AND SYSTEM
THE INTELLIGIBILITY
DURING A MULTIPARTY
COMMUNICATION SESSI

(75) Inventor: **Michael E. Knap** (US)

(73) Assignee: **Cisco Technology** (US)

(*) Notice: Subject to any disc
patent is extended
U.S.C. 154(b) by

(21) Appl. No.: **10/236,484**

(22) Filed: **Sep. 6, 2002**

(65) **Prior Publicati**

US 2004/0052218 A1 Mar

(51) **Int. Cl.**
H04M 3/56 (2006.01

(52) **U.S. Cl.** .....

(58) **Field of Classification Searc**
379/202.07, 203.07,
See application file for compl

(56) **References Cite**

U.S. PATENT DOCU

4,499,578 A 2/1985 Maroui
6,011,851 A 1/2000 Connor

US 7,180,997 B2

3

FIG. **4** is a block diagram illustrating a stereo mixer for the conference bridge of FIG. **2**, in accordance with one embodiment of the present invention;

FIG. **5** is a block diagram illustrating spatial placements of participants in a stereo conference stream generated by the stereo mixer of FIG. **6**, in accordance with one embodiment of the present invention;

FIG. **6** is a block diagram illustrating spatial movement of a conference moderator to a position of higher prominence in a stereo conference stream, in accordance with one embodiment of the present invention;

FIG. **7** is a block diagram illustrating the directional processors and summers of the stereo mixer of FIG. **6**, in accordance with one embodiment of the present invention; and

FIG. **8** is a flow diagram illustrating a method for improving the intelligibility of a moderator, during a conference call, in accordance with one embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

FIG. **1** illustrates a communication system **12** in accordance with one embodiment of the present invention. In this embodiment, the communication system **12** is a distributed system transmitting audio, video, voice, data and other suitable types of real-time and/or non real-time traffic between source and destination endpoints. Communication system **12** may be used to conduct multiple party telephone conference communication sessions. In accordance with a particular embodiment of the present invention, various components of communication system **12** may be configured to automatically improve the intelligibility of a moderator during a multi-party communication session. The disclosed embodiments allow the moderator to exercise control and influence over the telephone conference, without completely silencing all other participants. The various methods and systems by which this is accomplished are described and illustrated herein.

Referring to FIG. **1**, communication system **12** includes a network **14** connecting a plurality of communication devices **16** to each other and to standard analog telephones **18** through a gateway **20** and the public switched telephone network (PSTN) **22**. The communication devices **16**, standard analog telephones **18** and gateway **20** are connected to the network **14** and/or PSTN **22** through twisted pair, cable, fiber optic, radio frequency, infrared, microwave and/or any other suitable type of wireline or wireless links **28**.

In accordance with a particular embodiment, network **14** is the Internet, a wide area network (WAN), a local area network (LAN) or other suitable packet-switched network. In the Internet embodiment, the network **14** transmits information in Internet Protocol (IP) packets. Telephony voice information is transmitted in the Voice over IP (VoIP) format. Real-time IP packets such as VoIP packets are encapsulated in real-time transport protocol (RTP) packets for transmission over the network **14**. It will be understood that the network **14** may comprise any other suitable types of elements and links and that traffic may be otherwise suitably transmitted using other protocols and formats.

The communication devices **16** comprise IP or other digital telephones, personal and other suitable computers or computing devices, personal digital assistants (PDAs), cell or other mobile telephones or handset or any other device or set of devices such as the telephone **18** and gateway **20** combination capable of communicating real-time audio,

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video and/or other information over the network **14**. The communication devices **16** also communicate control information with the network **14** to control call setup, teardown and processing as well as call services.

For voice calls, the communication devices **16** comprise real-time applications that play traffic as it is received or substantially as it is received and to which packet delivery cannot be interrupted without severely degrading performance. A codec (coder/decoder) converts audio, video or other suitable signals generated by users from analog signals into digital form. The digital encoded data is encapsulated into IP or other suitable packets for transmission over the network **14**. IP packets received from the network **14** are converted back into analog signals and played to the user. It will be understood that the communication devices **16** may otherwise suitably encode and decode signals transmitted over or received from the network **14**.

The gateway **20** provides conversion between analog and/or digital formats. The standard analog telephones **18** communicate standard telephony signals through PSTN **22** to the gateway **20**. At the gateway **20**, the signals are converted to IP packets in the VoIP format. Similarly, VoIP packets received from the network **14** are converted into standard telephony signals for delivery to the destination telephone **18** through PSTN **22**. The gateway **20** also translates between the network call control system and the Signaling System 7 (SS7) protocol and/or other signaling protocols used in PSTN **22**.

In one embodiment, the network **14** includes a call manager **30** and a conference bridge **32**. The call manager **30** and the conference bridge **32** may be located in a cent facility or have their functionality distributed across at the periphery of the network **14**. The call manag the conference bridge **32** are connected to the n any suitable type of wireline or wireless l embodiment, the network **14** may be o call manager **30**, in which case the c devices **16** may communicate control infor with each other or with other suitable ements. In this embodiment, services are the communication devices **16** and/or othe twork elements.

The call manager sses calls in the network **14**. A call is any comm session between two or more parties. The y be persons and/or equipment such as comput s sessions may include real-time connections, c tions having real-time characteristics, non real-time nections and/or a combination of connection types. The call manager **30** is responsive to service requests from the communication devices **16**, including the standard telephones **18** through the gateway **20**. For example, the call manager **30** may provide voicemail, bridging, multicasting, call hold, conference call and other multiparty communications and/or other suitable services for the communications devices **16**. The call manager **30** provides services by performing the services, controlling performance of the services, delegating performance of the services and/or by otherwise initiating the services.

The conference bridge **32** provides conference call and other suitable audio, video, and/or real-time multiparty communication sessions between communication devices **16**. A multiparty communication session includes three or more parties exchanging audio and/or other suitable information. In particular, the conference bridge **32** receives media from participating devices **16** and, using suitable signal processing techniques, mixes the media to produce conference signals. During normal operation, each device **16** receives a conference signal that includes contributions from

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Exhibit 1006 – 007

[N]etwork 14 is the Internet . . . [which communicates using] Real-time **IP packets such as VoIP packets** . . . encapsulated in real-time transport protocol (RTP) packets.

A **codec (coder/decoder) converts audio, video** or other suitable **signals** generated by users from analog signals to digital form. **The digital encoded data is encapsulated into IP** or other suitable packets from transmission over the network 14

Ex. 1006 at 3:50-58, 4:9-14; *see also* IPR2023-0378, -379, Ex. 1003 ¶¶ 282-283; IPR2023-00378, Pet. at 51-52, 63-64 & Reply at 8; IPR2023-00379, Pet. at 59-61 & Reply at 11.

IPR2023-00371

The Prior Art Renders Obvious Claims 1-10 of the '997 Patent

INSTITUTION AND ISSUES

- Petitioner has demonstrated a reasonable likelihood that it will succeed on challenges to the patentability of :
  - Claims 1 and 3-7 as obvious under 35 U.S.C. § 103(a)over Knappe in view of Elliott, the VPN Textbook, and de Moraes
  - Claims 2 and 8-10 as obvious under 35 U.S.C. § 103(a) over Knappe in view of Elliott, the VPN Textbook, de Moraes, and Shepherd

I.D. at 42

- Patent Owner raised limited issues:
  - Claim Construction & VPN Textbook
  - Ground 1: Claims 1 and 7
  - Ground 2: Claim 2

POR at i-iii

GROUND 1: CLAIMS 1 AND 7

- Claim 1
  - Limitation 1.1
  - Limitations 1.5a & 1.5c
  - Limitation 1.5b
  - Limitation 1.5d
  - Limitation 1.6
- Claim 7
  - Limitation 7.2

RESPONSE TO LIMITATION [1.1] ARGUMENTS

[1.1] sending a first message to each of a plurality of multicast appliances over the Internet, wherein the first message comprises a group address which identifies a plurality of participants;

Pet. at 37

NEITHER KNAPPE NOR DE MORAES DISCLOSE THE "SENDING A FIRST MESSAGE TO EACH OF A PLURALITY OF MULTICAST APPLIANCES OVER THE INTERNET, WHEREIN THE FIRST MESSAGE COMPRISES A GROUP ADDRESS WHICH IDENTIFIES A PLURALITY OF PARTICIPANTS;" LIMITATION OF INDEPENDENT CLAIM 1 (*PETITIONERS' LIMITATION 1.1*)

POSR at 4

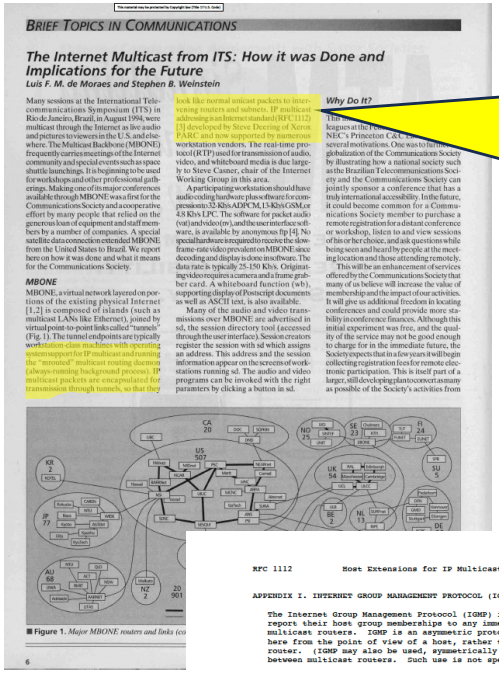
NEITHER KNAPPE NOR DE MORAES DISCLOSE "EACH OF THE MULTICAST APPLIANCES RECEIVING THE FIRST MESSAGE;" OF INDEPENDENT CLAIM 1 (*PETITIONERS' LIMITATION 1.2*)

POSR at 5

Petitioners thus argue that yet another reference on top of its combination, not part of the claimed combination (the RFC 1112 standard), could be applied using the claim as a template to a device that is neither taught nor suggested by De Moraes. Petitioners then proceed to craft their own theoretical prior art system, cavalierly applying the claimed technology to a reference-unspecified component "like the 'MBONE routers.'" *Id.* Petitioners picking a device "like the 'MBONE routers'," would fabricate a message from a reference that is not disclosed as part of the claimed combination and would manufacture a message and its content from whole cloth.

