Chow in view of Wengrovitz

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	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**

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'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			skill in the art in 2001 would have considered it obvious to modify the base system of Chow to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Chow* 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 (CSM 6), the system management platform collecting call log data from a plurality of network devices ; and	Service Provider Network Access Interface (SPNAI) 27 aggregates the traffic from the attached devices onto a single broadband interface. "SPNAI 27 provides connectivity to UDS pipe 1. CSM 6 is able to suport the various transport technologies implemented for the UDS pipe. CSM 6 converts all information (i.e., voice, data, multimedia and video) into packet (e.g., IP over ATM or voice over IP) based medium for transport to/from the NAM." (FIGURE 2; [0050]). "CSM keeps a log of all incoming and outgoing requests;" ([0088])	
	distributing the plurality of network devices (CSM 6) that each include a telephone line interface (SSI 21 includes a T/R interface), a computer data interface (SSI 21 includes a parallel port interface and an Ethernet interface),	Chow teaches a network architecture that integrates broadband subscriber services from a service provider (FIGURE 1; [0038]). Subscriber Site Interface (SSI) 21 connects with T/R telephones. "SSI 21 provides local access to the CPEs that may consist of existing residential T/R phone, ISDN/BRI phone, computer modem, fax machine, wireless residential base station (e.g., extension of public cellular service to	

Chow in view of Wengrovitz

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
Claim 9	A network device (CSM 6) comprising: a broadband network interface (**SPNAI 27**); a plurality of communication interfaces, including a telephone line interface (**SSI 21 includes a T/R interface**) and a computer data interface (**SSI 21 includes a parallel port interface and an Ethernet interface**), and;	Chow teaches a network architecture that integrates broadband subscriber services from a service provider (FIGURE 1; [0038]). Subscriber Site Interface (SSI) 21 connects with T/R telephones. "SSI 21 provides local access to the CPEs that may consist of existing residential T/R phone, ISDN/BRI phone, computer modem, fax machine, wireless residential base station (e.g., extension of public cellular service to home, PCS) and LAN, etc.." (FIGURE 2; [0049]). Subscriber Site Interface (SSI) 21 connects with computers. "SSI 21 provides local access to the CPEs that may consist of existing residential T/R phone, ISDN/BRI phone, computer modem, fax machine, wireless residential base station (e.g., extension of public cellular service to home, PCS) and LAN, etc.." (FIGURE 2; [0049]). Service Provider Network Access Interface (SPNAI) 27 aggregates the traffic from the attached devices onto a single broadband interface. "SPNAI 27 provides connectivity to UDS pipe 1. CSM 6 is able to suport the various transport technologies implemented for the UDS pipe. CSM 6 converts all information (i.e., voice, data, multimedia and video) into packet (e.g., IP over ATM or voice over IP) based medium for transport to/from the NAM." (FIGURE 2; [0050]).	
	a processor (**CPU 23**);	"As shown in FIG. 2, the CSM includes a number of inter-related elements, including an IP router 20, a speech processor 22, a CPU 23, memory 24 and a call processing inter-working unit 25." (FIGURE 2; [0047])	
	a machine-readable storage medium **memory 24 of CSM 6)** that stores processor-a machine-	"As shown in FIG. 2, the CSM includes a number of inter-related elements, including an IP router 20, a speech	

Chow in view of Wengrovitz

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
	readable storage medium that stores processor-executable instructions to provide SIP agents (**call processing inter-working unit 25 or the other software at CSM 6 that performs services**)	processor 22, a CPU 23, memory 24 and a call processsing inter-working unit 25." (FIGURE 2; [0047])	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface (**call processing inter-working unit 25 or the other software at CSM 6 that performs services**), and the instructions further causing the network device to implement a SIP proxy server (**CSM 6 includes instructions to mediate communications between the plurality of communication interfaces**) that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	**SIP user agents and SIP proxy:** CSM 6 includes software, such as call processing inter-working unit 25, that performs call processing. (FIGURE 2; [0047]).	**Base System**—Chow discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CSM 6). **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:

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			"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 Chow to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Chow* to participate in SIP-based telephony systems.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	"The integrated system of the present invention can be implemented with a broadband packet access network as the supporting infrastructure that enables the UDS pipe access to the service provider's core backbone network. Such a network is capable of supporting the traditional circuit-switched connection, IP-based connection less packets and mobile IP for personal mobility. It is envisioned that the intelligence of the network will be distributed to the home environment for the subscriber to control how service is rendered. The methodology and the service applications necessary to cost effectively integrate a UDS pipe for local access services with	

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Control Number 95/000,344 4-2-2009 Part of Paper No. 20090402
Chow in view of Wengrovitz APPENDIX J3

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
		integrated voice, data and multimedia applications from the home is an important objective of the present invention." ([0029])	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	Video phone 3. Figure 1.	
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	CSM 6 is contained in a single physical enclosure. Figure 1.	

