Osterhout in view of Kung

'519 Claim	Claim Limitations	Osterhout U.S. Patent 7,197,029	Secondary References
		memory), storage. Col. 2, ll. 60-62. SIP module 122 may transmit a Call Invite command to the recipient telephone device 120, in this instance a SIP-enabled device, to await a 200 OK or other acknowledgment message for processing the call. Col. 5, ll. 7-10.	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet network (a voice or other path, **such as VoIP or VOATM, may be established between the USB telephone 102 and the recipient telephone device 120). and to send call** log data to the system management platform **(control module 126 is a platform (software framework) running at host computers 106 and may receive and store call parameters).**	"Once the call setup is achieved via SIP messaging, a voice or other path, such as VoIP or VOATM, may be established between the USB telephone 102 and the recipient telephone device 120. Other voice path or other protocols may be used, such as voice over UDP or fax over TCP, or others known in the art. Call processing my proceed according to known messaging according to those protocols, once established." Col. 5, ll. 11-16. **System management platform:** Control module 126 is a platform (software framework) running at host computers and may receive and store call parameters. For example, a first control module 126 running at a first host computer 106 receives and stores call parameters for a call to second control module 126 running at a second host computer 106. Control modules 126 operate as a system managment platform collecting call log data for a plurality of network devices (host computers 106). "The control module 126 may receive and store desired call parameters for the user, for instance minimum call quality parameters which will be acceptable for the user to place a network-based call. For instance, the control module 126 may monitor the communications link 112 to determine line conditions or other variables for the placement of a digital network call. These variables may include signal-to-noise ratio (SNR), packet congestion or delay, or other parameters affecting the quality, features, costs or other aspects of a call." Col. 4, ll. 39-48.	**Base System**—Osterhout discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, host computer 106). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. *Kung*, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Kung teaches an IP central station 200 that stores a call log: "The present invention may include an activity log that may have user proactive bill management capability and be used in the aforementioned broadband communication system. The activity log may log, for example, incoming calls directory numbers (DNs) and outgoing call DNs in a database. The database containing the activity log may be provided at a central system location, such as the at IP Central Station 200." Col. 31, ll. 10-17. Figure 8 of Kung includes an example call log. The call log is stored at BRG 300 and/or IP central station 200. Col. 32, ll.9-10.

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'519 Claim	Claim Limitations	Osterhout U.S. Patent 7,197,029	Secondary References
		"In step 412, a determination may be made whether the transmission criteria for the user may be met. The control module 126 may record different criteria for different users, and present a user login screen to apply those criteria. If the determination of step 412 is that the transmission criteria are not met, then call processing proceeds to step 414. In step 414 a call may be dialed using the public switched telephone network via POTS and SS7 signaling, or other telephony standards. In step 416, call teardown of the public telephony network call may be completed and processing continues to step 424." Col. 6, ll. 17-27.	The system subscriber's customer premises equipment (broadband residential gateway 300) records the call log data and forwards the call log data to other locations, such as to IP central station 200, for billing purposes as an example. Figure 8; Col. 35, l. 37 - col. 36, l. 10. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Osterhout to include a system management platform, for example, to collect call log data from the network devices of Osterhout.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"The host computer 106 may also be connected to a data network, for instance via communications link 112 to the public Internet 116, to which a recipient telephone device 120 may in turn be connected." Col. 3, ll. 49-62.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	For instance, the SIP module 122 may transmit a Call Invite command to the recipient telephone device 120, in this instance a SIP-enabled device, to await a 200 OK or other acknowledgment message for processing the call. Once the call setup is achieved via SIP messaging, a voice or other path, such as VoIP or VOATM, may be established between the USB telephone 102 and the recipient telephone device 120. Other voice path or other protocols may be used, such as voice over UDP or fax over TCP, or others known in the art. Col. 5, ll. 7-16.	

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'519 Claim	Claim Limitations	Osterhout U.S. Patent 7,197,029	Secondary References
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	The '519 patent states that the SIP protocol stack "functions as the default SIP Proxy Server." Col. 24, ll. 27-28. SIP stack 124 of Osterhout acts as an intermediary and transmits and receives SIP commands.	
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	"If the criteria are met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call processing." Col. 4, l. 65 - Col. 5 - l. 6.	
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	For instance, the SIP module 122 may transmit a Call Invite command to the recipient telephone device 120, in this instance a SIP-enabled device, to await a 200 OK or other acknowledgment message for processing the call. Once the call setup is achieved via SIP messaging, a voice or other path, such as VoIP or VOATM, may be established between the USB telephone 102 and the recipient telephone device 120. Other voice path or other protocols may be used, such as voice over UDP or fax over TCP, or others known in the art. Call processing my proceed according to known messaging according to those protocols, once established. Col. 5, ll. 7-17.	
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface (video input).	"The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages	

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'519 Claim	Claim Limitations	Osterhout U.S. Patent 7,197,029	Secondary References
		or protocols may also be used for video teleconferencing or other applications, if the USB telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5.	

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248
Claim 1	A network device **(switch 50)** comprising: a plurality of communication interfaces, including a telephone line interface **(interface between SIP - unobservant phone 40 and switch 50)**, a computer data interface **(interface between location server 55 and switch 50)**, and a broadband network interface **(interface between internet 45 and switch 50)**;	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59. "Switching device 50 is coupled to a location server 55. Switching device 50 is also coupled to switching device 60 over the SIP-observant network 45 via one or more core switches operative on the network. Switching device 60 is in turn coupled to the SIP-observant phone 65. The SIP-observant and unobservant phones 40, 65, switching devices 50, 60, and location server 55, are interconnected via cables or other transmission media known in the art." Col. 3, ll. 60-67.
	a processor **(processor implementing emulation client 50a)**;	"According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 18-21.
	a machine-readable storage medium **(switch 50 including emulation client 50a)** which during use stores	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor. Col. 4, ll. 11-21.
	a call processing application **(emulation client 50a)** and	"Switching device 50 receives the dial request and invokes its emulation client 50a to convert the request into a SIP-observant format. In doing so, the emulation client 5a preferably transmits all or a portion of the telephone number to the location server 55 to determine whether the call is to be redirected to a different number or to a particular extension. Preferably, the location server 55 returns the

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248
		same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.
	service profiles **(user-defined call parameters stored in emulation client 50a or account information stored at location server 55)**,	Emulation client 50a stores user-defined call parameters received from location server 55 for call routing. "The location server 55 preferably contains rules and algorithms for redirecting calls to a location where a user of the SIP-observant phone 65 is scheduled to be. The user's location may vary based on a time and/or day of the call. Alternatively, the location server 55 contains rules and algorithms for redirecting calls made to a call center, to an appropriate extension or agent. The redirection may be based on, for instance, caller information, agent availability, account information, and the like." Col. 3, ll. 53-60.
	and which stores executable instructions to mediate communications between the plurality of communication interfaces **(instructions to mediate communications with SIP - unobservant phone 40)**,	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor. Col. 4., ll. 11-21.
	the instructions causing the network device to detect network signaling events or trigger points in a telephone call **(dial request received from SIP - unobservant phone 40)** and	In general terms, the SIP-unobservant phone 40 initiates telephonic communication with the SIP-observant phone 65 by transmitting a PBX dial request with a particular telephone number. Col. 4, ll. 31-34.
	invoke the call processing application **(emulation client 50a)** in response to the detected network signaling events or trigger points,	"Switching device 50 receives the dial request and invokes its emulation client 50a to convert the request into a SIP-observant format. In doing so, the emulation client 5a preferably transmits all or a portion of the telephone number to the location server 55 to determine whether the call is to be redirected to a different number or to a particular extension. Preferably, the location server 55 returns the

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248
		same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.
	the call processing application operating according to parameters defined in the service profiles	Emulation client 50a operates according to the user-defined call parameters received from location server 55 for call routing. "Switching device 50 receives the dial request and invokes its emulation client 50a to convert the request into a SIP-observant format. In doing so, the emulation client 5a preferably transmits all or a portion of the telephone number to the location server 55 to determine whether the call is to be redirected to a different number or to a particular extension. Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.
	wherein the network device consists of one or more customer premise equipment modules (switch 50).	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location." Col. 4, ll. 11-13.
Claim 2	The network device of claim 1, wherein the plurality of communication interfaces further includes a video streaming device interface.	"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.
Claim 3	The network device of claim 1, wherein the broadband network interface terminates a broadband network link that joins a customer premises to a packet carrier network.	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248
Claim 4	The network device of claim 1, wherein the instructions further cause the network device to route IP data between the computer data interface and the broadband network interface.	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.
Claim 5	The network device of claim 1, wherein the network device is contained in a single physical enclosure.	Switch 50 is contained in a single physical enclosure. Figure 2.
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to provide a SIP user agent to represent a telephone that uses the telephone line interface.	Emulation client 50a provides a SIP user agent that represents a telephone that uses the telephone line interface. FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.
Claim 7	The network device of claim 1, wherein the storage medium during use further stores call routing tables (**location server 95 returns a list of all potential routing numbers**), and the instructions further cause the network device to perform call routing for telephone calls that use the telephone line interface.	"the location server 95 identifies a particular number to where to route the call. Alternatively, the location server 95 returns a list of all potential routing numbers, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is engaged to retrieve redirection information if an initially dialed number does not result in a successful connection." Col. 5, ll. 35-42.
Claim 8	The network device of claim 1, wherein the storage medium during use further stores call routing tables (**location server 95 returns a list of all potential routing numbers**), and the instructions further cause the network device to perform call routing for telephone calls according to the call routing tables, the	"the location server 95 identifies a particular number to where to route the call. Alternatively, the location server 95 returns a list of all potential routing numbers, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is engaged to retrieve redirection

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248
	telephone calls using ~~the telephone line interface.~~	~~information if an initially dialed number~~ does not result in a successful connection." ~~Col. 5, ll. 35-42.~~
Claim 9	A network device (switch 50) comprising: a broadband network interface (interface between Internet 45 and switch 50); a plurality of communication interfaces, including a telephone line interface (interface between SIP - unobservant phone 40 and switch 50) and a computer data interface (interface between location server 55 and switch 50);	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 11-21.
	a processor (processor implementing emulation client 50a);	"According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 18-21.
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents (switch 50 including emulation client 50a)	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor. Col. 4, ll. 11-21.
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface (SIP module 122 represents USB phone 102), and the instructions further causing the network device to implement a SIP proxy server (SIP stack 124) that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	"In general terms, the SIP-unobservant phone 40 initiates telephonic communication with the SIP-observant phone 65 by transmitting a PBX dial request with a particular telephone number." Col. 4, ll. 31-34. "Switching device 50 receives the dial request and invokes its emulation client 50a to convert the request into a SIP-observant format. In doing so, the emulation

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248
		client 5a preferably transmits all or a portion of the telephone number to the location server 55 to determine whether the call is to be redirected to a different number or to a particular extension. Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.
Claim 11	~~The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.~~	~~"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.~~
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	Switch 50 is contained in a single physical enclosure. Figure 2.
~~Claim 13~~	~~A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (location server 55), the system management platform collecting call log data from a plurality of network devices ; and~~	~~"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 11-21.~~

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
		redirected to a different number or to a particular extension. Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.	
	wherein the network device consists of one or more customer premise equipment modules (switch 50).	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location." Col. 4, ll. 11-13.	
Claim 7	The network device of claim 1, wherein the storage medium during use further stores call routing tables **(location server 95 returns a list of all potential routing numbers)**, and the instructions further cause the network device to perform call routing for telephone calls that use the telephone line interface.	"the location server 95 identifies a particular number to where to route the call. Alternatively, the location server 95 returns a list of all potential routing numbers, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is engaged to retrieve redirection information if an initially dialed number does not result in a successful connection." Col. 5, ll. 35-42.	To the extent that Wengrovitz does not explicitly teach call routing tables, Chung (U.S. Patent 6584108) teaches call routing tables. "The extra digits are passed on to the private branch exchange which will use them to connect the call to the correct extension. Call routing is supported via a static mapping table in each MAC, but the embodiment is not so limited." (Col. 16, ll. 14-18). Chung teaches the use of call routing tables in order to efficiently route telephone calls and avoid the need for call routing through the private branch exchange. Col. 16., ll. 41-50. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize call routing tables as taught by

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
			Chung (U.S. Patent 6584108) to enable the network device telephones of Wengrovitz to efficiently route telephone calls, for example.
Claim 8	The network device of claim 1, wherein the storage medium during use further stores call routing tables **(location server 95 returns a list of all potential routing numbers)**, and the instructions further cause the network device to perform call routing for telephone calls according to the call routing tables, the telephone calls using the telephone line interface.	"the location server 95 identifies a particular number to where to route the call. Alternatively, the location server 95 returns a list of all potential routing numbers, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is engaged to retrieve redirection information if an initially dialed number does not result in a successful connection." Col. 5, ll. 35-42.	To the extent that Wengrovitz does not explicitly teach call routing tables, Chung (U.S. Patent 6584108) teaches call routing tables. "The extra digits are passed on to the private branch exchange which will use them to connect the call to the correct extension. Call routing is supported via a static mapping table in each MAC, but the embodiment is not so limited." (Col. 16, ll. 14-18). Chung teaches the use of call routing tables in order to efficiently route telephone calls and avoid the need for call routing through the private branch exchange. Col. 16., ll. 41-50. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize call routing tables as taught by Chung (U.S. Patent 6584108) to enable the network device telephones of Wengrovitz to efficiently route telephone calls, for example.