POSR at 4-5.

DE MORAES DISCLOSES IP MULTICASTING



BRIEF TOPICS IN COMMUNICATIONS

The Internet Multicast from ITS: How it was Done and Implications for the Future

Luis F. M. de Moraes and Stephen B. Weinstein

Many sessions at the International Telecommunications Symposium (ITS) in Rio de Janeiro, Brazil, in August 1994, were multicast through the Internet as live audio and pictures to viewers in the U.S. and elsewhere. The Multicast Backbone (MBONE) frequently carries meetings of the Internet community and special events such as space shuttle launchings. It is beginning to be used for workshops and other professional gatherings. Making one of its major conferences available through MBONE was a first for the Communications Society and a cooperative effort by many people that relied on the generous loan of equipment and staff members by a number of companies. A special satellite data connection extended MBONE from the United States to Brazil. We report here on how it was done and what it means for the Communications Society.

MBONE

MBONE, a virtual network layered on portions of the existing physical Internet [1,2] is composed of islands (such as multicast LANs like Ethernet), joined by virtual point-to-point links called "tunnels" (Fig. 1). The tunnel endpoints are typically workstation-class machines with operating system support for IP multicast and running the "mrouted" multicast routing daemon (always-running background process). IP multicast packets are encapsulated for transmission through tunnels, so that they look like normal unicast packets to intervening routers and subnets. IP multicast addressing is an Internet standard (RFC 1112) [3] developed by Steve Deering of Xerox PARC and now supported by numerous workstation vendors. The real-time protocol (RTP) used for transmission of audio, video, and whiteboard media is due largely to Steve Casner, chair of the Internet Working Group in this area.

A participating workstation should have audio coding hardware plus software for compression to 32-Kb/s ADPCM, 13-Kb/s GSM, or 4.8 Kb/s LPC. The software for packet audio (vat) and video (nv), and the user interface software, is available by anonymous ftp [4]. No special hardware is required to receive the slow-frame-rate video prevalent on MBONE since decoding and display is done in software. The data rate is typically 25–150 Kb/s. Originating video requires a camera and a frame grabber card. A whiteboard function (wb), supporting display of Postscript documents as well as ASCII text, is also available.

Many of the audio and video transmissions over MBONE are advertised in sd, the session directory tool (accessed through the user interface). Session creators register the session with sd which assigns an address. This address and the session information appear on the screens of workstations running sd. The audio and video programs can be invoked with the right parameters by clicking a button in sd.

Why Do It?

This [illegible] leagues at the [illegible] NEC's Princeton C&C [illegible] several motivations. One was to further globalization of the Communications Society by illustrating how a national society such as the Brazilian Telecommunications Society and the Communications Society can jointly sponsor a conference that has a truly international accessibility. In the future, it could become common for a Communications Society member to purchase a remote registration for a distant conference or workshop, listen to and view sessions of his or her choice, and ask questions while being seen and heard by people at the meeting location and those attending remotely.

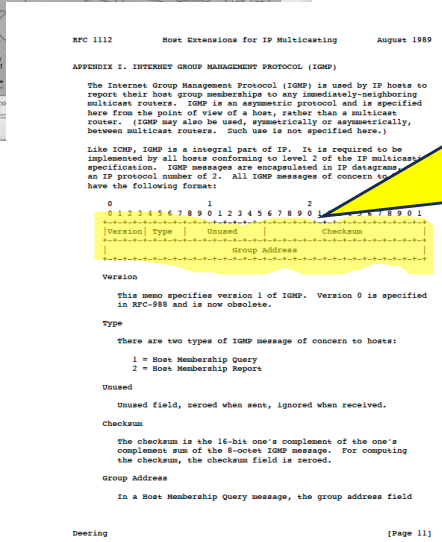
This will be an enhancement of services offered by the Communications Society that many of us believe will increase the value of membership and the impact of our activities. It will give us additional freedom in locating conferences and could provide more stability in conference finances. Although this initial experiment was free, and the quality of the service may not be good enough to charge for in the immediate future, the Society expects that in a few years it will begin collecting registration fees for remote electronic participation. This is itself part of a larger, still developing plan to connect as many as possible of the Society's activities from

Figure 1. *Major MBONE routers and links (co*

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The tunnel endpoints are typically workstations-class machines with operating system support for IP multicast … IP multicast addressing is a standard (RFC1112) …

Ex. 1022 at p. 7; *see also* Pet. at 37-38 & Ex. 1003 ¶ 134-136.



RFC 1112 Host Extensions for IP Multicasting August 1989

APPENDIX I. INTERNET GROUP MANAGEMENT PROTOCOL (IGMP)

The Internet Group Management Protocol (IGMP) is used by IP hosts to report their host group memberships to any immediately-neighboring multicast routers. IGMP is an asymmetric protocol and is specified here from the point of view of a host, rather than a multicast router. (IGMP may also be used, symmetrically or asymmetrically, between multicast routers. Such use is not specified here.)

Like ICMP, IGMP is a integral part of IP. It is required to be implemented by all hosts conforming to level 2 of the IP multicast specification. IGMP messages are encapsulated in IP datagrams, with an IP protocol number of 2. All IGMP messages of concern to hosts have the following format:

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|Version| Type  |    Unused     |           Checksum            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         Group Address                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Version

This memo specifies version 1 of IGMP. Version 0 is specified in RFC-988 and is now obsolete.

Type

There are two types of IGMP message of concern to hosts:

1 = Host Membership Query
2 = Host Membership Report

Unused

Unused field, zeroed when sent, ignored when received.

Checksum

The checksum is the 16-bit one's complement of the one's complement sum of the 8-octet IGMP message. For computing the checksum, the checksum field is zeroed.

Group Address

In a Host Membership Query message, the group address field

Deering [Page 11]

Like ICMP, IGMP is a integral part of IP. It is required to be implemented by all hosts conforming to level 2 of the IP multicasting specification. IGMP messages are encapsulated in IP datagrams, with an IP protocol number of 2. All IGMP messages of concern to hosts have the following format:

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|Version| Type  |    Unused     |           Checksum            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         Group Address                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Ex. 1030 at p. 7; *see also* Pet. at 37-38 & Ex. 1003 ¶ 134-136.

Zoho Corp. and Zoho Corp. Pvt., Ltd.
Exhibit 1030 – 011

POR at i-iii

REFERENCE TO RFC1112 TEACHES GROUP ADDRESS



BRIEF TOPICS IN COMMUNICATIONS

The Internet Multicast from ITS: How it was Done and Implications for the Future

Luis F. M. de Moraes and Stephen B. Weinstein

Why Do It?

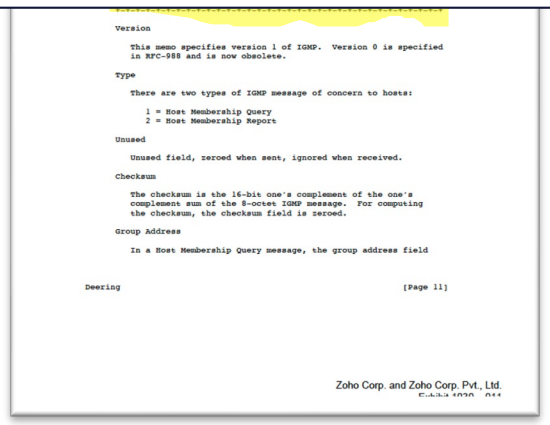
MBONE

Figure 1. Major MBONE routers

The tunnel endpoints are typically workstations-class machines with operating system

"A POSITA would have understood that using an Internet standard like RFC 1112 in an MBONE network as described in de Moraes would enable a host to send messages to a plurality of multicast appliances (e.g., MBONE and other multicast routers), such messages including "a group address which identifies a plurality of participants" as claimed in this element of claim 1."

Ex. 1003 ¶ 135



Version

This memo specifies version 1 of IGMP. Version 0 is specified in RFC-988 and is now obsolete.

Type

There are two types of IGMP message of concern to hosts:

1 = Host Membership Query
2 = Host Membership Report

Unused

Unused field, zeroed when sent, ignored when received.

Checksum

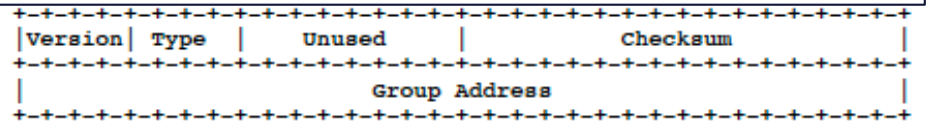
The checksum is the 16-bit one's complement of the one's complement sum of the 8-octet IGMP message. For computing the checksum, the checksum field is zeroed.

Group Address

In a Host Membership Query message, the group address field

Deering [Page 11]

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Ex. 1030 at p. 7; *see also* Pet. at 37-38 & Ex. 1003 ¶ 134-136.