Chow in view of Wengrovitz

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
	~~outgoing requests).~~	~~"CSM keeps a log of all incoming and outgoing requests;"~~ ([0088])	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	"The integrated system of the present invention can be implemented with a broadband packet access network as the supporting infrastructure that enables the UDS pipe access to the service provider's core backbone network. Such a network is capable of supporting the traditional circuit-switched connection, IP-based connection less packets and mobile IP for personal mobility. It is envisioned that the intelligence of the network will be distributed to the home environment for the subscriber to control how service is rendered. The methodology and the service applications necessary to cost effectively integrate a UDS pipe for local access services with integrated voice, data and multimedia applications from the home is an important objective of the present invention." ([0029])	**Base System**—Chow discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CSM 6). **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:

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'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			"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 Chow to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Chow* 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:** CSM 6 detects network signaling events. "Ted is using his video-phone and tells his CSM to call Paul for video-telephony. His CSM finds Paul's CSMN, 555-2222, in the CSM's address book stored, for example, in a database within memory 24 shown in FIG. 2 and sends a signaling request to the network to Paul's CSM. When Paul's CSM detects the incoming request, it will parse the signaling to determine who is the request for, the type of request, and other service data." ([0156-0159]).	**Base System**—Chow discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CSM 6). **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

Chow in view of Wengrovitz

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			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 skill in the art in 2001 would have considered it obvious to modify the base system of Chow to include these well-known claimed SIP components and apply the well-known techniques

Chow in view of Wengrovitz

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			taught by Wengrovitz, for example, to enable the telephones in *Chow* to participate in SIP-based telephony systems.

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
		finds Paul's CSMN, 555-2222, in the CSM's address book stored, for example, in a database within memory 24 shown in FIG. 2 and sends a signaling request to the network to Paul's CSM. When Paul's CSM detects the incoming request, it will parse the signaling to determine who is the request for, the type of request, and other service data. The CSM will determine that Paul is available to receive such a request, (i.e., Paul has not informed the CSM to redirect his calls) and will acknowledge the request, allocate the bandwidth from the UDS pipe (e.g., 384 Kbps), and direct the call to Paul's video-telephone." ([0156-0159]).	
	the call processing application operating according to parameters defined in the service profiles	**service profiles:** CSM 6 stores service profiles. "5. Provide residential subscriber database management (e.g., profile etc.);" (FIGURE 2; [0076]).	
	wherein the network device consists of one or more customer premise equipment modules (CSM 6).	"In accordance with the present invention, all of the existing Customer Premises Equipment ("CPE", i.e, equipment not provided to the customer by the service provider) is coupled to a Customer Service Manager (CSM) 6. As shown in FIG. 1, the CPEs may include a remote laptop computer interface 2, video phone 3, computer 4 and telephone 5. As in the typical home environment, the various CPEs may be located in different rooms within the home and are connected to CSM 6 by direct signal wire connection or other suitable means." ([0039])	
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:** CSM 6 includes software, such as call processing inter-working unit 25, that performs call processing. (FIGURE 2; [0047]).	**Base System**—Chow discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CSM 6). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			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 obvious to modify the base system of Chow to

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Chow* 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 (CSM 6), the system management platform collecting call log data from a plurality of network devices ; and	Service Provider Network Access Interface (SPNAI) 27 aggregates the traffic from the attached devices onto a single broadband interface. "SPNAI 27 provides connectivity to UDS pipe 1. CSM 6 is able to suport the various transport technologies implemented for the UDS pipe. CSM 6 converts all information (i.e., voice, data, multimedia and video) into packet (e.g., IP over ATM or voice over IP) based medium for transport to/from the NAM." (FIGURE 2; [0050]). "CSM keeps a log of all incoming and outgoing requests;" ([0088])	
	distributing the plurality of network devices (CSM 6) that each include a telephone line interface (**SSI 21 includes a T/R interface**), a computer data interface (**SSI 21 includes a parallel port interface and an Ethernet interface**), a broadband network interface terminating a link	Chow teaches a network architecture that integrates broadband subscriber services from a service provider (FIGURE 1; [0038]). Subscriber Site Interface (SSI) 21 connects with T/R telephones. "SSI 21 provides local access to the CPEs that may consist of existing residential T/R phone, ISDN/BRI phone, computer modem, fax machine, wireless residential base station (e.g., extension of public cellular service to home, PCS) and LAN, etc.." (FIGURE 2; [0049]).	