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
		redirected to a different number or to a particular extension. Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.	
	wherein the network device consists of one or more customer premise equipment modules (switch 50).	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location." Col. 4, ll. 11-13.	
Claim 2	The network device of claim 1, wherein the plurality of communication interfaces further includes a video streaming device interface.	"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.	To the extent that Wengrovitz does not explicitly teach a video streaming device interface, Osterhout (U.S. Patent 7,197,029) teaches interfaces for a telephone that make use of audio, video, and other media. Col. 6, ll. 1-5. Osterhout explains the need for video interfaces in network devices for video conferencing: "The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages or protocols may also be used for video teleconferencing or other applications, if the USB telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
			multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize Osterhout's video streaming device interface in Wengrovitz to provide an interface for video conferencing data, for example.
Claim 9	A network device (switch 50) comprising: a broadband network interface (interface between internet 45 and switch 50); a plurality of communication interfaces, including a telephone line interface (interface between SIP - unobservant phone 40 and switch 50) and a computer data interface (interface between location server 55 and switch 50);	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 11-21.	
	a processor (processor implementing emulation client 50a);	"According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 18-21.	
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents (switch 50 including emulation client 50a)	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on	

APPENDIX D3

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'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.	To the extent that Wengrovitz does not explicitly teach a video streaming device interface, Osterhout (U.S. Patent 7,197,029) teaches interfaces for a telephone that make use of audio, video, and other media. Col. 6, ll. 1-5. Osterhout explains the need for video interfaces in network devices for video conferencing: "The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages or protocols may also be used for video teleconferencing or other applications, if the USB telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize Osterhout's video streaming device interface in Wengrovitz to provide an interface for video conferencing data, for example.
Claim 13	~~A method for establishing a voice-over-packet network architecture, the method comprising:~~	~~"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a~~	

APPENDIX D5

Wengrovitz in view of Inbar

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (location server 55), the system management platform collecting call log data from a plurality of network devices ; and	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 11-21. Location server 55 collects call log data that is used in the routing of calls to particular locations. "The location database includes a plurality of location records 520, with each record preferably being headed and identified by a unique caller identifier (ID) 520a."	**Base System**—Wengrovitz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, switch 50). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. *Inbar*, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Inbar states: "The IPCenter preferably records usage and billing information, and, as described above, reports billing information to the Master-Server, or to a separate billing unit associated with the master server. In addition to usage and billing information, the IPCenter may report Quality-of-Service (QoS) information, and in some cases connectivity monitoring information, status information of connected devices and other information as may be defined." Figure 1; Col. 8, ll. 54-62. "The system preferably further comprises a billing mechanism for accumulating a transaction log at the subscriber end and retrieving data of said log to the master server." Col. 4, ll. 16-19. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Wengrovitz to include a system management platform, for example, to maintain these records in a centralized system and facilitate billing: "all of these services

Wengrovitz in view of Inbar

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
			have to be integrated with each other, with a central control and with billing servers and other functions." Col. 1, ll. 41-43.
	distributing the plurality of network devices **(switch 50)** that each include a telephone line interface **(interface between SIP - unobservant phone 40 and switch 50),** a computer data interface **(interface between location server 55 and switch 50),** a broadband network interface terminating a link from the shared packet network **(interface between internet 45 and switch 50)**	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 11-21.	
	a processor **(processor implementing emulation client 50a)**;	"According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 18-21.	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls **(switch 50 including emulation client 50a),**	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor. Col. 4, ll. 11-21.	

Wengrovitz in view of Inbar

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet network **(SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543).** and to send call log data to the system management platform **(location server 55).**	"Switching device 50 preferably creates a SIP INVITE request with the identified telephone number and transmits the request over the SIP-observant network 45. Switching device 60 receives the INVITE request and transmits the request to the SIP-observant phone 65. The SIP-observant phone 65 preferably alerts the callee of the incoming call by emitting, for example, a ringing sound. If the call is answered, switching device 60 indicates a successful connection by returning a SIP OK response to the emulation client 50a. The emulation client 50a translates the OK response to a PBX answer event and transmits it to the SIP-observant phone. In addition, the emulation client 50a confirms the receipt of the OK response by transmitting a SIP ACK message to switching device 60. The SIP-observant and observant phones 40, 65 may then engage in communication until one of the parties terminate the call." Col. 4, ll. 48-63. "Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 40-47. **System management platform:** Emulation client 50a sends call log data (the telephone number to be dialed, for example) to location server 55, and location server 55 returns the same number or a different number from the call log data stored at location server 55. Although not depicted, location server 55 may serve more than one switch. Location server 55 collects call log data that is used in the routing of calls to particular locations. "The location database	**Base System**—Wengrovitz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, switch 50). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. *Inbar*, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Inbar states: "The IPCenter preferably records usage and billing information, and, as described above, reports billing information to the Master-Server, or to a separate billing unit associated with the master server. In addition to usage and billing information, the IPCenter may report Quality-of-Service (QoS) information, and in some cases connectivity monitoring information, status information of connected devices and other information as may be defined." Figure 1; Col. 8, ll. 54-62. "The system preferably further comprises a billing mechanism for accumulating a transaction log at the subscriber end and retrieving data of said log to the master server." Col. 4, ll. 16-19. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Wengrovitz to include a system management platform, for example, to maintain these records in a centralized system and facilitate billing: "all of these services have to be integrated with each other, with a

Wengrovitz in view of Inbar

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
		includes a plurality of location records 520, with each record preferably being headed and identified by a unique caller identifier (ID) 520a." Emulation client 50a sends call log data (the telephone number to be dialed, for example) to location server 55, and location server 55 returns the same number or a different number from the call log data stored at location server 55. "Switching device 50 receives the dial request and invokes its emulation client 50a to convert the request into a SIP-observant format. In doing so, the emulation client 5a preferably transmits all or a portion of the telephone number to the location server 55 to determine whether the call is to be redirected to a different number or to a particular extension. Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.	central control and with billing servers and other functions." Col. 1, ll. 41-43.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.	

Wengrovitz in view of Inbar

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	"Switching device 50 preferably creates a SIP INVITE request with the identified telephone number and transmits the request over the SIP-observant network 45. Switching device 60 receives the INVITE request and transmits the request to the SIP-observant phone 65. The SIP-observant phone 65 preferably alerts the callee of the incoming call by emitting, for example, a ringing sound. If the call is answered, switching device 60 indicates a successful connection by returning a SIP OK response to the emulation client 50a. The emulation client 50a translates the OK response to a PBX answer event and transmits it to the SIP-observant phone. In addition, the emulation client 50a confirms the receipt of the OK response by transmitting a SIP ACK message to switching device 60. The SIP-observant and observant phones 40, 65 may then engage in communication until one of the parties terminate the call." Col. 4, ll. 48-63.	
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor. Col. 4, ll. 11-21.	
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-	

APPENDIX D5

Wengrovitz in view of Inbar

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
		unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.	
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	SIP is transport layer independent. Therefore, SIP supports the underlying transport protocol of IP over ATM.	Further in view of Osterhout D4-claim 18
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.	Further in view of Osterhout D3-claim 19

APPENDIX D3

Wengrovitz and Inbar further in view of Osterhout

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.	To the extent that Wengrovitz does not explicitly teach a video streaming device interface, Osterhout (U.S. Patent 7,197,029) teaches interfaces for a telephone that make use of audio, video, and other media. Col. 6, ll. 1-5. Osterhout explains the need for video interfaces in network devices for video conferencing: "The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages or protocols may also be used for video teleconferencing or other applications, if the USB telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize Osterhout's video streaming device interface in Wengrovitz to provide an interface for video conferencing data, for example.

APPENDIX D4

Wengrovitz and Inbar further in view of Osterhout

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	SIP is transport layer independent. Therefore, SIP supports the underlying transport protocol of IP over ATM.	To the extent that Wengrovitz does not explicitly teach ATM protocols, the use of ATM protocols in network devices was well known in the art by 2001. For example, Osterhout (U.S. Patent 7,197,029) teaches ATM protocols. Col. 6, ll. 1-5. Therefore the use of ATM protocols would have been a simple design choice to one of ordinary skill in the art.

APPENDIX D6

Wengrovitz in view of Kung

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (location server 55), the system management platform collecting call log data from a plurality of network devices ; and	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 11-21. Location server 55 collects call log data that is used in the routing of calls to particular locations. "The location database includes a plurality of location records 520, with each record preferably being headed and identified by a unique caller identifier (ID) 520a."	**Base System**—Wengrovitz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, switch 50). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. *Kung*, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Kung teaches an IP central station 200 that stores a call log: "The present invention may include an activity log that may have user proactive bill management capability and be used in the aforementioned broadband communication system. The activity log may log, for example, incoming calls directory numbers (DNs) and outgoing call DNs in a database. The database containing the activity log may be provided at a central system location, such as the at IP Central Station 200." Col. 31, ll. 10-17. Figure 8 of Kung includes an example call log. The call log is stored at BRG 300 and/or IP central station 200. Col. 32, ll.9-10. The system subscriber's customer premises equipment (broadband residential gateway 300) records the call log data and forwards the call log data to other locations, such as to IP central station 200, for billing purposes as an example. Figure 8;

Wengrovitz in view of Kung

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
			Col. 35, l. 37 - col. 36, l. 10. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Wengrovitz to include a system management platform, for example, to collect call log data from the network devices of Wengrovitz.
	distributing the plurality of network devices **(switch 50)** that each include a telephone line interface **(interface between SIP - unobservant phone 40 and switch 50),** a computer data interface **(interface between location server 55 and switch 50),** a broadband network interface terminating a link from the shared packet network **(interface between internet 45 and switch 50)**	"Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 11-21.	
	a processor **(processor implementing emulation client 50a);**	"According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Col. 4, ll. 18-21.	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls **(switch 50 including emulation client 50a),**	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a	

Wengrovitz in view of Kung

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
		UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor. Col. 4, ll. 11-21.	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet network **(SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543).** and to send call log data to the system management platform **(location server 55).**	"Switching device 50 preferably creates a SIP INVITE request with the identified telephone number and transmits the request over the SIP-observant network 45. Switching device 60 receives the INVITE request and transmits the request to the SIP-observant phone 65. The SIP-observant phone 65 preferably alerts the callee of the incoming call by emitting, for example, a ringing sound. If the call is answered, switching device 60 indicates a successful connection by returning a SIP OK response to the emulation client 50a. The emulation client 50a translates the OK response to a PBX answer event and transmits it to the SIP-observant phone. In addition, the emulation client 50a confirms the receipt of the OK response by transmitting a SIP ACK message to switching device 60. The SIP-observant and observant phones 40, 65 may then engage in communication until one of the parties terminate the call." Col. 4, ll. 48-63. "Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 40-47. **System management platform:** Emulation client 50a sends call log data (the telephone number to be dialed, for example) to location server 55, and location server 55 returns the same number or a different number from the call log data stored at	**Base System**—Wengrovitz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, switch 50). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. *Kung*, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Kung teaches an IP central station 200 that stores a call log: "The present invention may include an activity log that may have user proactive bill management capability and be used in the aforementioned broadband communication system. The activity log may log, for example, incoming calls directory numbers (DNs) and outgoing call DNs in a database. The database containing the activity log may be provided at a central system location, such as the at IP Central Station 200." Col. 31, ll. 10-17. Figure 8 of Kung includes an example call log. The call log is stored at BRG 300 and/or IP central station 200. Col. 32, ll.9-10. The system subscriber's customer premises equipment (broadband residential gateway 300)