RESPONSE TO LIMITATIONS [1.5A&C] ARGUMENTS

[1.5.a] the telephone participant calling into a gateway server;

[1.5.c] the gateway server establishing communication with a conferencing server;

KNAPPE DOES NOT DISCLOSE "THE TELEPHONE PARTICIPANT CALLING INTO A GATEWAY SERVER" LIMITATION OF INDEPENDENT CLAIM 1 (*PETITIONER'S LIMITATION 1.5A*)

POSR at 5

NEITHER KNAPPE NOR ELLIOTT DISCLOSE "THE GATEWAY SERVER ESTABLISHING COMMUNICATION WITH A CONFERENCING SERVER;" LIMITATION OF INDEPENDENT CLAIM 1 (*PETITIONERS' LIMITATION 1.5C*)

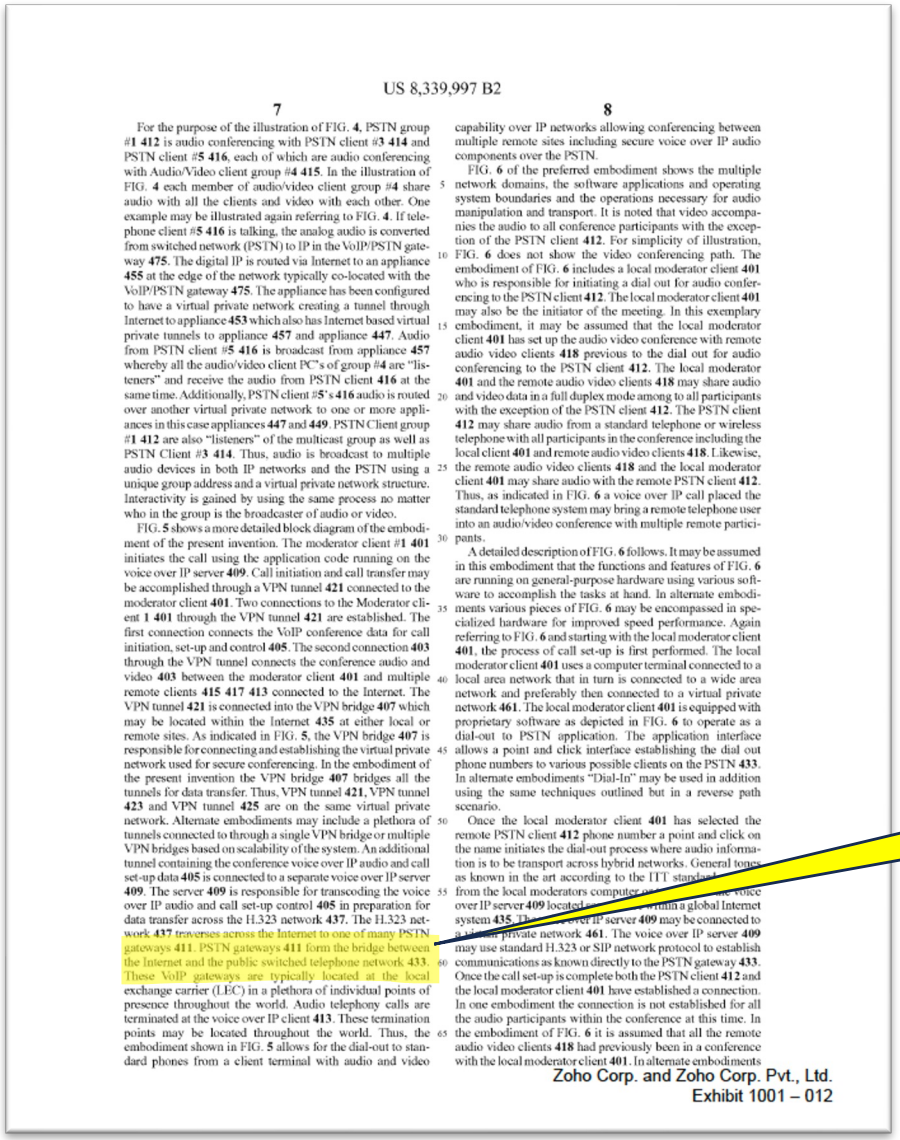
POSR at 8

and at Knappe, 4:18-28. Knappe describes the gateway as performing the functions of converting analog to digital, VOIP and other protocols. It is not disclosed as a server and is not a server. *See id*.

Conversely, a gateway server operates on a server-class computer and combines the functions of a gateway with server capabilities. It not only manages communications between diverse networks but also hosts applications, provides security features, and offers services such as caching, load balancing, or authentication. For example, gateway servers manage encryption, authentication, and authorization protocols to secure data transmission. Access control mechanisms ensure that only authorized users can join the conference, preventing unauthorized access. Exhibit 2001 (Dye Decl.) at ¶25.

POSR at 6-7

PATENT OWNER'S CONSTRUCTION AT ODDS WITH '997 PATENT SPECIFICATION AND CLAIMS



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For the purpose of the illustration of FIG. **4**, PSTN group #**1 412** is audio conferencing with PSTN client #**3 414** and PSTN client #**5 416**, each of which are audio conferencing with Audio/Video client group #**4 415**. In the illustration of FIG. **4** each member of audio/video client group #**4** share audio with all the clients and video with each other. One example may be illustrated again referring to FIG. **4**. If telephone client #**5 416** is talking, the analog audio is converted from switched network (PSTN) to IP in the VoIP/PSTN gateway **475**. The digital IP is routed via Internet to an appliance **455** at the edge of the network typically co-located with the VoIP/PSTN gateway **475**. The appliance has been configured to have a virtual private network creating a tunnel through Internet to appliance **453** which also has Internet based virtual private tunnels to appliance **457** and appliance **447**. Audio from PSTN client #**5 416** is broadcast from appliance **457** whereby all the audio/video client PC's of group #**4** are "listeners" and receive the audio from PSTN client **416** at the same time. Additionally, PSTN client #**5**'s **416** audio is routed over another virtual private network to one or more appliances in this case appliances **447** and **449**. PSTN Client group #**1 412** are also "listeners" of the multicast group as well as PSTN Client #**3 414**. Thus, audio is broadcast to multiple audio devices in both IP networks and the PSTN using a unique group address and a virtual private network structure. Interactivity is gained by using the same process no matter who in the group is the broadcaster of audio or video.

FIG. **5** shows a more detailed block diagram of the embodiment of the present invention. The moderator client #**1 401** initiates the call using the application code running on the voice over IP server **409**. Call initiation and call transfer may be accomplished through a VPN tunnel **421** connected to the moderator client **401**. Two connections to the Moderator client **1 401** through the VPN tunnel **421** are established. The first connection connects the VoIP conference data for call initiation, set-up and control **405**. The second connection **403** through the VPN tunnel connects the conference audio and video **403** between the moderator client **401** and multiple remote clients **415 417 413** connected to the Internet. The VPN tunnel **421** is connected into the VPN bridge **407** which may be located within the Internet **435** at either local or remote sites. As indicated in FIG. **5**, the VPN bridge **407** is responsible for connecting and establishing the virtual private network used for secure conferencing. In the embodiment of the present invention the VPN bridge **407** bridges all the tunnels for data transfer. Thus, VPN tunnel **421**, VPN tunnel **423** and VPN tunnel **425** are on the same virtual private network. Alternate embodiments may include a plethora of tunnels connected to through a single VPN bridge or multiple VPN bridges based on scalability of the system. An additional tunnel containing the conference voice over IP audio and call set-up data **405** is connected to a separate voice over IP server **409**. The server **409** is responsible for transcoding the voice over IP audio and call set-up control **405** in preparation for data transfer across the H.323 network **437**. The H.323 network **437** traverses across the Internet to one of many PSTN gateways **411**. PSTN gateways **411** form the bridge between the Internet and the public switched telephone network **433**. These VoIP gateways are typically located at the local exchange carrier (LEC) in a plethora of individual points of presence throughout the world. Audio telephony calls are terminated at the voice over IP client **413**. These termination points may be located throughout the world. Thus, the embodiment shown in FIG. **5** allows for the 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 voice over IP audio components over the PSTN.

FIG. **6** of the preferred embodiment shows the multiple network domains, the software applications and operating system boundaries and the operations necessary for audio manipulation and transport. It is noted that video accompanies the audio to all conference participants with the exception of the PSTN client **412**. For simplicity of illustration, FIG. **6** does not show the video conferencing path. The embodiment of FIG. **6** includes a local moderator client **401** who is responsible for initiating a dial out for audio conferencing to the PSTN client **412**. The local moderator client **401** may also be the initiator of the meeting. In this exemplary embodiment, it may be assumed that the local moderator client **401** has set up the audio video conference with remote audio video clients **418** previous to the dial out for audio conferencing to the PSTN client **412**. The local moderator **401** and the remote audio video clients **418** may share audio and video data in a full duplex mode among to all participants with the exception of the PSTN client **412**. The PSTN client **412** may share audio from a standard telephone or wireless telephone with all participants in the conference including the local client **401** and remote audio video clients **418**. Likewise, the remote audio video clients **418** and the local moderator client **401** may share audio with the remote PSTN client **412**. Thus, as indicated in FIG. **6** a voice over IP call placed the standard telephone system may bring a remote telephone user into an audio/video conference with multiple remote participants.

A detailed description of FIG. **6** follows. It may be assumed in this embodiment that the functions and features of FIG. **6** are running on general-purpose hardware using various software to accomplish the tasks at hand. In alternate embodiments various pieces of FIG. **6** may be encompassed in specialized hardware for improved speed performance. Again referring to FIG. **6** and starting with the local moderator client **401**, the process of call set-up is first performed. The local moderator client **401** uses a computer terminal connected to a local area network that in turn is connected to a wide area network and preferably then connected to a virtual private network **461**. The local moderator client **401** is equipped with proprietary software as depicted in FIG. **6** to operate as a dial-out to PSTN application. The application interface allows a point and click interface establishing the dial out phone numbers to various possible clients on the PSTN **433**. In alternate embodiments "Dial-In" may be used in addition using the same techniques outlined but in a reverse path scenario.