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
Claim 9	A network device (CSM 6) comprising: a broadband network interface (**SPNAI 27**); a plurality of communication interfaces, including a telephone line interface (**SSI 21 includes a T/R interface**) and a computer data interface (**SSI 21 includes a parallel port interface and an Ethernet interface**), and;	Chow teaches a network architecture that integrates broadband subscriber services from a service provider (FIGURE 1; [0038]). Subscriber Site Interface (SSI) 21 connects with T/R telephones. "SSI 21 provides local access to the CPEs that may consist of existing residential T/R phone, ISDN/BRI phone, computer modem, fax machine, wireless residential base station (e.g., extension of public cellular service to home, PCS) and LAN, etc.." (FIGURE 2; [0049]). Subscriber Site Interface (SSI) 21 connects with computers. "SSI 21 provides local access to the CPEs that may consist of existing residential T/R phone, ISDN/BRI phone, computer modem, fax machine, wireless residential base station (e.g., extension of public cellular service to home, PCS) and LAN, etc.." (FIGURE 2; [0049]). Service Provider Network Access Interface (SPNAI) 27 aggregates the traffic from the attached devices onto a single broadband interface. "SPNAI 27 provides connectivity to UDS pipe 1. CSM 6 is able to suport the various transport technologies implemented for the UDS pipe. CSM 6 converts all information (i.e., voice, data, multimedia and video) into packet (e.g., IP over ATM or voice over IP) based medium for transport to/from the NAM." (FIGURE 2; [0050]).	
	a processor (CPU 23);	"As shown in FIG. 2, the CSM includes a number of inter-related elements, including an IP router 20, a speech processor 22, a CPU 23, memory 24 and a call processsing inter-working unit 25." (FIGURE 2; [0047])	
	a machine-readable storage medium memory 24 of CSM 6) that stores processor-a machine-	"As shown in FIG. 2, the CSM includes a number of inter-related elements, including an IP router 20, a speech	

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
	readable storage medium that stores processor-executable instructions to provide SIP agents **(call processing inter-working unit 25 or the other software at CSM 6 that performs services)**	processor 22, a CPU 23, memory 24 and a call processsing inter-working unit 25." (FIGURE 2; [0047])	
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface **(call processing inter-working unit 25 or the other software at CSM 6 that performs services)**, and the instructions further causing the network device to implement a SIP proxy server **(CSM 6 includes instructions to mediate communications between the plurality of communication interfaces)** that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	**SIP user agents and SIP proxy:** CSM 6 includes software, such as call processing inter-working unit 25, that performs call processing. (FIGURE 2; [0047]).	**Base System**—Chow discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CSM 6). **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

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			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 Chow to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Chow* to participate in SIP-based telephony systems.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	"The integrated system of the present invention can be implemented with a broadband packet access network as the supporting infrastructure that enables the UDS pipe access to the service provider's core backbone network. Such a network is capable of supporting the traditional circuit-switched connection, IP-based connection less packets and mobile IP for personal mobility. It is envisioned that the intelligence of the network will be distributed to the home environment for the subscriber to control how service is rendered. The methodology and the service applications necessary to cost effectively integrate a UDS pipe for local access services with integrated voice, data and multimedia applications from the home is an important objective of the present invention."	

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
		([0029])	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	Video phone 3. Figure 1.	
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	CSM 6 is contained in a single physical enclosure. Figure 1.	