Wengrovitz in view of Kung

'819 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
		location server 55. Although not depicted, location server 55 may serve more than one switch. Location server 55 collects call log data that is used in the routing of calls to particular locations. "The location database includes a plurality of location records 520, with each record preferably being headed and identified by a unique caller identifier (ID) 520a." Emulation client 50a sends call log data (the telephone number to be dialed, for example) to location server 55, and location server 55 returns the same number or a different number from the call log data stored at location server 55. "Switching device 50 receives the dial request and invokes its emulation client 50a to convert the request into a SIP-observant format. In doing so, the emulation client 5a preferably transmits all or a portion of the telephone number to the location server 55 to determine whether the call is to be redirected to a different number or to a particular extension. Preferably, the location server 55 returns the same number or a different number if the call is to be redirected. In an alternative embodiment, the location server 55 returns a list of all potential numbers for redirecting the call, and each number is tried for a connection until a response is received. In yet another embodiment, the location server is only engaged if the initially dialed number does not result in a successful connection." Col. 4, ll. 34-47.	records the call log data and forwards the call log data to other locations, such as to IP central station 200, for billing purposes as an example. Figure 8; Col. 35, l. 37 - col. 36, l. 10. Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Wengrovitz to include a system management platform, for example, to collect call log data from the network devices of Wengrovitz.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol	

Wengrovitz in view of Kung

'819 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
		set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	"Switching device 50 preferably creates a SIP INVITE request with the identified telephone number and transmits the request over the SIP-observant network 45. Switching device 60 receives the INVITE request and transmits the request to the SIP-observant phone 65. The SIP-observant phone 65 preferably alerts the callee of the incoming call by emitting, for example, a ringing sound. If the call is answered, switching device 60 indicates a successful connection by returning a SIP OK response to the emulation client 50a. The emulation client 50a translates the OK response to a PBX answer event and transmits it to the SIP-observant phone. In addition, the emulation client 50a confirms the receipt of the OK response by transmitting a SIP ACK message to switching device 60. The SIP-observant and observant phones 40, 65 may then engage in communication until one of the parties terminate the call." Col. 4, ll. 48-63.	
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor. Col. 4, ll. 11-21.	

APPENDIX D6

Wengrovitz in view of Kung

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	"FIG. 2 is a schematic block diagram of a data communication network supporting an emulation service for a SIP-unobservant phone 40 initiating telephonic communication with a SIP-observant phone 65. The SIP-unobservant phone 40 communicates over a SIP-observant network 45 that preferably supports the SIP signaling protocol set forth in RFC 2543. The SIP-observant network 45 is preferably a wide area network such as the Internet." Col. 3, ll. 52-59.	
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	SIP is transport layer independent. Therefore, SIP supports the underlying transport protocol of IP over ATM.	
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.	Further in view of Osterhout D3-claim 19

APPENDIX D3

Wengrovitz and Kung further in view of Osterhout

'519 Claim	Claim Limitations	Wengrovitz U.S. Patent 7,035,248	Secondary References
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	"Session Initiation Protocol (SIP) is a signaling protocol for creating, modifying, and terminating multimedia sessions." Col. 1, ll. 19-21.	To the extent that Wengrovitz does not explicitly teach a video streaming device interface, Osterhout (U.S. Patent 7,197,029) teaches interfaces for a telephone that make use of audio, video, and other media. Col. 6, ll. 1-5. Osterhout explains the need for video interfaces in network devices for video conferencing: "The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages or protocols may also be used for video teleconferencing or other applications, if the USB telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize Osterhout's video streaming device interface in Wengrovitz to provide an interface for video conferencing data, for example.

APPENDIX E1

Baratz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596
Claim 1	A network device (telephony server 44 in Fig. 1) comprising: a plurality of communication interfaces, including a telephone line interface (**(a) remote subscriber interface module 172 in Fig. 4 and Fig. 1, including telephone interface 102 of Fig. 4 and/or** **(b) telephony service module 170 in Fig. 1))** a computer data interface (**network interface card 43 of Fig. 1**), and a broadband network interface (**internet interface module 45 in Fig. 1**)	System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer. Col. 5, ll. 18-30. Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony server 44 is coupled to one or more central office (CO) lines 48 originating from a public switched telephone network (PSTN) 12 via one or more telephony server modules (TSM) 170. It is not required that telephony server module 170 reside in telephony server 44. Telephony server module 170 may reside in any host computer 40 attached to network 37. CO lines 48 may include Plain Old Telephone Service (POTS), T1, E1, Integrated Services Digital Network (ISDN), Asynchronous Transfer Mode (ATM) or Frame Relay. In addition, telephony server 44 is coupled to one or more internet lines 50 through an internet interface module (IIM) 45." Col. 4, ll. 10-24.
	a processor (**the processor of server 44, which is stated to be implemented on a host PC**)	See e.g., Col. 8, ll. 29-30 ("Telephony server module 170 [on telephony server 44 in one example] communicates with the **host PC** within which it resides through PC interface 78 . . .").
	a machine-readable storage medium (**memory of server 44**) which during use stores	Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33.
	a call processing application (**at least a portion of pbx call processing module 154 of Fig. 6, within telephony server 44 of Fig 1**) and	**call processing application:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone

APPENDIX E1

Baratz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596
		server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."
	service profiles (**database of records stored on server 44, Col. 6 ll. 15-20**),	**service profiles:** "To support internal and external call capability, telephony server 44 maintains a database of user records. Each record holds relevant information such as extension number, user name, class of service, password, etc. Physical extensions are treated independently of logical extension numbers." Figure 1; Col. 6, ll. 16-20.
	and which stores executable instructions to mediate communications between the plurality of communication interfaces (**the portion the code within telephony server 44 that** **(a) directs packets received over network 37 to either internet interface module 45 or telephone service module 170, or** **(b) processes a call**),	**instructions to mediate communications** Voice packets received over network 37 from telephony clients 41 by telephony server 44 are directed to the telephony service module 170 within telephony server 44. Telephony service module 170 then depacketizes and converts the digital data into analog voice data for transmission onto telephone ll. 48. Col. 6, ll. 10-15. Further, the same telephony clients 41 connected to telephony server by network 37 have access to the internet through internet lines 50 or 14 connected to telephony server 44 through the internet interface module 45. Col. 4, ll. 22-28; Col. 7, ll. 12-18; Figs. 1 and 2. Thus, the portion of code within telephony server 44 that receives the voice packets from network 37 and provides them to either the telephony service module 170 or the internet interface module 45 are the instructions to mediate communications. The same analysis applies to telephony client 42, which communicates with telephony server 44 over cable 47, using the remote subscriber interface module 172 of telephony server 44.
	the instructions causing the network device to detect network signaling events or trigger points in a telephone call (**receipt and detection by telephony server 44 of a call, which telephony server 44 then processes**) and	Network server 44, which "provides centralized common management and all necessary resources for providing PBX switching control services" detects calls and in response invokes various processes that are classified as "call processing." Col. 5, ll. 1-3. For example, a feature supplied by system 10 is call switching. During call setup, network addresses are supplied to the called and calling parties by telephony server 44 in response to server 44 detecting the call. Col. 6. ll. 48-53.
	invoke the call processing application in response to the detected network signaling events or trigger points (**PBX call processing module 154 in telephony**	**invoke the call processing application:** At least a portion of PBX call processing module 154 is invoked in response to receipt of packets over network

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596
	server 44, Fig. 6),	37, for example. "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44. Functions include the standard PBX capabilities described above." Col. 10, ll. 20-22. Also, "telephony server 44 provides centralized common management and all necessary resources for providing PBX switching control services." Col. 5, ll. 1-3.
	the call processing application operating according to parameters defined in the service profiles	PBX call processing module 154 operates according to parameters defined in the service profiles. "To support internal and external call capability, telephony server 44 maintains a database of user records. Each record holds relevant information such as extension number, user name, class of service, password, etc. Physical extensions are treated independently of logical extension numbers." Figure 1; Col. 6, ll. 16-20.
	wherein the network device consists of one or more customer premise equipment modules (system 10 or telephony server 44).	System 10 or telephony server 44 consist of one or more customer premise equipment modules. System 10 is a private branch exchange, which is intended for a customer premise. (Abstract)
Claim 2	The network device of claim 1, wherein the plurality of communication interfaces further includes a video streaming device interface.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.
Claim 3	The network device of claim 1, wherein the broadband network interface terminates a broadband network link that joins a customer premises to a packet carrier network.	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596
Claim 4	The network device of claim 1, wherein the instructions further cause the network device to route IP data between the computer data interface and the broadband network interface.	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.
Claim 5	The network device of claim 1, wherein the network device is contained in a single physical enclosure.	Telephony server 44 is contained in a single physical enclosure. Figure 1.
Claim 7	The network device of claim 1, wherein the storage medium during use further stores call routing tables, and the instructions further cause the network device to perform call routing for telephone calls that use the telephone line interface.	"An automatic call distribution component 32 provides automatic call routing in accordance with one of the well known standards such as, automatic number identification (ANI) and calling line identification (CLI). Outside callers need not go through IVR system 34 to reach their party automatic call distribution component 32 utilizes call data transmitted from the CO to determine the called party's extension." Col. 7, ll. 60-66.
Claim 8	The network device of claim 1, wherein the storage medium during use further stores call routing tables, and the instructions further cause the network device to perform call routing for telephone calls according to the call routing tables, the telephone calls using the telephone line interface.	"An automatic call distribution component 32 provides automatic call routing in accordance with one of the well known standards such as, automatic number identification (ANI) and calling line identification (CLI). Outside callers need not go through IVR system 34 to reach their party automatic call distribution component 32 utilizes call data transmitted from the CO to determine the called party's extension." Col. 7, ll. 60-66.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising:	**shared packet network:** network 37 carries digitized voice and control date over network transmitted by telephony servers 44 and telephony clients 41 as well as

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596
	locating a system management platform in a shared packet network (**network 37**), the system management platform collecting call log data from a plurality of network devices ; and	other non-voice data packets originating from other devices on network 37, such file server 46 and hub or router 48. Col. 46-49; Col. 4, ll. 49-53. **system management platform collecting call log data from a plurality of network devices:** Billing system 36, which collects call log data from PBX kernel 10, which includes a plurality of network devices (telephony servers 44). "A billing system 36 provides traditional billing capability to system 10 for tracking telephone usage, call times, call costs, etc." Col. 7, ll. 50-52. Alternatively, or in combination, system administrator 20 provides configuration and administration features, including user configuration, hardware and software additions and deletions, maintaining a database of users, defining class of service, monitoring system status, and generating reports. Col. 7, ll. 28-34.
	distributing the plurality of network devices (**telephony servers 44; pbx kernel 10 includes one or more telephony servers 44**) that each include a telephone line interface (**telephone service module 170**); a computer data interface (**network interface card 43**), a broadband network interface terminating a link from the shared packet network (**internet interface module 45**)	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.
	a processor (**the processor of server 44, which is stated to be implemented on a host PC**)	See e.g., Col. 8, ll. 29-30 ("Telephony server module 170 [on telephony server 44 in one example] communicates with the host PC within which it resides through PC interface 78 . . . ").
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**pbx call processing module 154**),	Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."
	the instructions causing each network device to route telephone calls in a peer-to-	PBX kernel 10, which includes a plurality of network devices (telephony servers

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596
	peer fashion over the shared packet network (**the calls within network 37 occur in a peer to peer fashion**), and to send call log data to the system management platform (**call log data is sent from PBX kernel 10 to billing system 36**).	44), sends call log data to billing system 36. "A billing system 36 provides traditional billing capability to system 10 for tracking telephone usage, call times, call costs, etc." Col. 7, ll. 50-52.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.