Once the local moderator client **401** has selected the remote PSTN client **412** phone number a point and click on the name initiates the dial-out process where audio information is to be transport across hybrid networks. General tones as known in the art according to the ITT standard [illegible] from the local moderators computer [illegible] voice over IP server **409** located [illegible] via a global Internet system **435**. The voice over IP server **409** may be connected to a [illegible] private network **461**. The voice over IP server **409** may use standard H.323 or SIP network protocol to establish communications as known directly to the PSTN gateway **433**. Once the call set-up is complete both the PSTN client **412** and the local moderator client **401** have established a connection. In one embodiment the connection is not established for all the audio participants within the conference at this time. In the embodiment of FIG. **6** it is assumed that all the remote audio video clients **418** had previously been in a conference with the local moderator client **401**. In alternate embodiments

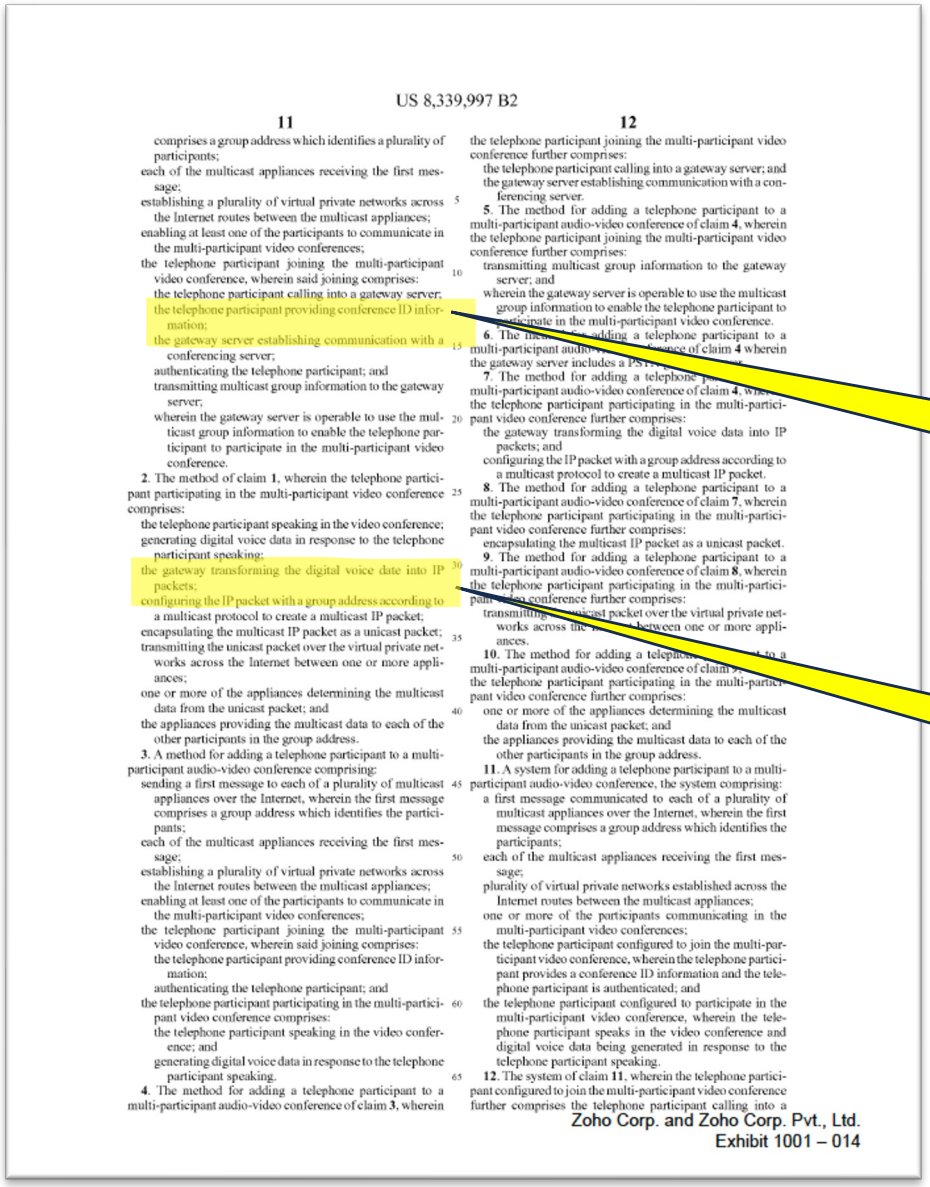
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Exhibit 1001 – 012

- Specification makes no reference to:
  - "server-class computers"
  - "caching"
  - "load balancing"
- Specification refers to the interface between telephone participants and conferencing server as "gateways":

PSTN gateways 411 form the bridge between the Internet and the public switched telephone network 433.

Reply at 8-9

PATENT OWNER'S CONSTRUCTION AT ODDS WITH '997 PATENT SPECIFICATION AND CLAIMS



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comprises a group address which identifies a plurality of participants;
each of the multicast appliances receiving the first message;
establishing a plurality of virtual private networks across the Internet routes between the multicast appliances;
enabling at least one of the participants to communicate in the multi-participant video conferences;
the telephone participant joining the multi-participant video conference, wherein said joining comprises:
the telephone participant calling into a gateway server;
the telephone participant providing conference ID information;
the gateway server establishing communication with a conferencing server;
authenticating the telephone participant; and
transmitting multicast group information to the gateway server;
wherein the gateway server is operable to use the multicast group information to enable the telephone participant to participate in the multi-participant video conference.

2. The method of claim 1, wherein the telephone participant participating in the multi-participant video conference comprises:
the telephone participant speaking in the video conference;
generating digital voice data in response to the telephone participant speaking;
the gateway transforming the digital voice date into IP packets;
configuring the IP packet with a group address according to a multicast protocol to create a multicast IP packet;
encapsulating the multicast IP packet as a unicast packet;
transmitting the unicast packet over the virtual private networks across the Internet between one or more appliances;
one or more of the appliances determining the multicast data from the unicast packet; and
the appliances providing the multicast data to each of the other participants in the group address.

3. A method for adding a telephone participant to a multi-participant audio-video conference comprising:
sending a first message to each of a plurality of multicast appliances over the Internet, wherein the first message comprises a group address which identifies the participants;
each of the multicast appliances receiving the first message;
establishing a plurality of virtual private networks across the Internet routes between the multicast appliances;
enabling at least one of the participants to communicate in the multi-participant video conferences;
the telephone participant joining the multi-participant video conference, wherein said joining comprises:
the telephone participant providing conference ID information;
authenticating the telephone participant; and
the telephone participant participating in the multi-participant video conference comprises:
the telephone participant speaking in the video conference; and
generating digital voice data in response to the telephone participant speaking.

4. The method for adding a telephone participant to a multi-participant audio-video conference of claim 3, wherein

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the telephone participant joining the multi-participant video conference further comprises:
the telephone participant calling into a gateway server; and
the gateway server establishing communication with a conferencing server.

5. The method for adding a telephone participant to a multi-participant audio-video conference of claim 4, wherein the telephone participant joining the multi-participant video conference further comprises:
transmitting multicast group information to the gateway server; and
wherein the gateway server is operable to use the multicast group information to enable the telephone participant to participate in the multi-participant video conference.

6. The [illegible] adding a telephone participant to a multi-participant audio-[illegible]nce of claim 4 wherein the gateway server includes a PS[illegible]

7. The method for adding a telephone [illegible] multi-participant audio-video conference of claim 4, [illegible] the telephone participant participating in the multi-participant video conference further comprises:
the gateway transforming the digital voice data into IP packets; and
configuring the IP packet with a group address according to a multicast protocol to create a multicast IP packet.

8. The method for adding a telephone participant to a multi-participant audio-video conference of claim 7, wherein the telephone participant participating in the multi-participant video conference further comprises:
encapsulating the multicast IP packet as a unicast packet.

9. The method for adding a telephone participant to a multi-participant audio-video conference of claim 8, wherein the telephone participant participating in the multi-participant [illegible] conference further comprises:
transmitting [illegible]icast packet over the virtual private networks across the [illegible] between one or more appliances.

10. The method for adding a telephone [illegible] to a multi-participant audio-video conference of claim [illegible] the telephone participant participating in the multi-participant video conference further comprises:
one or more of the appliances determining the multicast data from the unicast packet; and
the appliances providing the multicast data to each of the other participants in the group address.

11. A system for adding a telephone participant to a multi-participant audio-video conference, the system comprising:
a first message communicated to each of a plurality of multicast appliances over the Internet, wherein the first message comprises a group address which identifies the participants;
each of the multicast appliances receiving the first message;
plurality of virtual private networks established across the Internet routes between the multicast appliances;
one or more of the participants communicating in the multi-participant video conferences;
the telephone participant configured to join the multi-participant video conference, wherein the telephone participant provides a conference ID information and the telephone participant is authenticated; and
the telephone participant configured to participate in the multi-participant video conference, wherein the telephone participant speaks in the video conference and digital voice data being generated in response to the telephone participant speaking.

12. The system of claim 11, wherein the telephone participant configured to join the multi-participant video conference further comprises the telephone participant calling into a

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Exhibit 1001 – 014

- Claims refer to "gateway server" and "gateway" interchangeably:

the telephone participant calling into a gateway server;

the gateway transforming the digital voice date [sic] into IP packets

Reply at 9

RESPONSE TO LIMITATION [1.5B] ARGUMENTS

[1.5.b] the telephone participant providing conference ID information;

NEITHER KNAPPE NOR ELLIOTT DISCLOSE "THE TELEPHONE PARTICIPANT PROVIDING CONFERENCE ID INFORMATION;" LIMITATION OF INDEPENDENT CLAIM 1 (*PETITIONERS' LIMITATION 1.5B*) POSR at 7

client via VPN is decrypted and shared to the PSTN client. Moreover, the session ID of Elliott is different for each participant and specific to one-to-one session between the call centre and one of the participants, the conference ID is consistent for a multicast video conferencing between multiple users. Hence, the session ID of Elliott cannot be used to show the conference ID present in the element.