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
		([0088])	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	"The integrated system of the present invention can be implemented with a broadband packet access network as the supporting infrastructure that enables the UDS pipe access to the service provider's core backbone network. Such a network is capable of supporting the traditional circuit-switched connection, IP-based connection less packets and mobile IP for personal mobility. It is envisioned that the intelligence of the network will be distributed to the home environment for the subscriber to control how service is rendered. The methodology and the service applications necessary to cost effectively integrate a UDS pipe for local access services with integrated voice, data and multimedia applications from the home is an important objective of the present invention." ([0029])	**Base System**—Chow discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CSM 6). **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: "this SIP-Telephony Service Interface (SIP-TSI) is

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			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 Chow to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Chow* 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:** CSM 6 detects network signaling events. "Ted is using his video-phone and tells his CSM to call Paul for video-telephony. His CSM finds Paul's CSMN, 555-2222, in the CSM's address book stored, for example, in a database within memory 24 shown in FIG. 2 and sends a signaling request to the network to Paul's CSM. When Paul's CSM detects the incoming request, it will parse the signaling to determine who is the request for, the type of request, and other service data." ([0156-0159]).	**Base System**—Chow discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, CSM 6). **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

Chow in view of Girard-SIP

'519 Claim	Claim Limitations	Chow U.S. Publication No. 2003/0185203	Secondary References
			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 Chow to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Chow* to participate in SIP-based telephony systems.

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108
Claim 1	A network device (MAC 100) comprising: a plurality of communication interfaces, including a telephone line interface (analog voice ports), a computer data interface (serial port 104), and a broadband network interface (Ethernet port 102);	The MAC uses a software-configurable wide area network (WAN) trunk to seamlessly integrate data, voice, and video into existing networks using common switch and network hardware ... (Col. 6, ll. 23-27) The interface pots of the MAC of an embodiment comprise a single Ethernet port 102, two serial ports 104-106 that support speeds up to 2 Mbps, and either six analog voice ports or a single digital voice access port (T1/E1) ... (Col. 6, ll. 45-49)
	a processor (CPU 199);	With reference to FIG. 1, the interface ports of the MAC 100 are coupled to a central processing unit (CPU) 199... (Col. 7, ll. 46-47)
	a machine-readable storage medium (memory devices 110-114) which during use stores	"Furthermore, multiple memory devices 110-114 are coupled to the CPU 199 of one embodiment, wherein dynamic random access memory (DRAM) 110 is supported in 4, 8, 16, and 32 Mb single inline memory modules (SIMMs) and flash memory 112-114, or nonvolatile memory, is supported in 4, 8, and 16 Mb memory devices, but the embodiment is not so limited. A 32-bit SIMM socket supports up to 64-Mbyte of program memory and data storage memory 110. A 32-bit SIMM socket supports up to 32-Mbyte of flash memory 112. Furthermore, a 512-Kbyte boot flash 114 is provided, but the embodiment is not so limited. Moreover, a 2-Mbyte on-board flash memory 112 supports system configuration." (Col. 7, ll. 53-65)
	a call processing application (processing subsystem 200) and	FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28).
	service profiles (MAC stores service configurations),	The MAC, in providing voice services, supports a wide range of call management configurations comprising configurations for dial on-net extensions, off-net numbers, and local calls, but the embodiment is not so limited. The MAC supports call set-up options comprising, but not limited to: on-net dialing; auto-dial, or private line automatic ringdown (PLAR); off-net dialing; and tandem switching. (Col. 15, l. 63-Col. 16, l. 3).
	and which stores executable instructions to mediate communications between the	FIG. 10 shows the data and voice flows when using a serial port as a network

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	plurality of communication interfaces (routing engine and I/O handler 1004),	interface in an embodiment of the MAC 1000 of the present invention. Data is received into a routing engine 1002 and passed to an I/O handler 1004. Voice channels are received from a private branch exchange, keyset, or telephone 1006. The received voice channels are routed to a voice call handler 1008, a voice compression engine 10100, and to an I/O handler 1004. The I/O handler 1004 couples the data and voice to serial port 0 as the network interface. (Col. 14, ll. 23-32) The MAC of an embodiment supports on-net dialing when a call originator dials an extension by entering a phone number; the call is connected within the wide area network. For on-net calls, a flexible call numbering plan allows dialing to any port on any system in the network by dialing a unique prefix that identifies the port or group of ports on the destination system. (Col. 16, ll. 4-10). The MAC of an embodiment supports off-net dialing, wherein when a caller dials "9", or another pre-programmed digit/digits, the MAC automatically connects the caller extension to a channel connected directly to a PSTN. (Col. 16, ll. 23-26).
	the instructions causing the network device to detect network signaling events or trigger points in a telephone call (call setup message) and	The MAC supports analog voice streams using E&M, FXS, and FXO voice signal standards. (Col. 7, ll. 2-7) In operation, an analog voice port or digital voice port provides a voice signal to the Voice Signal Process module 202. The Voice Signal Process module 202 translates the voice signal to a call setup message. (Col. 7, ll. 29-32).
	invoke the call processing application in response to the detected network signaling events or trigger points (call setup procedure),	The call setup procedure comprises calling the Voice Channel Manager 206 to allocate a DSP for connection to the receiving voice port via the PCM bus. Further, the call setup procedure signals the Voice Signal Process module 202 to enable the allocated DSP to start the DTMF sampling for the dialing digits. Upon collection of enough digits by consulting the dial-mapper 208, a setup message is provided to the Tandem Switch Module 210. The Tandem Switch Module 210, using the Dial-peer and Dial-mapper 208, locates the permanent victual connection (PVC) number of the remote extension in order to provide a setup message. (Col. 7, ll. 32-45)