APPENDIX E8

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
		6, ll. 16-20.	
	wherein the network device consists of one or more customer premise equipment modules (system 10 or telephony server 44).	System 10 or telephony server 44 consist of one or more customer premise equipment modules. System 10 is a private branch exchange, which is intended for a customer premise. (Abstract)	
Claim 2	The network device of claim 1, wherein the plurality of communication interfaces further includes a video streaming device interface.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.	To the extent that Baratz does not explicitly teach a video streaming device interface, Osterhout (U.S. Patent 7,197,029) teaches interfaces for a telephone that make use of audio, video, and other media. Col. 6, ll. 1-5. Osterhout explains the need for video interfaces in network devices for video conferencing: "The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages or protocols may also be used for video teleconferencing or other applications, if the USB telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize Osterhout's video streaming device interface in Baratz to provide an interface for

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			video conferencing data, for example.
Claim 9	A network device (telephony server 44 in Fig. 1) comprising: a broadband network interface (internet interface module 45 in Fig. 1); a plurality of communication interfaces, including a telephone line interface ((a) remote subscriber interface module 172 in Fig. 4 and Fig. 1, including telephone interface 102 of Fig. 4 and/or (b) telephony service module 170 in Fig. 1)) and a computer data interface (network interface card 43 of Fig. 1);	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony server 44 is coupled to one or more central office (CO) lines 48 originating from a public switched telephone network (PSTN) 12 via one or more telephony server modules (TSM) 170. It is not required that telephony server module 170 reside in telephony server 44. Telephony server module 170 may reside in any host computer 40 attached to network 37. CO lines 48 may include Plain Old Telephone Service (POTS), T1, E1, Integrated Services Digital Network (ISDN), Asynchronous Transfer Mode (ATM) or Frame Relay. In addition, telephony server 44 is coupled to one or more internet lines 50 through an internet interface module (IIM) 45." Col. 4, ll. 10-24. Telephony server modules and their associated software may	

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
		6, ll. 16-20.	
	~~wherein the network device~~ consists of one or more customer premise equipment modules (system 10 or telephony server 44).	System 10 or telephony server 44 consist of one or more ~~customer premise equipment modules. System 10~~ is a private ~~branch~~ exchange, which is intended for a customer premise. (Abstract)	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to provide a SIP user agent to represent a telephone that uses the telephone line interface.	**SIP user agent:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network de for establishing a voice-over-packet netw architecture (*e.g.*, system 10 or telephony se 44). **Known Technique**— A person having ordi skill in the art in 2001 would have been w aware of SIP technology since it became Internet Engineering Task Force (IETF) stan in 1999 as RFC 2543. Osterhout teaches a user agent to represent a non-SIP telephone a SIP proxy server that mediates all communications. For example, Osterhout states: "If the criteria met, the control module 126 may set up remainder of the resources necessary to establi SIP-based connection to a recipient teleph device 120. The control module may invoke module 122 and SIP stack 124 to transmit, rec parse SIP commands, a Transfer Con Protocol/Internet Protocol (TCP/IP) client 130 Internet or other network interface, and a l Time Protocol (RTP) stack 134 to man streaming media and other information for processing." Col. 4, l. 65 - Col. 5 - l. 6.

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephon device itself may be both POTS and SIP enable in which case the base of the device contains b(telephone (RJ-11 or other) connections plus a network connection or port for SIP, with contrc logic residing in the telephone device and no computer or other host being necessary. In a ye further embodiment, the telephone device may contain control logic and connections for each (POTS, USB and SIP for maximum connectivit) **Improved System**—A person having ordi skill in the art in 2001 would have considere obvious to modify the base system of Barat: include these well-known claimed components and apply the well-known technic taught by Osterhout, for example, to enable telephones in *Baratz* to participate in SIP-b telephony systems.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (network 37), the system management platform collecting call log data from a plurality of network devices ; and	**shared packet network:** network 37 carries digitized voice and control date over network transmitted by telephony servers 44 and telephony clients 41 as well as other non-voice data packets originating from other devices on network 37, such file server 46 and hub or router 48. Col. 46-49; Col. 4, ll. 49-53. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony	

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
Claim 9	A network device (telephony server 44 in Fig. 1) comprising: a broadband network interface (internet interface module 45 in Fig. 1); a plurality of communication interfaces, including a telephone line interface (**(a) remote subscriber interface module 172 in Fig. 4 and Fig. 1, including telephone interface 102 of Fig. 4 and/or** **(b) telephony service module 170 in Fig. 1))** and a computer data interface (**network interface card 43 of Fig. 1**);	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30. Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony server 44 is coupled to one or more central office (CO) lines 48 originating from a public switched telephone network (PSTN) 12 via one or more telephony server modules (TSM) 170. It is not required that telephony server module 170 reside in telephony server 44. Telephony server module 170 may reside in any host computer 40 attached to network 37. CO lines 48 may include Plain Old Telephone Service (POTS), T1, E1, Integrated Services Digital Network (ISDN), Asynchronous Transfer Mode (ATM) or Frame Relay. In addition, telephony server 44 is coupled to one or more internet lines 50 through an internet interface module (IIM) 45." Col. 4, ll. 10-24.	

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	a processor (the processor of server 44, which is stated to be implemented on a host PC)	See e.g., Col. 8, ll. 29-30 ("Telephony server module 170 [on telephony server 44 in one example] communicates with the host PC within which it resides through PC interface 78 . . . ").	
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents **(memory of server 44)**	Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33.	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface **(at least a portion of pbx call processing module 154 of Fig. 6, within telephony server 44 of Fig 1)**, and the instructions further causing the network device to implement a SIP proxy server **(SIP stack 124)** that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	**SIP user agent:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ." **SIP proxy server:** Voice packets received over network 37 from telephony clients 41 by telephony server 44 are directed to the telephony service module 170 within telephony server 44. Telephony service module 170 then depacketizes and converts the digital data into analog voice data for transmission onto telephone lines 48. Col. 6, ll. 10-15. Further, the same telephony clients 41 connected to telephony server by network 37 have access to the internet through internet lines 50 or 14 connected to telephony server 44 through the internet interface module 45. Col. 4, ll. 22-28; Col. 7, ll. 12-18; Figs. 1 and 2.	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Osterhout teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Osterhout states: "If the criteria are met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephone device itself may be both POTS and SIP enabled, in which case the base of the device contains both telephone (RJ-11 or other) connections plus a network connection or port for SIP, with control logic residing in the telephone device and no computer or other host being necessary. In a yet further embodiment, the telephone device may contain control logic and connections for each of POTS, USB and SIP for maximum connectivity." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed,	

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
		it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.	To the extent that Baratz does not explicitly teach a video streaming device interface, Osterhout (U.S. Patent 7,197,029) teaches interfaces for a telephone that make use of audio, video, and other media. Col. 6, ll. 1-5. Osterhout explains the need for video interfaces in network devices for video conferencing: "The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages or protocols may also be used for video teleconferencing or other applications, if the USB telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize Osterhout's video streaming device interface in Baratz to provide an interface for video conferencing data, for example.

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	Telephony server 44 is contained in a single physical enclosure.	To the extent that Baratz does not explicitly teach that the network device is contained in a single physical enclosure, the use of a single physical enclosure was well known in the art by 2001. For example, Osterhout teaches providing a call processing application and communication interfaces in a single physical enclosure. Figure 1. Therefore the use of a single physical enclosure would have been a simple design choice to one of ordinary skill in the art.

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	module 170); a computer data interface (network interface card 43); a broadband network interface terminating a link from the shared packet network (internet interface module 45)	telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.	
	a processor (the processor of server 44, which is stated to be implemented on a host PC)	See e.g., Col. 8, ll. 29-30 ("Telephony server module 170 [on telephony server 44 in one example] communicates with the host PC within which it resides through PC interface 78 . . . ").	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (pbx call processing module 154);	Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet network (the calls within network 37 occur in a peer to peer fashion). and to send call log data to the system management platform (call log data is sent from PBX kernel 10 to billing system 36).	PBX kernel 10, which includes a plurality of network devices (telephony servers 44), sends call log data to billing system 36. "A billing system 36 provides traditional billing capability to system 10 for tracking telephone usage, call times, call costs, etc." Col. 7, ll. 50-52.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	**SIP signaling:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network de for establishing a voice-over-packet netw architecture (*e.g.*, system 10 or telephony se 44). **Known Technique**— A person having ordi skill in the art in 2001 would have been w aware of SIP technology since it became

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			Internet Engineering Task Force (IETF) stan in 1999 as RFC 2543. Osterhout teaches a user agent to represent a non-SIP telephone a SIP proxy server that mediates all communications. For example, Osterhout states: "If the criteria met, the control module 126 may set up remainder of the resources necessary to establi SIP-based connection to a recipient teleph device 120. The control module may invoke module 122 and SIP stack 124 to transmit, rec parse SIP commands, a Transfer Con Protocol/Internet Protocol (TCP/IP) client 130 Internet or other network interface, and a I Time Protocol (RTP) stack 134 to man streaming media and other information for processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephon device itself may be both POTS and SIP enable in which case the base of the device contains b telephone (RJ-11 or other) connections plus a network connection or port for SIP, with contr logic residing in the telephone device and no computer or other host being necessary. In a ye further embodiment, the telephone device may contain control logic and connections for each POTS, USB and SIP for maximum connectivit **Improved System**—A person having ordi skill in the art in 2001 would have considere obvious to modify the base system of Bara

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			include these well-known claimed components and apply the well-known technic taught by Osterhout, for example, to enable telephones in *Baratz* to participate in SIP-b telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	**SIP proxy server:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network de for establishing a voice-over-packet netw architecture (*e.g.*, system 10 or telephony se 44). **Known Technique**— A person having ordi skill in the art in 2001 would have been w aware of SIP technology since it became Internet Engineering Task Force (IETF) stand in 1999 as RFC 2543. Osterhout teaches a user agent to represent a non-SIP telephone a SIP proxy server that mediates all communications. For example, Osterhout states: "If the criteria met, the control module 126 may set up remainder of the resources necessary to establi SIP-based connection to a recipient teleph device 120. The control module may invoke module 122 and SIP stack 124 to transmit, rec parse SIP commands, a Transfer Con Protocol/Internet Protocol (TCP/IP) client 130 Internet or other network interface, and a I Time Protocol (RTP) stack 134 to man streaming media and other information for

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephon device itself may be both POTS and SIP enable in which case the base of the device contains b telephone (RJ-11 or other) connections plus a network connection or port for SIP, with contr logic residing in the telephone device and no computer or other host being necessary. In a ye further embodiment, the telephone device may contain control logic and connections for each POTS, USB and SIP for maximum connectivit **Improved System**—A person having ordi skill in the art in 2001 would have considere obvious to modify the base system of Barat include these well-known claimed components and apply the well-known technic taught by Osterhout, for example, to enable telephones in *Baratz* to participate in SIP-b telephony systems.

Baratz in view of Osterhout

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	a processor (the processor of server 44, which is stated to be implemented on a host PC)	See e.g., Col. 8, ll. 29-30 ("Telephony server module 170 [on telephony server 44 in one example] communicates with the host PC within which it resides through PC interface 78 . . . ").	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (pbx call processing module 154),	Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet network (the calls within network 37 occur in a peer to peer fashion). and to send call log data to the system management platform (call log data is sent from PBX kernel 10 to billing system 36).	PBX kernel 10, which includes a plurality of network devices (telephony servers 44), sends call log data to billing system 36. "A billing system 36 provides traditional billing capability to system 10 for tracking telephone usage, call times, call costs, etc." Col. 7, ll. 50-52.	
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.	To the extent that Baratz does not explicitly teach a video streaming device interface, Osterhout (U.S. Patent 7,197,029) teaches interfaces for a telephone that make use of audio, video, and other media. Col. 6, ll. 1-5. Osterhout explains the need for video interfaces in network devices for video conferencing: "The native media applications may likewise include an audio/visual module 134b, such as an audio management tool such as an MP3 codec, RealAudio or other package. A video management tool such as Avid, RealVideo or other packages or protocols may also be used for video teleconferencing or other applications, if the USB

Baratz in view of Osterhout

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			telephone 102, host computer 106 or other resources are equipped with video input. Video or combined audio/video streams again may be output over data network or telephony links. Other multimedia applications are possible." Col. 5, l. 62 - Col. 6, l. 5. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize Osterhout's video streaming device interface in Baratz to provide an interface for video conferencing data, for example.