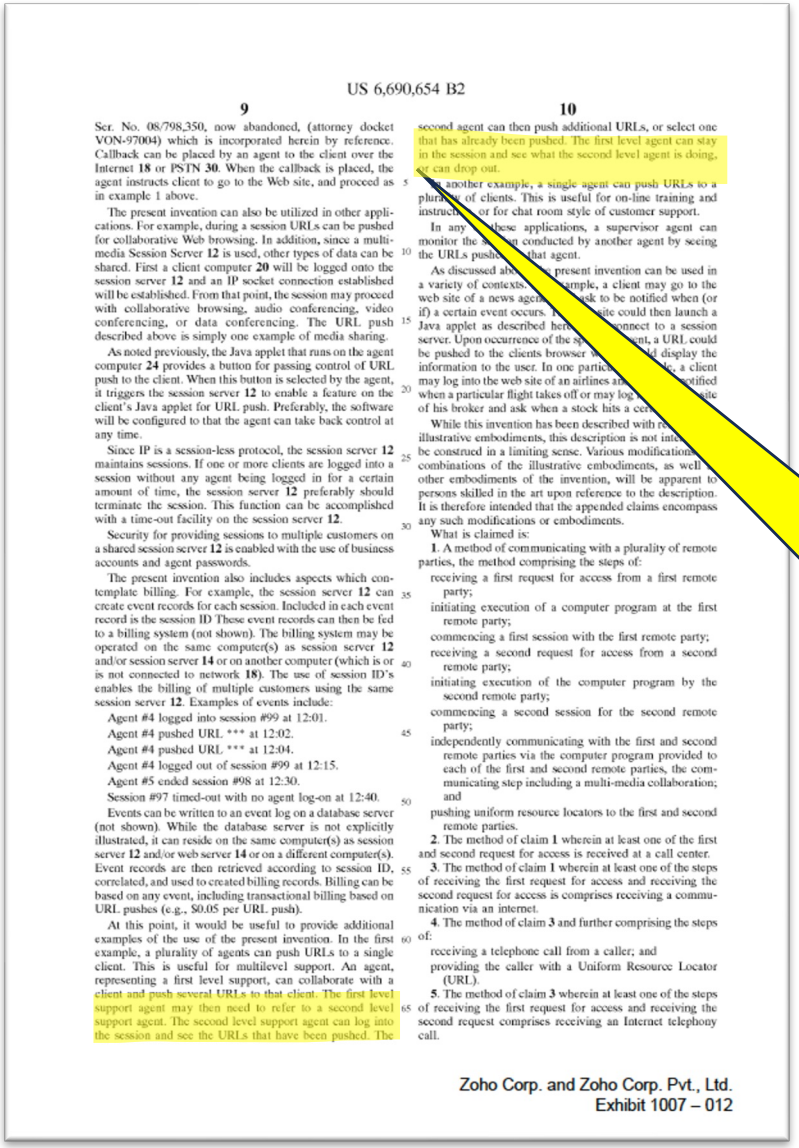
POR at 22-23

CONFERENCE ID INFORMATION DOES NOT REQUIRE MULTIPLE USERS

- Conference ID information is not limited as Patent Owner argues
  - Could be specific to one-to-one sessions
  - Could be different for each of multiple participants
- Elliott's session ID is not specific to one-to-one sessions:
  - Used to log the client into the session with the session ID for a given conference
  - Multiple agents can log to the same conference with the session ID

Reply at 10-11

ELLIOTT'S SESSION ID IS NOT LIMITED TO ONE-TO-ONE SESSIONS



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Ser. No. 08/798,350, now abandoned, (attorney docket VON-97004) which is incorporated herein by reference. Callback can be placed by an agent to the client over the Internet **18** or PSTN **30**. When the callback is placed, the agent instructs client to go to the Web site, and proceed as in example 1 above.

The present invention can also be utilized in other applications. For example, during a session URLs can be pushed for collaborative Web browsing. In addition, since a multimedia Session Server **12** is used, other types of data can be shared. First a client computer **20** will be logged onto the session server **12** and an IP socket connection established will be established. From that point, the session may proceed with collaborative browsing, audio conferencing, video conferencing, or data conferencing. The URL push described above is simply one example of media sharing.

As noted previously, the Java applet that runs on the agent computer **24** provides a button for passing control of URL push to the client. When this button is selected by the agent, it triggers the session server **12** to enable a feature on the client's Java applet for URL push. Preferably, the software will be configured to that the agent can take back control at any time.

Since IP is a session-less protocol, the session server **12** maintains sessions. If one or more clients are logged into a session without any agent being logged in for a certain amount of time, the session server **12** preferably should terminate the session. This function can be accomplished with a time-out facility on the session server **12**.

Security for providing sessions to multiple customers on a shared session server **12** is enabled with the use of business accounts and agent passwords.

The present invention also includes aspects which contemplate billing. For example, the session server **12** can create event records for each session. Included in each event record is the session ID These event records can then be fed to a billing system (not shown). The billing system may be operated on the same computer(s) as session server **12** and/or session server **14** or on another computer (which is or is not connected to network **18**). The use of session ID's enables the billing of multiple customers using the same session server **12**. Examples of events include:

Agent #4 logged into session #99 at 12:01.
Agent #4 pushed URL *** at 12:02.
Agent #4 pushed URL *** at 12:04.
Agent #4 logged out of session #99 at 12:15.
Agent #5 ended session #98 at 12:30.
Session #97 timed-out with no agent log-on at 12:40.

Events can be written to an event log on a database server (not shown). While the database server is not explicitly illustrated, it can reside on the same computer(s) as session server **12** and/or web server **14** or on a different computer(s). Event records are then retrieved according to session ID, correlated, and used to created billing records. Billing can be based on any event, including transactional billing based on URL pushes (e.g., $0.05 per URL push).

At this point, it would be useful to provide additional examples of the use of the present invention. In the first example, a plurality of agents can push URLs to a single client. This is useful for multilevel support. An agent, representing a first level support, can collaborate with a client and push several URLs to that client. The first level support agent may then need to refer to a second level support agent. The second level support agent can log into the session and see the URLs that have been pushed. The

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second agent can then push additional URLs, or select one that has already been pushed. The first level agent can stay in the session and see what the second level agent is doing, or can drop out.

In another example, a single agent can push URLs to a plurality of clients. This is useful for on-line training and instruction or for chat room style of customer support.

In any of these applications, a supervisor agent can monitor the session conducted by another agent by seeing the URLs pushed to that agent.

As discussed above, the present invention can be used in a variety of contexts. For example, a client may go to the web site of a news agency and ask to be notified when (or if) a certain event occurs. The site could then launch a Java applet as described herein to connect to a session server. Upon occurrence of the specified event, a URL could be pushed to the clients browser which would display the information to the user. In one particular example, a client may log into the web site of an airlines and ask to be notified when a particular flight takes off or may log into the web site of his broker and ask when a stock hits a certain price.

While this invention has been described with reference to illustrative embodiments, this description is not intended to be construed in a limiting sense. Various modifications and combinations of the illustrative embodiments, as well as other embodiments of the invention, will be apparent to persons skilled in the art upon reference to the description. It is therefore intended that the appended claims encompass any such modifications or embodiments.

What is claimed is:

1. A method of communicating with a plurality of remote parties, the method comprising the steps of:

receiving a first request for access from a first remote party;

initiating execution of a computer program at the first remote party;

commencing a first session with the first remote party;

receiving a second request for access from a second remote party;

initiating execution of the computer program by the second remote party;

commencing a second session for the second remote party;

independently communicating with the first and second remote parties via the computer program provided to each of the first and second remote parties, the communicating step including a multi-media collaboration; and

pushing uniform resource locators to the first and second remote parties.

2. The method of claim **1** wherein at least one of the first and second request for access is received at a call center.

3. The method of claim **1** wherein at least one of the steps of receiving the first request for access and receiving the second request for access is comprises receiving a communication via an internet.

4. The method of claim **3** and further comprising the steps of:

receiving a telephone call from a caller; and

providing the caller with a Uniform Resource Locator (URL).

5. The method of claim **3** wherein at least one of the steps of receiving the first request for access and receiving the second request comprises receiving an Internet telephony call.

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Exhibit 1007 – 012

- Elliott's session ID is not specific to one-to-one sessions:
  - Multiple agents can log to the same conference with the same session ID

The first level support agent may then need to refer to a second level support agent. The second level support agent can log into the session … The first level agent can stay in the session and see what the second level agent is doing or can drop out.