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	the call processing application operating according to parameters defined in the service profiles	The MAC, in providing voice services, supports a wide range of call management configurations comprising configurations for dial on-net extensions, off-net numbers, and local calls, but the embodiment is not so limited. The MAC supports call set-up options comprising, but not limited to: on-net dialing; auto-dial, or private line automatic ringdown (PLAR); off-net dialing; and tandem switching. (Col. 15, l. 63-Col. 16, l. 3).
	wherein the network device consists of one or more customer premise equipment modules **(MAC 100 connects to a PBX 198).**	"The MAC of an embodiment comprises two PIMs that are used for network access, the DVM and AVM. The DVM, functionally equivalent to the MTM, provides connectivity to a digital private branch exchange 198 or channel bank. The voice channels can be either mapped to time slots on the primary PCM bus for voice compression or mapped to time slots on another PCM bus through a cross-connect switch." (Col. 10, ll. 1-7)
Claim 2	The network device of claim 1, wherein the plurality of communication interfaces further includes a video streaming device interface **(WAN trunk).**	"Furthermore, the UIO port receives video traffic. Following circuit emulation, the video traffic is transported using the WAN trunk. The MAC supports analog voice streams using Ear and Mouth (E&M) (2 wire and 4 wire with immediate dial, delay dial and wink start), Foreign Exchange Station (FXS) (ground start and loop start), and Foreign Exchange Office (FXO) (ground start and loop start) voice signal standards." (Col. 6, l. 67-Col. 7, ll. 1-7)
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.	The interface pots of the MAC of an embodiment comprise a single Ethernet port 102, two serial ports 104-106 that support speeds up to 2 Mbps, and either six analog voice ports or a single digital voice access port (T1/E1) ... (Col. 6, ll. 45-49)
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.	The interface pots of the MAC of an embodiment comprise a single Ethernet port 102, two serial ports 104-106 that support speeds up to 2 Mbps, and either six analog voice ports or a single digital voice access port (T1/E1) ... (Col. 6, ll. 45-49)
Claim 5	The network device of claim 1, wherein the network device is contained in a	MAC 100 is contained in a single physical enclosure. Figure 1.

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	single physical enclosure.	
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 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)
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 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)
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 **(SNMP-based MIB)**, the system management platform collecting call log data from a plurality of network devices ; and	Management and configuration of the MAC of one embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13). "Furthermore, the routing table includes information elements comprising a preference-order and a last-use timestamp. The last-use timestamp comprises a last update to entry timestamp and an entry creation timestamp, but the embodiment is not so limited. The preference-order defines an explicit selection priority that should be give to each table entry. The last use timestamp tracks the last time a call was routed to the entry. The last use timestamp, in tracking successfully completed calls and call attempts, is an important attribute in that it allows load balancing." (Col. 26, ll. 17-26)
	distributing the plurality of network devices **(MAC 100)** that each include a telephone line interface **(analog voice ports)**,	The MAC uses a software-configurable wide area network (WAN) trunk to seamlessly integrate data, voice, and video into existing networks using common switch and network hardware ... (Col. 6, ll. 23-27)