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	wherein the network device consists of one or more customer premise equipment modules (system 10 or telephony server 44).	System 10 or telephony server 44 consist of one or more customer premise equipment modules. System 10 is a private branch exchange, which is intended for a customer premise. (Abstract)	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to provide a SIP user agent to represent a telephone that uses the telephone line interface.	**SIP user agent:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (network 37), the system management platform collecting call log data from a plurality of network devices ; and	**shared packet network:** network 37 carries digitized voice and control date over network transmitted by telephony servers 44 and telephony clients 41 as well as other non-voice data packets originating from other devices on network 37, such file server 46 and hub or router 48. Col. 46-49; Col. 4, ll. 49-53. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony	

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
Claim 9	A network device (telephony server 44 in Fig. 1) comprising: a broadband network interface (internet interface module 45 in Fig. 1); a plurality of communication interfaces, including a telephone line interface ((a) remote subscriber interface module 172 in Fig. 4 and Fig. 1, including telephone interface 102 of Fig. 4 and/or (b) telephony service module 170 in Fig. 1)) and a computer data interface (network interface card 43 of Fig. 1);	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30. Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony server 44 is coupled to one or more central office (CO) lines 48 originating from a public switched telephone network (PSTN) 12 via one or more telephony server modules (TSM) 170. It is not required that telephony server module 170 reside in telephony server 44. Telephony server module 170 may reside in any host computer 40 attached to network 37. CO lines 48 may include Plain Old Telephone Service (POTS), T1, E1, Integrated Services Digital Network (ISDN), Asynchronous Transfer Mode (ATM) or Frame Relay. In addition, telephony server 44 is coupled to one or more internet lines 50 through an internet interface module (IIM) 45." Col. 4, ll. 10-24.	

Office Action Appendix
Control Number 95/000,344 4-2-2009 Part of Paper No. 20090402
APPENDIX E3

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	a processor (the processor of server 44, which is stated to be implemented on a host PC)	See e.g., Col. 8, ll. 29-30 ("Telephony server module 170 [on telephony server 44 in one example] communicates with the host PC within which it resides through PC interface 78 . . . ").	
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents (memory of server 44)	Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33.	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface (at least a portion of pbx call processing module 154 of Fig. 6, within telephony server 44 of Fig 1), and the instructions further causing the network device to implement a SIP proxy server (SIP stack 124) that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	SIP user agent: Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ." SIP proxy server: Voice packets received over network 37 from telephony clients 41 by telephony server 44 are directed to the telephony service module 170 within telephony server 44. Telephony service module 170 then depacketizes and converts the digital data into analog voice data for transmission onto telephone lines 48. Col. 6, ll. 10-15. Further, the same telephony clients 41 connected to telephony server by network 37 have access to the internet through internet lines 50 or 14 connected to telephony server 44 through the internet interface module 45. Col. 4, ll. 22-28; Col. 7, ll. 12-18; Figs. 1 and 2.	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to	

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
		telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.	
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	Telephony server 44 is contained in a single physical enclosure.	To the extent that Baratz does not explicitly teach that the network device is contained in a single physical enclosure, the use of a single physical enclosure was well known in the art by 2001. For example, Wengrovitz teaches providing a call processing application and communication interfaces in a single physical enclosure. Figure 2. Therefore the use of a single physical enclosure would have been a simple design choice to one of ordinary skill in the art.

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'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	route telephone calls in a peer-to-peer fashion over the shared packet network (the calls within network 37 occur in a peer to peer fashion). and to send call log data to the system management platform (call log data is sent from PBX kernel 10 to billing system 36).	(telephony servers 44), sends call log data to billing system 36. "A billing system 36 provides traditional billing capability to system 10 for tracking telephone usage, call times, call costs, etc." Col. 7, ll. 50-52.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	**SIP signaling:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21.

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	**SIP proxy server:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." **Improved System**—A person having ordinary

Baratz in view of Wengrovitz

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	wherein the network device consists of one or more customer premise equipment modules (system 10 or telephony server 44)	System 10 or telephony server 44 consist of one or more customer premise equipment modules. System 10 is a private branch exchange, which is intended for a customer premise. (Abstract)	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to provide a SIP user agent to represent a telephone that uses the telephone line interface.	**SIP user agent:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6.

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (network 37), the system management platform collecting call log data from a plurality of network devices ; and	**shared packet network:** network 37 carries digitized voice and control date over network transmitted by telephony servers 44 and telephony clients 41 as well as other non-voice data packets originating from other devices on network 37, such file server 46 and hub or router 48. Col. 46-49; Col. 4, ll. 49-53. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony server 44 is coupled to one or more central office (CO) lines 48 originating from a public switched telephone network	

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
Claim 9	A network device (telephony server 44 in Fig. 1) comprising: a broadband network interface (internet interface module 45 in Fig. 1); a plurality of communication interfaces, including a telephone line interface ((a) remote subscriber interface module 172 in Fig. 4 and Fig. 1, including telephone interface 102 of Fig. 4 and/or (b) telephony service module 170 in Fig. 1)) and a computer data interface (network interface card 43 of Fig. 1);	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed, it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30. Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33. "On the network side, telephony server 44 is coupled to network 37 via network interface cards (NIC) 43 installed in a host computer 40. On the telephone network side, telephony server 44 is coupled to one or more central office (CO) lines 48 originating from a public switched telephone network (PSTN) 12 via one or more telephony server modules (TSM) 170. It is not required that telephony server module 170 reside in telephony server 44. Telephony server module 170 may reside in any host computer 40 attached to network 37. CO lines 48 may include Plain Old Telephone Service (POTS), T1, E1, Integrated Services Digital Network (ISDN), Asynchronous Transfer Mode (ATM) or Frame Relay. In addition, telephony server 44 is coupled to one or more internet lines 50 through an internet interface module (IIM) 45." Col. 4, ll. 10-24.	

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	a processor (the processor of server 44, which is stated to be implemented on a host PC)	See e.g., Col. 8, ll. 29-30 ("Telephony server module 170 [on telephony server 44 in one example] communicates with the host PC within which it resides through PC interface 78 . . . ").	
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents (memory of server 44)	Telephony server modules and their associated software may be installed in any host computer attached to network 37. Figure 1; Col. 5, ll. 31-33.	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface (at least a portion of pbx call processing module 154 of Fig. 6, within telephony server 44 of Fig 1), and the instructions further causing the network device to implement a SIP proxy server (SIP stack 124) that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	SIP user agent: Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ." SIP proxy server: Voice packets received over network 37 from telephony clients 41 by telephony server 44 are directed to the telephony service module 170 within telephony server 44. Telephony service module 170 then depacketizes and converts the digital data into analog voice data for transmission onto telephone lines 48. Col. 6, ll. 10-15. Further, the same telephony clients 41 connected to telephony server by network 37 have access to the internet through internet lines 50 or 14 connected to telephony server 44 through the internet interface module 45. Col. 4, ll. 22-28; Col. 7, ll. 12-18; Figs. 1 and 2.	Base System—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). Known Technique— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control.

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	"System 10 also supports regular telephone sets 42 that are not coupled to a host computer. In this case, referring to FIG. 1, telephone 42 is coupled directly to a remote subscriber interface module 172, installed in a host computer, via a traditional cable line 47, independent of network 37. Remote subscriber interface module 172 provides PBX services to telephones lacking a host computer and connects them to network 37. Depending on the type of LAN cabling installed,	

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
		it might be possible to piggy back line 47 onto an unused pair within the LAN cabling. For example, when using 4 pair cable to wire an Ethernet LAN, two spare pairs are available. These spare pairs may be used to connect the telephone sets that are not connected to a host computer." Col. 5, ll. 18-30.	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	Non-voice traffic, e.g., voice data processed by server 44. "Alternatively, control port 16 may be accessed through network 37. A voice port 18 includes a physical port and associated software for use by external devices or systems. Such applications include ATM, Frame Relay, other network transport or switching equipment or even an audio device such as a tape recorder, radio, etc." Col. 7, ll. 21-26.	
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	Telephony server 44 is contained in a single physical enclosure.	To the extent that Baratz does not explicitly teach that the network device is contained in a single physical enclosure, the use of a single physical enclosure was well known in the art by 2001. For example, Girard teaches providing a call processing application and communication interfaces in a single physical enclosure. Figure 2. Therefore the use of a single physical enclosure would have been a simple design choice to one of ordinary skill in the art.

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	network 37 occur in a peer to peer fashion). and to send call log data to the system management platform (call log data is sent from PBX kernel 10 to billing system 36).	capability to system 10 for tracking telephone usage, call times, call costs, etc." Col. 7, ll. 50-52.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	**SIP signaling:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6.

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Baratz* to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	**SIP proxy server:** Referring to Figure 6, "One level up, is a PBX call processing module which functions to control all call processes within telephone server 44." Col. 10, ll. 20-22. Figure 6 shows "the software architecture for telephony server 44 . . . ."	**Base System**—Baratz discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, system 10 or telephony server 44). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Baratz to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Baratz* to participate in SIP-based

Baratz in view of Girard-SIP

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
			telephony systems.

Baratz in view of Chung

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	wherein the network device consists of one or more customer premise equipment modules (system 10 or telephony server 44).	System 10 or telephony server 44 consist of one or more customer premise equipment modules. System 10 is a private branch exchange, which is intended for a customer premise. (Abstract)	
Claim 7	The network device of claim 1, wherein the storage medium during use further stores call routing tables, and the instructions further cause the network device to perform call routing for telephone calls that use the telephone line interface.	"An automatic call distribution component 32 provides automatic call routing in accordance with one of the well known standards such as, automatic number identification (ANI) and calling line identification (CLI). Outside callers need not go through IVR system 34 to reach their party automatic call distribution component 32 utilizes call data transmitted from the CO to determine the called party's extension." Col. 7, ll. 60-66.	To the extent that Baratz does not explicitly teach call routing tables, Chung (U.S. Patent 6584108) teaches call routing tables. "The extra digits are passed on to the private branch exchange which will use them to connect the call to the correct extension. Call routing is supported via a static mapping table in each MAC, but the embodiment is not so limited." (Col. 16, ll. 14-18). Chung teaches the use of call routing tables in order to efficiently route telephone calls and avoid the need for call routing through the private branch exchange. Col. 16., ll. 41-50. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize call routing tables as taught by Chung (U.S. Patent 6584108) to enable the network device telephones of Baratz to efficiently route telephone calls, for example.
Claim 8	The network device of claim 1, wherein the storage medium during use further stores call	"An automatic call distribution component 32 provides automatic call routing in accordance with one of the well	To the extent that Baratz does not explicitly teach

Baratz in view of Chung

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	routing tables, and the instructions further cause the network device to perform call routing for telephone calls according to the call routing tables, the telephone calls using the telephone line interface.	known standards such as, automatic number identification (ANI) and calling line identification (CLI). Outside callers need not go through IVR system 34 to reach their party automatic call distribution component 32 utilizes call data transmitted from the CO to determine the called party's extension." Col. 7, ll. 60-66.	call routing tables, Chung (U.S. Patent 6584108) teaches call routing tables. "The extra digits are passed on to the private branch exchange which will use them to connect the call to the correct extension. Call routing is supported via a static mapping table in each MAC, but the embodiment is not so limited." (Col. 16, ll. 14-18). Chung teaches the use of call routing tables in order to efficiently route telephone calls and avoid the need for call routing through the private branch exchange. Col. 16., ll. 41-50. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize call routing tables as taught by Chung (U.S. Patent 6584108) to enable the network device telephones of Baratz to efficiently route telephone calls, for example.

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APPENDIX E10

Baratz in view of Czajkowski

'519 Claim	Claim Limitations	Baratz U.S. Patent 5,742,596	Secondary References
	wherein the network device consists of one or more customer premise equipment modules (system 10 or telephony server 44).	System 10 or telephony server 44 consist of one or more customer premise equipment modules. System 10 is a private branch exchange, which is intended for a customer premise. (Abstract)	
Claim 5	The network device of claim 1, wherein the network device is contained in a single physical enclosure.	Telephony server 44 is contained in a single physical enclosure. Figure 1.	To the extent that Baratz does not explicitly teach that the network device is contained in a single physical enclosure, the use of a single physical enclosure was well known in the art by 2001. For example, Czajkowski (U.S. Patent 6,526,058) teaches providing a call processing application and communication interfaces in a single physical enclosure. Figure 2; Col. 7, ll. 25-27. Therefore the use of a single physical enclosure would have been a simple design choice to one of ordinary skill in the art.