Ex. 1007 at 9:60-10-4; *see also* Reply at 11

RESPONSE TO LIMITATION [1.5D] ARGUMENTS

[1.5.d] authenticating the telephone participant; and

KNAPPE DOES NOT DISCLOSE THE "AUTHENTICATING THE TELEPHONE PARTICIPANT;" LIMITATION OF INDEPENDENT CLAIM 1 (*PETITIONERS' LIMITATION 1.5D*)

POSR at 9

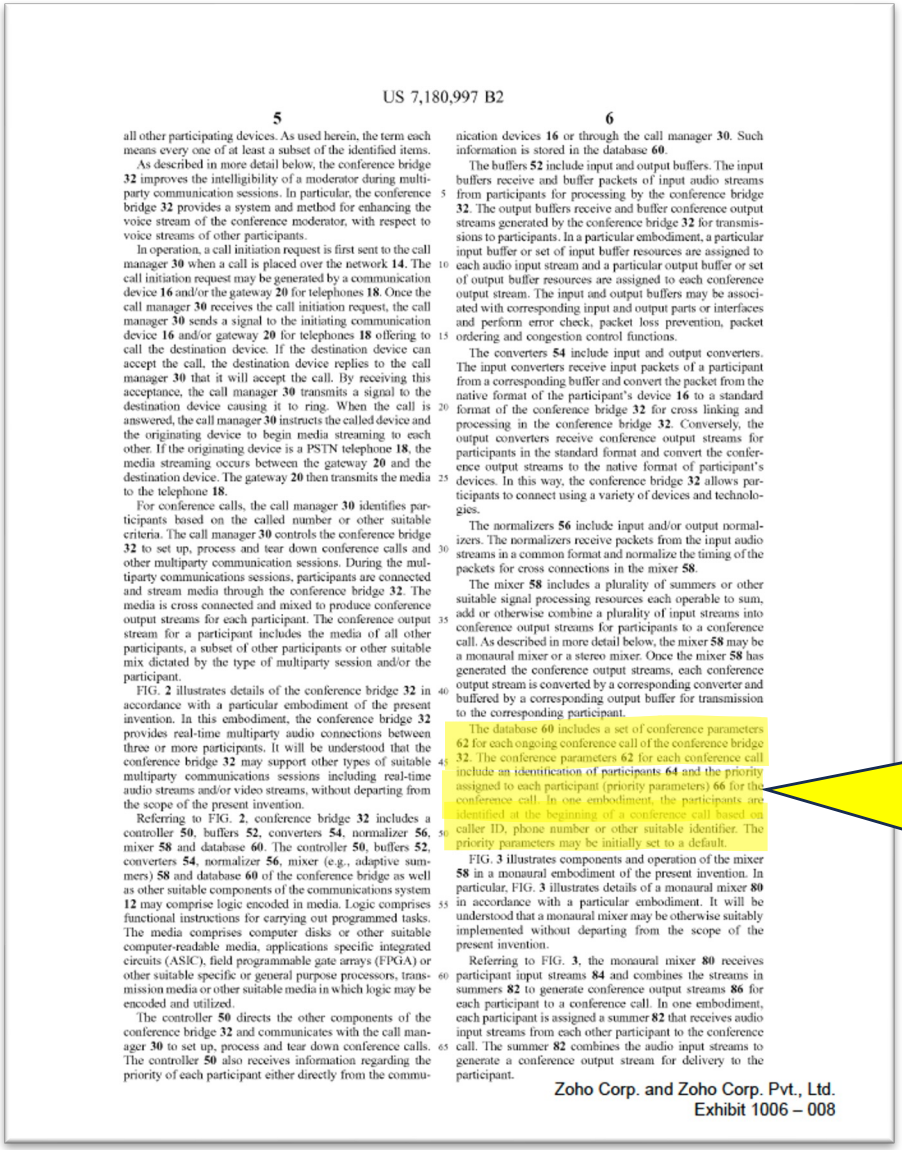
Knappe teaches identifying a participant using phone number or caller ID.

. . .

techniques. The authentication process checks the provided credentials against stored records to confirm the identity. Hence, Knappe only discloses information regarding identification and fails to perform authentication.

POR at 23-24

KNAPPE TEACHES AUTHENTICATING TELEPHONE PARTICIPANTS



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all other participating devices. As used herein, the term each means every one of at least a subset of the identified items.

As described in more detail below, the conference bridge **32** improves the intelligibility of a moderator during multi-party communication sessions. In particular, the conference bridge **32** provides a system and method for enhancing the voice stream of the conference moderator, with respect to voice streams of other participants.

In operation, a call initiation request is first sent to the call manager **30** when a call is placed over the network **14**. The call initiation request may be generated by a communication device **16** and/or the gateway **20** for telephones **18**. Once the call manager **30** receives the call initiation request, the call manager **30** sends a signal to the initiating communication device **16** and/or gateway **20** for telephones **18** offering to call the destination device. If the destination device can accept the call, the destination device replies to the call manager **30** that it will accept the call. By receiving this acceptance, the call manager **30** transmits a signal to the destination device causing it to ring. When the call is answered, the call manager **30** instructs the called device and the originating device to begin media streaming to each other. If the originating device is a PSTN telephone **18**, the media streaming occurs between the gateway **20** and the destination device. The gateway **20** then transmits the media to the telephone **18**.

For conference calls, the call manager **30** identifies participants based on the called number or other suitable criteria. The call manager **30** controls the conference bridge **32** to set up, process and tear down conference calls and other multiparty communication sessions. During the multiparty communications sessions, participants are connected and stream media through the conference bridge **32**. The media is cross connected and mixed to produce conference output streams for each participant. The conference output stream for a participant includes the media of all other participants, a subset of other participants or other suitable mix dictated by the type of multiparty session and/or the participant.

FIG. **2** illustrates details of the conference bridge **32** in accordance with a particular embodiment of the present invention. In this embodiment, the conference bridge **32** provides real-time multiparty audio connections between three or more participants. It will be understood that the conference bridge **32** may support other types of suitable multiparty communications sessions including real-time audio streams and/or video streams, without departing from the scope of the present invention.

Referring to FIG. **2**, conference bridge **32** includes a controller **50**, buffers **52**, converters **54**, normalizer **56**, mixer **58** and database **60**. The controller **50**, buffers **52**, converters **54**, normalizer **56**, mixer (e.g., adaptive summers) **58** and database **60** of the conference bridge as well as other suitable components of the communications system **12** may comprise logic encoded in media. Logic comprises functional instructions for carrying out programmed tasks. The media comprises computer disks or other suitable computer-readable media, applications specific integrated circuits (ASIC), field programmable gate arrays (FPGA) or other suitable specific or general purpose processors, transmission media or other suitable media in which logic may be encoded and utilized.

The controller **50** directs the other components of the conference bridge **32** and communicates with the call manager **30** to set up, process and tear down conference calls. The controller **50** also receives information regarding the priority of each participant either directly from the communication devices **16** or through the call manager **30**. Such information is stored in the database **60**.

The buffers **52** include input and output buffers. The input buffers receive and buffer packets of input audio streams from participants for processing by the conference bridge **32**. The output buffers receive and buffer conference output streams generated by the conference bridge **32** for transmissions to participants. In a particular embodiment, a particular input buffer or set of input buffer resources are assigned to each audio input stream and a particular output buffer or set of output buffer resources are assigned to each conference output stream. The input and output buffers may be associated with corresponding input and output ports or interfaces and perform error check, packet loss prevention, packet ordering and congestion control functions.

The converters **54** include input and output converters. The input converters receive input packets of a participant from a corresponding buffer and convert the packet from the native format of the participant's device **16** to a standard format of the conference bridge **32** for cross linking and processing in the conference bridge **32**. Conversely, the output converters receive conference output streams for participants in the standard format and convert the conference output streams to the native format of participant's devices. In this way, the conference bridge **32** allows participants to connect using a variety of devices and technologies.

The normalizers **56** include input and/or output normalizers. The normalizers receive packets from the input audio streams in a common format and normalize the timing of the packets for cross connections in the mixer **58**.

The mixer **58** includes a plurality of summers or other suitable signal processing resources each operable to sum, add or otherwise combine a plurality of input streams into conference output streams for participants to a conference call. As described in more detail below, the mixer **58** may be a monaural mixer or a stereo mixer. Once the mixer **58** has generated the conference output streams, each conference output stream is converted by a corresponding converter and buffered by a corresponding output buffer for transmission to the corresponding participant.

The database **60** includes a set of conference parameters **62** for each ongoing conference call of the conference bridge **32**. The conference parameters **62** for each conference call include an identification of participants **64** and the priority assigned to each participant (priority parameters) **66** for the conference call. In one embodiment, the participants are identified at the beginning of a conference call based on caller ID, phone number or other suitable identifier. The priority parameters may be initially set to a default.

FIG. **3** illustrates components and operation of the mixer **58** in a monaural embodiment of the present invention. In particular, FIG. **3** illustrates details of a monaural mixer **80** in accordance with a particular embodiment. It will be understood that a monaural mixer may be otherwise suitably implemented without departing from the scope of the present invention.

Referring to FIG. **3**, the monaural mixer **80** receives participant input streams **84** and combines the streams in summers **82** to generate conference output streams **86** for each participant to a conference call. In one embodiment, each participant is assigned a summer **82** that receives audio input streams from each other participant to the conference call. The summer **82** combines the audio input streams to generate a conference output stream for delivery to the participant.

Zoho Corp. and Zoho Corp. Pvt., Ltd.
Exhibit 1006 – 008

- Knappe is not limited to telephone numbers or caller ID:
  - Other forms of identification disclosed include employee numbers, IP addresses, "or other suitable identifier"
- To the extent "authentication" requires checking ID against store records, this is disclosed in Knappe

The database 60 includes a set of conference parameters 62 …. The conference parameters 62 … include an identification of participants 64 and the priority assigned to each participant (priority parameters) 66 for the conference call. In one embodiment, the participants are identified at the beginning of a conference call based on caller ID, phone number, or other suitable identifier.

Ex. 1006 at 6:45-47; *see also* Reply at 12-13

RESPONSE TO LIMITATION [1.6] ARGUMENTS

[1.6] wherein the gateway server is operable to use the multicast group information to enable the telephone participant to participate in the multi-participant video conference.

> **NEITHER KNAPPE NOR DE MORAES NOR ELLIOTT DISCLOSE THE "WHEREIN THE GATEWAY SERVER IS OPERABLE TO USE THE MULTICAST GROUP INFORMATION TO ENABLE THE TELEPHONE PARTICIPANT TO PARTICIPATE IN THE MULTI-PARTICIPANT VIDEO CONFERENCE." LIMITATION OF INDEPENDENT CLAIM 1 (*PETITIONERS' LIMITATION 1.6*)**
>
> POSR at 10

As set forth above, neither Knappe nor Elliott teach a gateway server.

Moreover, De Moraes's teaching is inconsistent with the PSTN network or PSTN telephones. As set forth in De Moraes, "A participating workstation should have audio coding hardware plus software for compression to 32-Kb/sADPCM,13-Kb/sGSM,or 4.8 Kb/s LPC."

. . .

In summary, Knappe does not disclose a gateway server. Additionally, although De Moraes' work transmits via multicast, it does not teach incorporating PSTN telephones into conference calls.

POR at 24-25

DE MORAES COMBINED WITH KNAPPE OR ELLIOT TEACH LIMITATION [1.6]

- Petition relied on the combinations of Knappe **or** Elliott with de Moraes
  - Patent owner never addressed the Elliott/de Moraes combination for this element Reply at 13
  - "Gateway server" argument fails as explained before
- Patent owner argues that de Moraes "cannot be applied to PSTN telephones" because de Moraes teaches that "[a] participant workstation should have audio coding hardware plus software for compression to 32-Kb/sADPCM, 13Kb/sGSM, or 4.8 Kb/s LPC."
  - Petition does not rely on de Moraes being applied to PSTN telephones.
  - "A POSITA would have known **to implement Knappe's gateway server 20** to use multicast group information from Knappe's desitnation devices (e.g., multicast appliances) to enable a telephone participant (e.g., telephones 18) to participate in the multi-party video conference."
  - The conventional IP multicast techniques described in de Moraes would apply to the gateway server, not the PSTN telephones.
  - De Moraes' audio coding hardware and software configuration in the workstations misses the point.

Pet. at 50 & Reply at 14

RESPONSE TO LIMITATION [7.2] ARGUMENTS

[7.2] configuring the IP packet with a group address according to a multicast protocol to create a multicast IP packet.

DE MORAES DOES NOT DISCLOSE THE "CONFIGURING THE IP PACKET WITH A GROUP ADDRESS ACCORDING TO A MULTICAST PROTOCOL TO CREATE A MULTICAST IP PACKET." LIMITATION OF DEPENDENT CLAIM 7 ~~*(PETITIONERS' LIMITATION 2.8)*~~

POSR at 15

De Moraes excludes the Public Switched Telephone Network (PSTN) or PSTN telephones, thereby omitting crucial information essential for understanding the transmission of multicast data within a network incorporating PSTN infrastructure. Specifically, De Moraes fails to address the inclusion of PSTN telephones capable of facilitating the transfer of multicast group address. Moreover, De Moraes' design fails to specify whether the configuration of multicast IP packets occurs at the gateway.

POSR at 15-16

DE MORAES ALONE NOT RELEVANT – KNAPPE AND DE MORAES TEACH LIMITATION [7.2]

- Petitioner does not rely on de Moraes but on the combination of Knappe and de Moraes:
  - Addressing de Moraes in isolation fails
- Knappe and de Moraes teach the claimed limitation
  - Combined references teach the inclusion of PSTN telephone participants in a multicast conference network via a gateway.
  - Knappe teaches that the IP addressing with respect to telephone participants takes place at the gateway.
  - De Moraes teaches that the IP addressing at the gateway can be done according to the IP multicasting protocol, RFC1112, which includes configuring an IP multicast packet with a group address.

Reply at 15

GROUND 2: CLAIM 2

- Patent Owner Sur Reply Identified Wrong Limitation Numbers
- Claim 2
  - Limitation 2.5
  - Limitation 2.6
  - Limitation 2.7
  - Limitation 2.8

RESPONSE TO LIMITATION [2.5] ARGUMENTS

[2.5] encapsulating the multicast IP packet as a unicast packet;

NEITHER KNAPPE NOR DE MORAES NOR SHEPHERD DISCLOSE THE "THE METHOD OF CLAIM 1, WHEREIN THE TELEPHONE PARTICIPANT PARTICIPATING IN THE MULTI-PARTICIPANT VIDEO CONFERENCE COMPRISES: ENCAPSULATING THE MULTICAST IP PACKET AS A UNICAST PACKET;" LIMITATION OF DEPENDENT CLAIM 2

POSR at 11

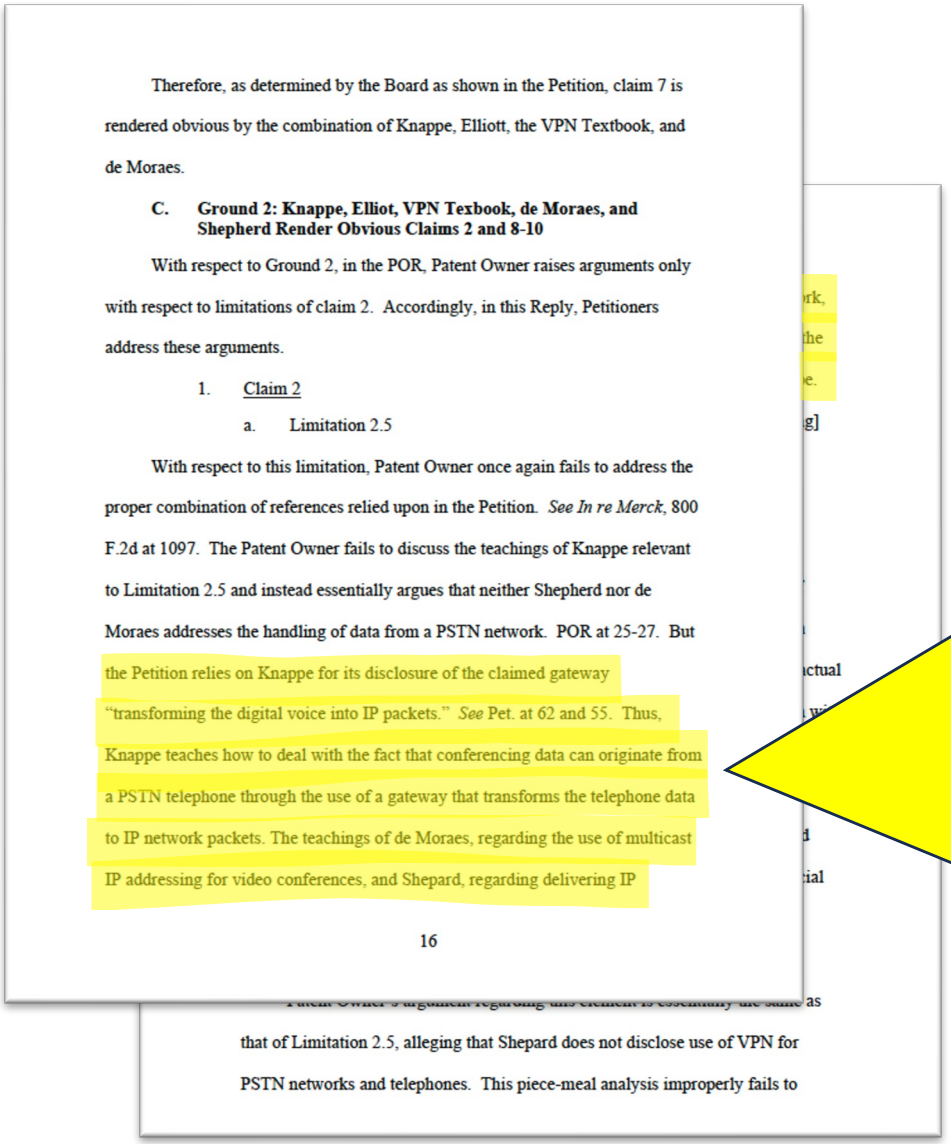
- Patent Owner Response failed to address Knappe's teachings

Reply at 16

  - New argument presented in the Sur Reply adding Knappe to argument (p.13)
  - Largely rehashing de Moraes no being applicable to PSTN telephones
- In Sur Reply Patent Owner alleges waiver
  - Patent Owner was confused about limitation numbering – no waiver
  - Petitioner's Reply in fact responded "to Patent Owner's arguments regarding the 'encapsulating' element of claim 2"

See POSR at 11

NO WAIVER – KNAPPE WITH DE MORAES AND SHEPHERD TEACHES LIMITATION [2.5]



Therefore, as determined by the Board as shown in the Petition, claim 7 is rendered obvious by the combination of Knappe, Elliott, the VPN Textbook, and de Moraes.

C. Ground 2: Knappe, Elliot, VPN Texbook, de Moraes, and Shepherd Render Obvious Claims 2 and 8-10

With respect to Ground 2, in the POR, Patent Owner raises arguments only with respect to limitations of claim 2. Accordingly, in this Reply, Petitioners address these arguments.

1. Claim 2

a. Limitation 2.5

With respect to this limitation, Patent Owner once again fails to address the proper combination of references relied upon in the Petition. *See In re Merck*, 800 F.2d at 1097. The Patent Owner fails to discuss the teachings of Knappe relevant to Limitation 2.5 and instead essentially argues that neither Shepherd nor de Moraes addresses the handling of data from a PSTN network. POR at 25-27. But the Petition relies on Knappe for its disclosure of the claimed gateway "transforming the digital voice into IP packets." *See* Pet. at 62 and 55. Thus, Knappe teaches how to deal with the fact that conferencing data can originate from a PSTN telephone through the use of a gateway that transforms the telephone data to IP network packets. The teachings of de Moraes, regarding the use of multicast IP addressing for video conferences, and Shepard, regarding delivering IP

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that of Limitation 2.5, alleging that Shepard does not disclose use of VPN for PSTN networks and telephones. This piece-meal analysis improperly fails to

Petition relies on Knappe for its disclosure of the claimed gateway "transforming the digital voice into IP packets." … Thus Knappe teaches how to deal with the fact that conferring data can originate from a PSTN telephone through the use of a gateway that transforms the telephone data to IP network packets. The teachings of de Moraes, regarding the use of multicast IP addressing for videoconferences, and Shephard [sic], regarding delivering IP multicast messages such as those disclosed by de Moraes over a unicast network, would have been considered by a POSITA in order to "reduce costs borne by the content provider to deliver content to destination devices" according to Knappe.