Control Number 95/000,344 | **Office Action Appendix 4-2-2009** | **Part of Paper No. 20090402**

APPENDIX F1

Czajkowski

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058
Claim 1	A network device (CPE 2 which may include the computer functionality of PC 3) comprising: a plurality of communication interfaces, including a telephone line interface (borscht circuit 52), a computer data interface (ethernet interface 23), and a broadband network interface (xDSL modem 21);	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1; Col. 5, ll. 23-34. Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1. BORSCHT circuit 25 is a telephone line interface. "The BORSCHT circuit (Battery, Over-voltage protection, Ringing, Supervision, Coding, Hybrid, Test) is a known function which provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3. CPE 2: The software 61 and control processor 25 described above may reside in a variety of locations. "This computer functionality can be contained within the CPE 2, the telephones 4a and 4b, or within the wider area network outside of the customer premises." Figure 2; Col. 7, ll. 25-27.
	a processor; (processor of CPE 2 or processor in PC 3)	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)

Czajkowski

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058
	a machine-readable storage medium (memory of CPE 2) which during use stores	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34.
	a call processing application (application 61) and	**call processing application:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60).
	service profiles (application 61 includes pre-determined service conditions),	**service profiles:** Application 61 includes pre-determined service conditions. "This service allows the customer to control routing of calls to telephone lines attached to the CPE2, based on some pre-determined condition that is known to the software 61 on the PC 3. This might include routing according to the time of day (e.g. diverting calls from a child's line to an adult's line after a certain time) or diverting a call from one line to another based on known absence of the recipient or location of the recipient elsewhere on the premises." Figure 2; Col. 9, ll. 43-50. Application 61 provides service profiles associated with voice mail services. "This application provides an extended voice service that can be implemented by the PC attached to the data interface. It is applicable as an extended possible scenario for both examples 1 and 2 above. The PC is used to provide a local and private (to the customer premises) voice-mail facility." Figure 2; Col. 12, ll. 3-8.
	and which stores executable instructions to mediate communications between the plurality of communication interfaces (control processor 25),	**instructions to mediate communications** Control processor 25 creates ATM virtual circuits for transferring packet switched

Czajkowski

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058
		cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2: Col. 6, ll. 35-39. "The control processor 25 is arranged to enable and disable virtual circuits by controlling the ATM switch 22." Figure 2: Col. 6, ll. 55-56. The software 61 and control processor 25 described above may reside in a variety of locations. "This computer functionality can be contained within the CPE 2, the telephones 4a and 4b, or within the wider area network outside of the customer premises." Figure 2; Col. 7, ll. 25-27.
	the instructions causing the network device to detect network signaling events or trigger points in a telephone call (receipt and detection by control processor 25 of a telephone call) and	Control processor 25 detects network signaling events or trigger points in a telephone call. "The control processor is also arranged to identify hook switch events on any of the subscriber's phones 4, and additionally receives ringing messages from the exchange and signals the BORSCHT function to provide a ringing signal when appropriate to each of the voice phones 4." Figure 2; Col. 6, ll. 56-61. "An incoming voice call is signaled to the CPE 2 by an AAL2 cell which has an indicator field set to indicate that it is a control message. This is switched by the ATM/AAL2 switch 22 to the control processor 25." Figure 2; Col. 9, ll. 54-57.
	invoke the call processing application in response to the detected network signaling events or trigger points (control processor 25 sends a message to application 61),	The control processor 25 invokes application 61 in response to the detected network signaling events or trigger points. "The control processor 25 informs the PC 3 of the incoming call details by means of a message across the API 55. The PC application 61 listening on the appropriate port for the API 55 responds with an acknowledgement. In the absence of such an acknowledgement, the control processor 25 is configured to recognise that there is not an interested PC application 61 live on the PC (or that the PC is not on), and routes the incoming call to the designated telephone line immediately." Figure 2; Col. 9, ll. 57-65.

Czajkowski

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058
	the call processing application operating according to parameters defined in the service profiles **(application 61 controls the call based on pre-determined service conditions)**	Application 61 operates according to service conditions. "Once the CPE control processor 25 has received an acknowledgement from the PC application 61, it waits for a predetermined time-out period for further instructions from the PC 3 (say up to one or two seconds). During this time, the PC application 61 determines the action required given the information on the incoming call, such as a re-routing because of the current time. The PC application 61 then sends a recognised instruction across the API 55 to the control processor 25 on the CPE. The control processor 25 then completes the connection of the incoming call, routing it as appropriate." Figure 2; Col. 9, ll. 66 - Col. 10, ll. 9.
	wherein the network device consists of one or more customer premise equipment modules **(CPE 1, PC 3, and telephony appliances 4)**.	CPE 2, PC 3, and telephony appliances 4 consist of one or more customer premise equipment modules. (ABSTRACT).
Claim 2	The network device of claim 1, wherein the plurality of communication interfaces further includes a video streaming device interface.	"Telephone access networks have historically always been connection orientated, typically using dedicated copper wire circuits between subscribers and the exchange or central office. Typically these access networks were designed to carry only voice with a bandwidth of less than 4 kHz. However in recent years with the growth of the internet and demand for other multi-media services such as video-on-demand and video conferencing, subscribers of telephone companies have demanded additional bandwidth over the access network to provide adequate Internet and multi-media services access." Col. 1, ll. 18-27.
Claim 3	The network device of claim 1, wherein the broadband network interface terminates a broadband network link that joins a customer premises to a packet carrier network.	Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1.
Claim 4	The network device of claim 1, wherein the instructions further cause the network device to route IP data between the computer data interface and the broadband network interface.	Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS

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'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058
		lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1.
Claim 5	The network device of claim 1, wherein the network device is contained in a single physical enclosure.	CPE 2: The software 61 and control processor 25 described above may reside in a variety of locations. "This computer functionality can be contained within the CPE 2, the telephones 4a and 4b, or within the wider area network outside of the customer premises." Figure 2; Col. 7, ll. 25-27.
Claim 7	The network device of claim 1, wherein the storage medium during use further stores call routing tables, and the instructions further cause the network device to perform call routing for telephone calls that use the telephone line interface.	"The following is a non-exhaustive list of services which can be implemented using the inventive methods described herein: Time-dependent or predetermined re-routing of incoming call Routing of incoming call based on the CLI information." Col. 8, ll. 56-61.
Claim 8	The network device of claim 1, wherein the storage medium during use further stores call routing tables, and the instructions further cause the network device to perform call routing for telephone calls according to the call routing tables, the telephone calls using the telephone line interface.	"The following is a non-exhaustive list of services which can be implemented using the inventive methods described herein: Time-dependent or predetermined re-routing of incoming call Routing of incoming call based on the CLI information." Col. 8, ll. 56-61.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network **(control processor 25 sends messages across the API 55 to an application 61 on the PC 3)**, the system management platform collecting call log data from a plurality of network devices ; and	"In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29. "The flow diagram for the specific example 3 shown in FIG. 5 is just one possible service, in which the PC application makes a log of the number being called. This example is kept simple to illustrate the principle, and it will be appreciated that additional messages could be sent by the control processor 25 to allow the PC application to determine for example the duration of the call, etc." Col. 11, ll. 30-

Czajkowski in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	more customer premise equipment modules (CPE 1, PC 3, and telephony appliances 4).	more customer premise equipment modules. (ABSTRACT).	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to provide a SIP user agent to represent a telephone that uses the telephone line interface.	**SIP user agent:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60).	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Osterhout teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Osterhout states: "If the criteria are met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephone device itself may be both POTS and SIP enabled, in which case the base of the device contains both

Czajkowski in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			telephone (RJ-11 or other) connections plus a network connection or port for SIP, with control logic residing in the telephone device and no computer or other host being necessary. In a yet further embodiment, the telephone device may contain control logic and connections for each of POTS, USB and SIP for maximum connectivity." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Czajkowski to participate in SIP-based telephony systems.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**), the system management platform collecting call log data from a plurality of network devices ; and	"In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29. "The flow diagram for the specific example 3 shown in FIG. 5 is just one possible service, in which the PC application makes a log of the number being called. This example is kept simple to illustrate the principle, and it will be appreciated	

Czajkowski in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 9	A network device (**CPE 2 which may include the computer functionality of PC 3**) comprising: a broadband network interface (**xDSL modem 21**); a plurality of communication interfaces, including a telephone line interface (**borscht circuit 52**) and a computer data interface (**ethernet interface 23**), and;	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34. Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1. BORSCHT circuit 25 is a telephone line interface. "The BORSCHT circuit (Battery, Over-voltage protection, Ringing, Supervision, Coding, Hybrid, Test) is a known function which provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	

Czajkowski in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	a processor (**processor of CPE 2 or processor in PC 3**);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents (**memory of CPE 2**)	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34.	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface (**application 61**), and the instructions further causing the network device to implement a SIP proxy server (**control processor 25**) that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	**SIP user agent:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60). **SIP proxy server:** Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22.	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Osterhout teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Osterhout states: "If the criteria are

Czajkowski in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephone device itself may be both POTS and SIP enabled, in which case the base of the device contains both telephone (RJ-11 or other) connections plus a network connection or port for SIP, with control logic residing in the telephone device and no computer or other host being necessary. In a yet further embodiment, the telephone device may contain control logic and connections for each of POTS, USB and SIP for maximum connectivity." Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Czajkowski to participate in SIP-based telephony systems.

Czajkowski in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34.	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	"Telephone access networks have historically always been connection orientated, typically using dedicated copper wire circuits between subscribers and the exchange or central office. Typically these access networks were designed to carry only voice with a bandwidth of less than 4 kHz. However in recent years with the growth of the internet and demand for other multi-media services such as video-on-demand and video conferencing, subscribers of telephone companies have demanded additional bandwidth over the access network to provide adequate Internet and multi-media services access." Col. 1, ll. 17-27.	
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical	CPE 2: The software 61 and control processor 25 described above may reside in a variety of locations. "This computer	

APPENDIX F2

Czajkowski in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	enclosure.	functionality can be contained within the CPE 2, the telephones 4a and 4b, or within the wider area network outside of the customer premises." Figure 2; Col. 7, ll. 25-27.	

APPENDIX F6

Czajkowski in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	more customer premise ~~equipment modules (CPE 1, PC 3, and telephony appliances 4).~~	more customer premise equipment modules ~~(ABSTRACT).~~	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to provide a SIP user agent to represent a telephone that uses the telephone line interface.	SIP user agent: Application 61 performs a list of call processing services. (FIGURE 2; 8/60).	Base System—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). Known Technique— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and

Czajkowski in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**), the system management platform collecting call log data from a plurality of network devices ; and	"In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29. "The flow diagram for the specific example 3 shown in FIG. 5 is just one possible service, in which the PC application	

Czajkowski in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 9	A network device (**CPE 2 which may include the computer functionality of PC 3**) comprising: a broadband network interface (**xDSL modem 21**); a plurality of communication interfaces, including a telephone line interface (**borscht circuit 52**) and a computer data interface (**ethernet Interface 23**), and;	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34. Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1. BORSCHT circuit 25 is a telephone line interface. "The BORSCHT circuit (Battery, Over-voltage protection, Ringing, Supervision, Coding, Hybrid, Test) is a known function which provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	

APPENDIX F3

Czajkowski in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	a processor (processor of CPE 2 or processor in PC 3);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents (memory of CPE 2)	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34.	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface (application 61), and the instructions further causing the network device to implement a SIP proxy server (control processor 25) that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	SIP user agent: Application 61 performs a list of call processing services. (FIGURE 2; 8/60). SIP proxy server: Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22.	Base System—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). Known Technique— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching

APPENDIX F3

Czajkowski in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Czajkowski* to participate in SIP-

Czajkowski in view of Wengrovitz

'519 Claims	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			based telephony systems.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34.	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	"Telephone access networks have historically always been connection orientated, typically using dedicated copper wire circuits between subscribers and the exchange or central office. Typically these access networks were designed to carry only voice with a bandwidth of less than 4 kHz. However in recent years with the growth of the internet and demand for other multi-media services such as video-on-demand and video conferencing, subscribers of telephone companies have demanded additional bandwidth over the access network to provide adequate Internet and multi-media services access." Col. 1, ll. 17-27.	

Czajkowski in view of Wengrovitz

'519 Claims	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	CPE 2: The software 61 and control processor 25 described above may reside in a variety of locations. "This computer functionality can be contained within the CPE 2, the telephones 4a and 4b, or within the wider area network outside of the customer premises." Figure 2; Col. 7, ll. 25-27.	