Reply at 16-17 & Pet. 34 and 60

RESPONSE TO LIMITATION [2.6] ARGUMENTS

[2.6] transmitting the unicast packet over the virtual private networks across the Internet between one or more appliances;

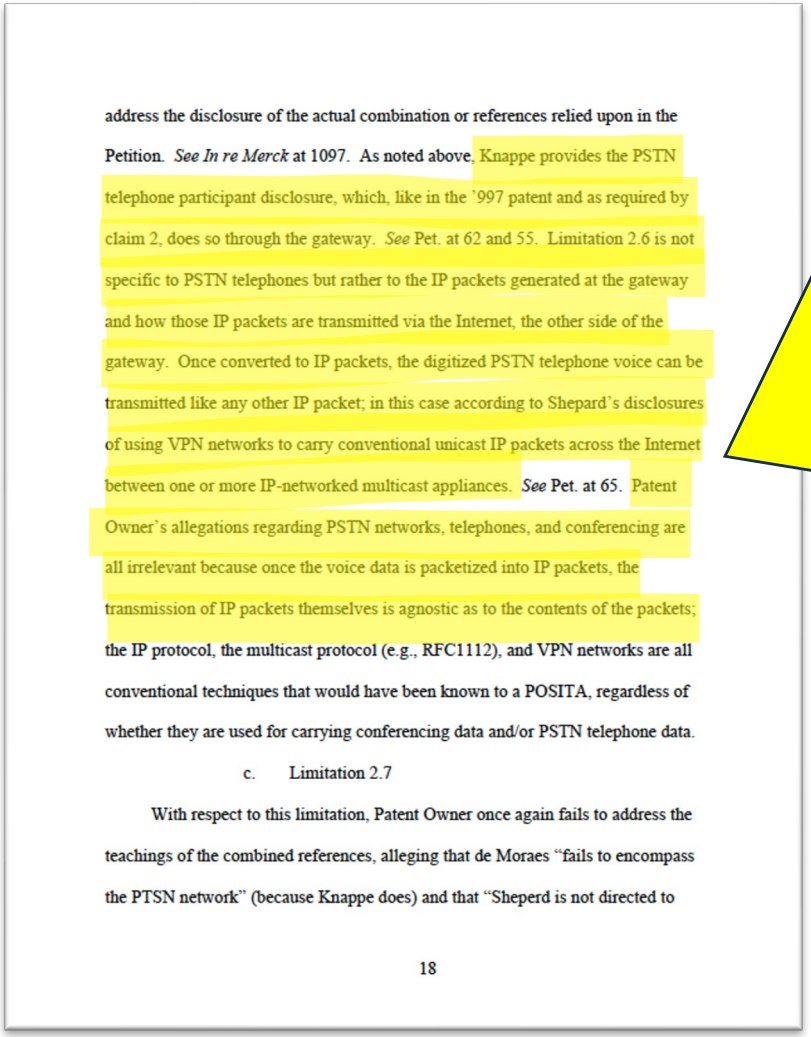
NEITHER KNAPPE NOR SHEPHERD DISCLOSE THE "TRANSMITTING THE UNICAST PACKET OVER THE VIRTUAL PRIVATE NETWORKS ACROSS THE INTERNET BETWEEN ONE OR MORE APPLIANCES;" LIMITATION OF DEPENDENT CLAIM 2 (~~*PETITIONERS' LIMITATION 2.5)*~~

POSR at 12

- Patent Owner's argument is limited to Shephard in isolation
  - Shepherd "focuses on the delivery and reception of multicast content across unicast networks …"
  - Shepherd's relevance to the realm of videoconferencing or conferencing is notably absent …
  - Shepherd's disclosure lacks any reference to use of a VPN (Virtual Private Network) for PSTN networks and telephones.

POR at 27-28

SHEPPARD DISCLOSES LIMITATION [2.6]



address the disclosure of the actual combination or references relied upon in the Petition. *See In re Merck* at 1097. As noted above, Knappe provides the PSTN telephone participant disclosure, which, like in the '997 patent and as required by claim 2, does so through the gateway. *See* Pet. at 62 and 55. Limitation 2.6 is not specific to PSTN telephones but rather to the IP packets generated at the gateway and how those IP packets are transmitted via the Internet, the other side of the gateway. Once converted to IP packets, the digitized PSTN telephone voice can be transmitted like any other IP packet; in this case according to Shepard's disclosures of using VPN networks to carry conventional unicast IP packets across the Internet between one or more IP-networked multicast appliances. *See* Pet. at 65. Patent Owner's allegations regarding PSTN networks, telephones, and conferencing are all irrelevant because once the voice data is packetized into IP packets, the transmission of IP packets themselves is agnostic as to the contents of the packets; the IP protocol, the multicast protocol (e.g., RFC1112), and VPN networks are all conventional techniques that would have been known to a POSITA, regardless of whether they are used for carrying conferencing data and/or PSTN telephone data.

c. Limitation 2.7

With respect to this limitation, Patent Owner once again fails to address the teachings of the combined references, alleging that de Moraes "fails to encompass the PTSN network" (because Knappe does) and that "Sheperd is not directed to

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Knappe provides the PSTN telephone participant disclosure, which, like the '997 patent and as required by claim 2, does so through the gateway. … Limitation 2.6 is not specific to PSTN telephones but rather to the IP packets generated at the gateway and how those IP packets are transmitted via the Internet, the other side of the gateway. Once converted to IP packets, the digitized PSTN telephone voice can be transmitted like any other IP packet; in this case according to Shepard's [sic] disclosures of using VPN networks to carry conventional unicast IP packets across the Internet between one or more IP-networked multicast appliances. … Patent Owner's allegations regarding PSTN networks, telephones, and conferencing are all irrelevant because once the voice data is packetized into IP packets, the transmission of IP packets themselves is agnostic as to the content of the packets …

Reply at 18 & Pet. at 65

RESPONSE TO LIMITATION [2.7] ARGUMENTS

[2.7] one or more of the appliances determining the multicast data from the unicast packet; and

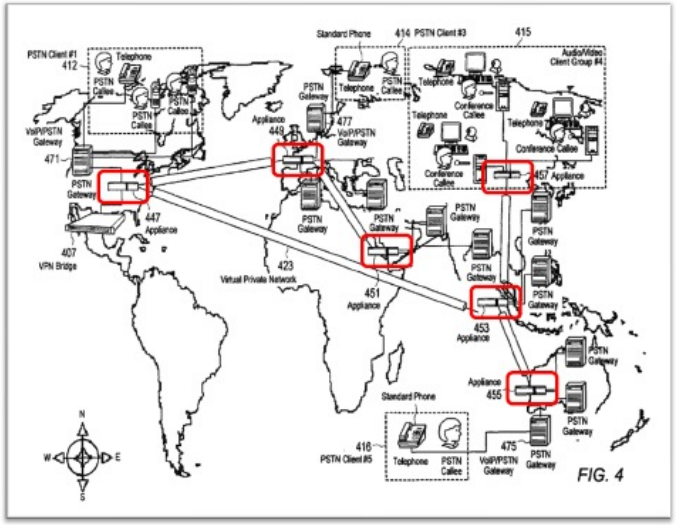
NEITHER KNAPPE NOR SHEPHERD NOR DE MORAES DISCLOSE "ONE OR MORE OF THE APPLIANCES DETERMINING THE MULTICAST DATA FROM THE UNICAST PACKET" OF DEPENDENT CLAIM 2 (~~*PETITIONERS' LIMITATION 2.6)*~~

POSR at 14

Neither De Moraes nor Shepherd teach the determination of multicast data from unicast data performed by communication devices at the user end. To the extent performed in their disparate systems, any determination of multicast data is performed by routers positioned between end user devices or communication devices (16).

POSR at 14

PATENT OWNER FAILED TO RESPOND TO PETITIONER'S ACTUAL ARGUMENT



- Claim requires "one or more of the appliances" to perform the "determining"
  - In the '997, the appliances are not limited to "devices at the user end" as PO argues
  - In Fig. "Appliances" 455, 451, 457, 453, 449, and 447 are routers between VPN tunnels and gateways, bridges, and other devices that then connect to end user devices
  - The appliances then provide the multicast data to the participants' communication devices either directly (e.g. Appliance 457) or via gateways (e.g., Appliance 455).
- PO misrepresents de Moraes:
  - De Moraes teaches the multicast protocol operations taking place at end user workstations "with operating system support for IP multicast"
  - Figure 2 shows "MBONE routers" as end-user "workstations" – e.g., workstation connected to "Surenet", "T1 router" etc.

Reply at 20-21

RESPONSE TO LIMITATION [2.8] ARGUMENTS

[2.8] the appliances providing the multicast data to each of the other participants in the group address.

NEITHER KNAPPE NOR SHEPHERD NOR DE MORAES DISCLOSE "THE APPLIANCES PROVIDING THE MULTICAST DATA TO EACH OF THE OTHER PARTICIPANTS IN THE GROUP ADDRESS." LIMITATION OF DEPENDENT CLAIM 2 (~~*PETITIONER'S LIMITATION 2.7)*~~

POSR at 14

- Patent Owner's arguments are repetitive of previously addressed arguments:
  - De Moraes "fails to encompass the PSTN network …" – Knappe does
  - Shepherd "is not directed to videoconferencing … [and] it fails to address the intricate process of handling data from PSTN telephones … - the combination of references does.
  - Neither de Moraes nor Shepherd teach the determination of multicast data from unicast data performed by communication devices at the user end – not required by the claim and in fact disclosed in de Moraes.

POSR at 15