Czajkowski in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	~~more customer premise equipment modules (CPE 1, PC 3, and telephony appliances 4).~~	~~more customer premise equipment modules. (ABSTRACT).~~	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to provide a SIP user agent to represent a telephone that uses the telephone line interface.	**SIP user agent:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60).	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls:

Czajkowski in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			"this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**), the system management platform collecting call log data from a plurality of network devices ; and	"In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29. "The flow diagram for the specific example 3 shown in FIG. 5 is just one possible service, in which the PC application makes a log of the number being called. This example is kept simple to illustrate the principle, and it will be appreciated	

Czajkowski in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 9	A network device (**CPE 2 which may include the computer functionality of PC 3**) comprising: a broadband network interface (**xDSL modem 21**); a plurality of communication interfaces, including a telephone line interface (**borscht circuit 52**) and a computer data interface (**ethernet interface 23**), and;	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34. Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1. BORSCHT circuit 25 is a telephone line interface. "The BORSCHT circuit (Battery, Over-voltage protection, Ringing, Supervision, Coding, Hybrid, Test) is a known function which provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	

Czajkowski in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	a processor (**processor of CPE 2 or processor in PC 3**);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents (**memory of CPE 2**)	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34.	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface (**application 61**), and the instructions further causing the network device to implement a SIP proxy server (**control processor 25**) that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	**SIP user agent**: Application 61 performs a list of call processing services. (FIGURE 2; 8/60). **SIP proxy server**: Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22.	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC

Czajkowski in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.

Czajkowski in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34.	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	"Telephone access networks have historically always been connection orientated, typically using dedicated copper wire circuits between subscribers and the exchange or central office. Typically these access networks were designed to carry only voice with a bandwidth of less than 4 kHz. However in recent years with the growth of the internet and demand for other multi-media services such as video-on-demand and video conferencing, subscribers of telephone companies have demanded additional bandwidth over the access network to provide adequate Internet and multi-media services access." Col. 1, ll. 17-27.	
Claim 12	The network device of claim 9, wherein the	CPE 2: The software 61 and control processor 25 described	

Czajkowski in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	network device is contained in a single physical enclosure.	above may reside in a variety of locations. "This computer functionality can be contained within the CPE 2, the telephones 4a and 4b, or within the wider area network outside of the customer premises." Figure 2; Col. 7, ll. 25-27.	

Czajkowski in view of Chung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	more customer premise ~~equipment modules (CPE 1, PC 3, and telephony appliances 4).~~	~~more customer premise equipment modules. (ABSTRACT).~~	
Claim 7	The network device of claim 1, wherein the storage medium during use further stores call routing tables, and the instructions further cause the network device to perform call routing for telephone calls that use the telephone line interface.	"The following is a non-exhaustive list of services which can be implemented using the inventive methods described herein: Time-dependent or predetermined re-routing of incoming call Routing of incoming call based on the CLI information." Col. 8, ll. 56-61.	To the extent that Czajkowski does not explicitly teach call routing tables, Chung (U.S. Patent 6584108) teaches call routing tables. "The extra digits are passed on to the private branch exchange which will use them to connect the call to the correct extension. Call routing is supported via a static mapping table in each MAC, but the embodiment is not so limited." (Col. 16, ll. 14-18). Chung teaches the use of call routing tables in order to efficiently route telephone calls and avoid the need for call routing through the private branch exchange. Col. 16., ll. 41-50. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize call routing tables as taught by Chung (U.S. Patent 6584108) to enable the network device telephones of Czajkowski to efficiently route telephone calls, for example.
Claim 8	The network device of claim 1, wherein the storage medium during use further stores call routing tables, and the instructions further cause the network device to perform call routing for telephone calls according to the call routing	"The following is a non-exhaustive list of services which can be implemented using the inventive methods described herein: Time-dependent or predetermined re-routing of incoming call Routing of incoming call based on the CLI information." Col. 8, ll. 56-61.	To the extent that Czajkowski does not explicitly teach call routing tables, Chung (U.S. Patent 6584108) teaches call routing tables. "The extra digits are passed on to the

APPENDIX F8

Czajkowski in view of Chung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	tables, the telephone calls using the telephone line interface.		private branch exchange which will use them to connect the call to the correct extension. Call routing is supported via a static mapping table in each MAC, but the embodiment is not so limited." (Col. 16, ll. 14-18). Chung teaches the use of call routing tables in order to efficiently route telephone calls and avoid the need for call routing through the private branch exchange. Col. 16., ll. 41-50. Therefore, it would have been obvious to one skilled in the art at the time the invention was made to utilize call routing tables as taught by Chung (U.S. Patent 6584108) to enable the network device telephones of Czajkowski to efficiently route telephone calls, for example.

APPENDIX F9

Czajkowski in view of Inbar

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**), the system management platform collecting call log data from a plurality of network devices ; and	"In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29. "The flow diagram for the specific example 3 shown in FIG. 5 is just one possible service, in which the PC application makes a log of the number being called. This example is kept simple to illustrate the principle, and it will be appreciated that additional messages could be sent by the control processor 25 to allow the PC application to determine for example the duration of the call, etc." Col. 11, ll. 30-36.	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. Inbar, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Inbar states: "The IPCenter preferably records usage and billing information, and, as described above, reports billing information to the Master-Server, or to a separate billing unit associated with the master server. In addition to usage and billing information, the IPCenter may report Quality-of-Service (QoS) information, and in some cases connectivity monitoring information, status information of connected devices and other information as may be defined." Figure 1; Col. 8, ll. 54-62. "The system preferably further comprises a billing mechanism for accumulating a transaction log at the subscriber end and retrieving data of said log to the master server." Col. 4, ll. 16-19. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include a system management platform, for example, to maintain these records in a centralized system and facilitate billing: "all of these services

Czajkowski in view of Inbar

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			have to be integrated with each other, with a central control and with billing servers and other functions." Col. 1, ll. 41-43.
	distributing the plurality of network devices (**CPE 2 which may include the computer functionality of PC 3**) that each include a telephone line interface (**borscht circuit 52**), a computer data interface (**ethernet interface 23**), a broadband network interface terminating a link from the shared packet network (**xDSL modem 21**)	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34. Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1. BORSCHT circuit 25 is a telephone line interface. "The BORSCHT circuit (Battery, Over-voltage protection, Ringing, Supervision, Coding, Hybrid, Test) is a known function which	

Czajkowski in view of Inbar

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (**processor of CPE 2 or processor in PC 3**);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**application 61**),	**call processing application:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60).	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet. and to send call log data to the system management platform (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**).	**System management platform:** Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. Inbar, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Inbar states: "The IPCenter preferably records usage and billing information, and, as described above, reports billing information to the Master-Server, or to a separate billing unit associated with the master server. In addition to usage and billing information, the IPCenter may report Quality-of-Service (QoS) information, and in some cases connectivity monitoring information, status information of connected devices and other information as may

Czajkowski in view of Inbar

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			be defined." Figure 1; Col. 8, ll. 54-62. "The system preferably further comprises a billing mechanism for accumulating a transaction log at the subscriber end and retrieving data of said log to the master server." Col. 4, ll. 16-19. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include a system management platform, for example, to maintain these records in a centralized system and facilitate billing: "all of these services have to be integrated with each other, with a central control and with billing servers and other functions." Col. 1, ll. 41-43.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1."	
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple	

Czajkowski in view of Inbar

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1; Col. 5, ll. 23-34.	
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	"Telephone access networks have historically always been connection orientated, typically using dedicated copper wire circuits between subscribers and the exchange or central office. Typically these access networks were designed to carry only voice with a bandwidth of less than 4 kHz. However in recent years with the growth of the internet and demand for other multi-media services such as video-on-demand and video conferencing, subscribers of telephone companies have demanded additional bandwidth over the access network to provide adequate Internet and multi-media services access."	

APPENDIX F9

Czajkowski in view of Inbar

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		Col. 1, ll. 17-27.	

APPENDIX F5

Czajkowski in view of Inbar and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (**processor of CPE 2 or processor in PC 3**);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**application 61**),	**call processing application:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60).	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet. and to send call log data to the system management platform (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**).	**System management platform:** Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-

Czajkowski in view of Inbar and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Osterhout teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Osterhout states: "If the criteria are met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephone device itself may be both POTS and SIP enabled, in which case the base of the device contains both telephone (RJ-11 or other) connections plus a network connection or port for SIP, with control logic residing in the telephone device and no computer or other host being necessary. In a yet further embodiment, the telephone device may contain control logic and connections for each of POTS, USB and SIP for maximum connectivity." Improved System—A person having ordinary skill in the art in 2001 would have considered it

Czajkowski in view of Inbar and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Czajkowski to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Osterhout teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Osterhout states: "If the criteria are met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call

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Czajkowski in view of Inbar and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephone device itself may be both POTS and SIP enabled, in which case the base of the device contains both telephone (RJ-11 or other) connections plus a network connection or port for SIP, with control logic residing in the telephone device and no computer or other host being necessary. In a yet further embodiment, the telephone device may contain control logic and connections for each of POTS, USB and SIP for maximum connectivity." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Czajkowski to participate in SIP-based telephony systems.

APPENDIX F6

Czajkowski in view of Inbar and further in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		function which provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (**processor of CPE 2 or processor in PC 3**);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**application 61**),	**call processing application:** Application 61 performs a list of call-processing services. (FIGURE 2; 8/60).	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet. and to send call log data to the system management platform (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**).	**System management platform:** Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2).

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'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	**Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack."

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Czajkowski in view of Inbar and further in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the

Czajkowski in view of Inbar and further in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.

Czajkowski in view of Inbar and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (processor of CPE 2 or processor in PC 3);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**application 61**),	**call processing application:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60)	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet and to send call log data to the system management platform (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**).	**System management platform:** Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-

APPENDIX F7

Czajkowski in view of Inbar and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it

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Czajkowski in view of Inbar and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP

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Czajkowski in view of Inbar and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.

APPENDIX F10

Czajkowski in view of Kung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
Claim 13	A method for establishing a voice-over-packet network architecture, the method comprising: locating a system management platform in a shared packet network (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**), the system management platform collecting call log data from a plurality of network devices ; and	"In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29. "The flow diagram for the specific example 3 shown in FIG. 5 is just one possible service, in which the PC application makes a log of the number being called. This example is kept simple to illustrate the principle, and it will be appreciated that additional messages could be sent by the control processor 25 to allow the PC application to determine for example the duration of the call, etc." Col. 11, ll. 30-36.	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. Kung, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Kung teaches an IP central station 200 that stores a call log: "The present invention may include an activity log that may have user proactive bill management capability and be used in the aforementioned broadband communication system. The activity log may log, for example, incoming calls directory numbers (DNs) and outgoing call DNs in a database. The database containing the activity log may be provided at a central system location, such as the at IP Central Station 200." Col. 31, ll. 10-17. Figure 8 of Kung includes an example call log. The call log is stored at BRG 300 and/or IP central station 200. Col. 32, ll.9-10. The system subscriber's customer premises equipment (broadband residential gateway 300) records the call log data and forwards the call log data to other locations, such as to IP central station 200, for billing purposes as an example. Figure 8;

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Czajkowski in view of Kung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			Col. 35, l. 37 - col. 36, l. 10. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include a system management platform, for example, to collect call log data from the network devices of Czajkowski.
	distributing the plurality of network devices (**CPE 2 which may include the computer functionality of PC 3**) that each include a telephone line interface (**borscht circuit 52**), a computer data interface (**ethernet interface 23**), a broadband network interface terminating a link from the shared packet network (**xDSL modem 21**)	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1: Col. 5, ll. 23-34. Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well as a number of POTS lines 4a and 4b for example, each with a	

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Czajkowski in view of Kung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1. BORSCHT circuit 25 is a telephone line interface. "The BORSCHT circuit (Battery, Over-voltage protection, Ringing, Supervision, Coding, Hybrid, Test) is a known function which provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (**processor of CPE 2 or processor in PC 3**);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**application 61**),	**call processing application:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60).	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet. and to send call log data to the system management platform (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**).	**System management platform:** Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	**Base System**— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**—A person having ordinary skill in the art in 2001 would have been well-aware of system management platform technology. Kung, for example, describes a system management platform collecting call log data from a plurality of network devices. For example, Kung teaches an IP central station 200 that stores a call log: "The present invention may include an activity log that may have user proactive bill management capability and be used in the aforementioned broadband communication

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Czajkowski in view of Kung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			system. The activity log may log, for example, incoming calls directory numbers (DNs) and outgoing call DNs in a database. The database containing the activity log may be provided at a central system location, such as the at IP Central Station 200." Col. 31, ll. 10-17. Figure 8 of Kung includes an example call log. The call log is stored at BRG 300 and/or IP central station 200. Col. 32, ll.9-10. The system subscriber's customer premises equipment (broadband residential gateway 300) records the call log data and forwards the call log data to other locations, such as to IP central station 200, for billing purposes as an example. Figure 8; Col. 35, l. 37 - col. 36, l. 10. Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include a system management platform, for example, to collect call log data from the network devices of Czajkowski.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"Digital Subscriber Line (DSL) services are broadband services in the sense that digital information is sent over a high-bandwidth channel above the baseband voice channel on a single pair of wires. For example, "the subscriber's CPE 2 provides high speed data access to a PC for example as well	

Czajkowski in view of Kung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		as a number of POTS lines 4a and 4b for example, each with a different number." Figure 2; Col. 6, l. 66 – Col. 7, l. 1."	
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	To support broadband communication over a legacy network, xDSL technology is used in a Customer Premises Equipment (CPE) module which is connected to a PC and telephones. An XDSL modem 21 is provided as the broadband network interface. "FIG. 1 shows a typical arrangement in which xDSL is used in a legacy access network 5 to provide multiple voice and data services to network subscribers 1. The legacy access network 5 typically comprises twisted copper pair cables, one pair running from each of a plurality of subscribers 1 to the exchange (central office) or cabinet 6. By installing suitable xDSL Line Terminating Equipment (LTE) 7 at the exchange or cabinet 6 and xDSL Customer Premises Equipment (CPE) 2 at subscriber premises 1, subscribers 1 are able to obtain multiple voice 4 and/or data 3 services over the single twisted copper pair cable or line 5 originally dedicated to them by the network operator." Figure 1; Col. 5, ll. 23-34.	
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	
Claim 19	The method of claim 13, wherein the plurality of	"Telephone access networks have historically always been	

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Czajkowski in view of Kung

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
	network devices each further include a video streaming device interface.	connection orientated, typically using dedicated copper wire circuits between subscribers and the exchange or central office. Typically these access networks were designed to carry only voice with a bandwidth of less than 4 kHz. However in recent years with the growth of the internet and demand for other multi-media services such as video-on-demand and video conferencing, subscribers of telephone companies have demanded additional bandwidth over the access network to provide adequate Internet and multi-media services access." Col. 1, ll. 17-27.	

APPENDIX F5

Czajkowski in view of Kung and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (processor of CPE 2 or processor in PC 3);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (application 61),	**call processing application:** Application 61 performs a list of call processing services. (FIGURE 2; 8/60).	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet, and to send call log data to the system management platform (control processor **25 sends messages across the API 55 to an application 61 on the PC 3).**	**System management platform:** Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the	**Base System—** Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique—** A person having ordinary skill in the art in 2001 would have been well-

Czajkowski in view of Kung and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Osterhout teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Osterhout states: "If the criteria are met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephone device itself may be both POTS and SIP enabled, in which case the base of the device contains both telephone (RJ-11 or other) connections plus a network connection or port for SIP, with control logic residing in the telephone device and no computer or other host being necessary. In a yet further embodiment, the telephone device may contain control logic and connections for each of POTS, USB and SIP for maximum connectivity." Improved System—A person having ordinary skill in the art in 2001 would have considered it

Czajkowski in view of Kung and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Czajkowski to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	Base System— Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). Known Technique— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Osterhout teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Osterhout states: "If the criteria are met, the control module 126 may set up the remainder of the resources necessary to establish a SIP-based connection to a recipient telephone device 120. The control module may invoke SIP module 122 and SIP stack 124 to transmit, receive parse SIP commands, a Transfer Control Protocol/Internet Protocol (TCP/IP) client 130 for Internet or other network interface, and a Real Time Protocol (RTP) stack 134 to manage streaming media and other information for call

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Czajkowski in view of Kung and further in view of Osterhout

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			processing." Col. 4, l. 65 - Col. 5 - l. 6. Osterhout explains the need for transparently selecting SIP-based, POTS, or other telephone service: "In a further embodiment, the telephone device itself may be both POTS and SIP enabled, in which case the base of the device contains both telephone (RJ-11 or other) connections plus a network connection or port for SIP, with control logic residing in the telephone device and no computer or other host being necessary. In a yet further embodiment, the telephone device may contain control logic and connections for each of POTS, USB and SIP for maximum connectivity." **Improved System**—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Czajkowski to participate in SIP-based telephony systems.

APPENDIX F6

Czajkowski in view of Kung and further in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		function which provides a physical interface to a standard analog telephone handset." Figure 2; Col. 6, ll. 14-17. Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (processor of CPE 2 or processor in PC 3);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**application 61**),	**call processing application**: Application 61 performs a list of call-processing services. (FIGURE 2; 8/60).	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet. and to send call log data to the system management platform (**control processor 25 sends messages across the API 55 to an application 61 on the PC 3**).	**System management platform**: Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2).

Czajkowski in view of Kung and further in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	**Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack."

Office Action Appendix
Control Number 95/000,344 4-2-2009 Part of Paper No. 20090402
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Czajkowski in view of Kung and further in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Wengrovitz teaches a SIP user agent to represent a non-SIP telephone and a SIP proxy server that mediates all SIP communications. For example, Wengrovitz states: "Switching device 50 is preferably a private branch exchange (PBX) unit managing incoming and outgoing calls for a particular location. Switching device 50 includes an emulation client 50a for converting incoming SIP messages into PBX messages and outgoing PBX messages into SIP messages. In its simplest form, the emulation client 50a takes the

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Czajkowski in view of Kung and further in view of Wengrovitz

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			role of a UAC 15 in the data communication network. According to one embodiment of the invention, the emulation client 50a is implemented as a software program executing on the internal PBX processor." Wengrovitz, Col. 4, ll. 11-21. Wengrovitz explains the need for enabling legacy telephones to make and receive SIP calls: "there is a need in the current art for a system and method for enabling legacy telephones to participate seamlessly in SIP-based telephony systems. Such a system and method should allow legacy telephones to seamlessly make and receive SIP calls with other legacy telephones as well as with telephones with SIP functionality without requiring that such legacy telephones be equipped with their own SIP stack." Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.

APPENDIX F7

Czajkowski in view of Kung and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		Ethernet interface 23 is a computer data interface. "The ethernet interface 23 provides for a connection to a subscriber PC 3 and/or other subscriber data appliances such as further PCs or a set-top box." Figure 2; Col. 6, ll. 1-3.	
	a processor (processor of CPE 2 or processor in PC 3);	PC 3 is a personal computer running applications and inherently includes a processor (FIGURE 2; 7/28)	
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (application 61),	call processing application: Application 61 performs a list of call processing services. (FIGURE 2; 8/60)	
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet and to send call log data to the system management platform (control processor 25 sends messages across the API 55 to an application 61 on the PC 3).	System management platform: Control processor 25 sends messages across the API 55 to an application 61 on the PC 3. Although not depicted, PC 3 may serve more than one CPE 2. "In these cases, the control processor 25 sends messages across the API 55 to an application 61 on the PC 3 listening on the appropriate port. Where information about the called number is required, a copy of the outgoing voice traffic path is routed through the multicast ATM AAL2 switch 22 to the PC such that it can decode the DTMF tones being sent to the PSTN network. Preferably, DTMF decoding is implemented in the PC 3 using software running on the PC." Col. 11, ll. 22-29.	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the	Base System—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). Known Technique— A person having ordinary skill in the art in 2001 would have been well-

Czajkowski in view of Kung and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
		ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. **Improved System**—A person having ordinary skill in the art in 2001 would have considered it

Czajkowski in view of Kung and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	Control processor 25 creates ATM virtual circuits for transferring packet switched cells across the network. Control processor 25 mediates communications to the external network by controlling ATM switch 22. "Control processor 25 allocates a virtual circuit between the ATM-POTS interface 24 and the Gateway 9 to the PSTN, the DTMF tones being loaded into ATM cells for transport across the access network 5 via the xDSL modems 21 in the same way as speech samples later in the call." Figure 2; Col. 6, ll. 35-39.	**Base System**—Czajkowski discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CPE 2). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an Internet Engineering Task Force (IETF) standard in 1999 as RFC 2543. Girard teaches a SIP user agent (SIP User Agent in the softswitch utilizing the SIP signaling pathway established for initial call setup) to represent a non-SIP telephone and a SIP proxy server that is a concatenation of a UAC and UAS. For example, Girard states that the Application Server includes a SIP Proxy Server as defined by prosecuting counsel for the '519 Patent: "Using a SIP User Agent Client (UAC), telephony applications running on the APPLICATION SERVER may create connections between any two network endpoints in any connectivity domain (IP, ATM, PSTN, etc.) that is addressable by the SOFTSWITCH and the MGs under its control. The APPLICATION SERVER also contains a SIP

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Czajkowski in view of Kung and further in view of Girard-SIP

'519 Claim	Claim Limitations	Czajkowski U.S. Patent 6,526,058	Secondary References
			User Agent Server (UAS)." Girard, Figure 2; Pages 1 and 6. Girard explains the need for enabling legacy telephones to make and receive SIP calls: "this SIP-Telephony Service Interface (SIP-TSI) is capable of supporting a level of telephony application functionality commensurate with Time Division Multiplex device interfaces used in legacy PSTN voice and facsimile telephony applications." Girard, Page 1. Improved System—A person having ordinary skill in the art in 2001 would have considered it obvious to modify the base system of Czajkowski to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Czajkowski* to participate in SIP-based telephony systems.

APPENDIX G1

Gerszberg

'519 Claim	Claim Limitations	Gerszberg U.S. Patent 6,510,152
Claim 1	A network device (ISD/IRG 22) comprising: a plurality of communication interfaces, including a telephone line interface (residential interface 115, cordless phone interface 123, conventional analog lines 505), a computer data interface (audio/video interface 120, Ethernet connections 501), and a broadband network interface (cable/xDSL modem 114);	Gerszberg teaches a residence gateway which may be an integrated residence gateway (IRG) disposed near the customer's premises for multiplexing and coordinating many digital services onto a single twisted-pair line or coaxial cable (or both). "[A]n intelligent services director (ISD) 22 may be coupled to a telephone central office 34 via a twisted-pair wire, hybrid fiber interconnection, wireless and/or other customer connection 30, a connector block 26, and/or a main distribution frame (MDF) 28. Referring briefly to FIG. 1B, and according to the present invention, the ISD 22 is replaced by either a residential gateway 22-2 (when an interexchange carrier partners with a cable television service provider) or an integrated residential gateway 22-1 (when an interexchange carrier is integrated with the cable television service provider." Figure 1; Col. 4, ll. 22-32. The IRG connects with analog telephones 15. "The IRG 22 may connect with a variety of devices including analog and digital voice telephones 15." Figure 1E; Figure 2; Figure 5, Col. 7, ll. 33-35. The IRG connects with personal computers 14. "personal computers utilizing cable modem bandwidth Internet services are typically coupled to IRG 22 to coaxial cable lines run within the home. Alternatively, services are provided via an Ethernet interface 119 or other high bandwidth interface." Figure 1E; Figure 2; Figure 5, Col. 7, ll. 49-53. The IRG aggregates the traffic from the attached devices onto a single broadband interface 114. "In exemplary embodiments, the ISD/IRG 22 multiplexes traffic from the various components of the PDN 500 (e.g., Ethernet, Screen Phone, Tip/Ring, ISDN, coaxial house cable) either between other devices on the PDN and/or onto DSL/cable modem 114 for transport over loop twisted pair to the Central Office or coax toward the cable television headend." Figure 2; Figure 5; Col. 14, ll. 52-58.
	a processor; (central processing unit 102)	"As shown in FIG. 2, in some embodiments the IRG 22 may include a controller 100 which may have any of a variety of elements such as a central processing unit 102." Figure 2; Col. 7, ll. 64-66.
	a machine-readable storage medium (memory of ISD/IRG 22) which during use	"As shown in FIG. 2, in some embodiments the IRG 22 may include a controller