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	a computer data interface (serial port 104), a broadband network interface terminating a link from the shared packet network (Ethernet port 102)	The interface pots of the MAC of an embodiment comprise a single Ethernet port 102, two serial ports 104-106 that support speeds up to 2 Mbps, and either six analog voice ports or a single digital voice access port (T1/E1) ... (Col. 6, ll. 45-49)
	a processor (CPU 199);	With reference to FIG. 1, the interface ports of the MAC 100 are coupled to a central processing unit (CPU) 199... (Col. 7, ll. 46-47)
	a machine-readable storage medium (memory devices 110-114) storing processor-executable instructions to control telephone calls (processing subsystem 200),	FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28).
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion over the shared packet network (network comprising multiple MACs of an embodiment of the present invention), and to send call log data to the system management platform (SNMP-based MIB)).	"FIG. 38 is a network comprising multiple MACs of an embodiment of the present invention." (Col. 5, ll. 48-49) Furthermore, the routing table includes information elements comprising a preference-order and a last-use timestamp... The last-use timestamp, in tracking successfully completed calls and call attempts, is an important attribute in that it allows load balancing. Moreover, network peers may be associated with various administrative metrics that reflect various attributes of the network interface... (Col. 26, ll. 24-29). Management and configuration of the MAC of one embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13).

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	The interface pots of the MAC of an embodiment comprise a single Ethernet port 102, two serial ports 104-106 that support speeds up to 2 Mbps, and either six analog voice ports or a single digital voice access port (T1/E1) ... (Col. 6, ll. 45-49)
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	"It is therefore an object of the invention to integrate data, voice, and video onto public and private packet-based or cell-based multiservice networks comprising Frame Relay, Asynchronous Transfer Mode (ATM), High-level Data Link Control (HDLC), Internet Protocol (IP), and Time Division Multiplexed (TDM) networks, and leased line carrier services." (Col. 3, ll. 10-16)
Claim 18	The method of claim 13, wherein the shared packet network uses ATM protocols.	"It is therefore an object of the invention to integrate data, voice, and video onto public and private packet-based or cell-based multiservice networks comprising Frame Relay, Asynchronous Transfer Mode (ATM), High-level Data Link Control (HDLC), Internet Protocol (IP), and Time Division Multiplexed (TDM) networks, and leased line carrier services." (Col. 3, ll. 10-16)
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface (WAN trunk).	"Furthermore, the UIO port receives video traffic. Following circuit emulation, the video traffic is transported using the WAN trunk. The MAC supports analog voice streams using Ear and Mouth (E&M) (2 wire and 4 wire with immediate dial, delay dial and wink start), Foreign Exchange Station (FXS) (ground start and loop start), and Foreign Exchange Office (FXO) (ground start and loop start) voice signal standards." (Col. 6, l. 67-Col. 7, ll. 1-7)

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'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
		202. The Voice Signal Process module 202 translates the voice signal to a call setup message. (Col. 7, ll. 29-32).	
	invoke the call processing application in response to the detected network signaling events or trigger points (call setup procedure),	The call setup procedure comprises calling the Voice Channel Manager 206 to allocate a DSP for connection to the receiving voice port via the PCM bus. Further, the call setup procedure signals the Voice Signal Process module 202 to enable the allocated DSP to start the DTMF sampling for the dialing digits. Upon collection of enough digits by consulting the dial-mapper 208, a setup message is provided to the Tandem Switch Module 210. The Tandem Switch Module 210, using the Dial-peer and Dial-mapper 208, locates the permanent victual connection (PVC) number of the remote extension in order to provide a setup message. (Col. 7, ll. 32-45)	
	the call processing application operating according to parameters defined in the service profiles	The MAC, in providing voice services, supports a wide range of call management configurations comprising configurations for dial on-net extensions, off-net numbers, and local calls, but the embodiment is not so limited. The MAC supports call set-up options comprising, but not limited to: on-net dialing; auto-dial, or private line automatic ringdown (PLAR); off-net dialing; and tandem switching. (Col. 15, l. 63-Col. 16, l. 3).	
	wherein the network device consists of one or more customer premise equipment modules (MAC 100 connects to a PBX 198).	"The MAC of an embodiment comprises two PIMs that are used for network access, the DVM and AVM. The DVM, functionally equivalent to the MTM, provides connectivity to a digital private branch exchange 198 or channel bank. The voice channels can be either mapped to time slots on the primary PCM bus for voice compression or mapped to time slots on another PCM bus through a cross-connect switch." (Col. 10, ll. 1-7)	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to	**SIP user agent:** "FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The	**Base System—** Chung discloses a network device for establishing a voice-over-packet network

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	provide a SIP user agent to represent a telephone that uses the telephone line interface.	internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28)."	architecture (*e.g.*, MAC 100). **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 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

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			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 Chung to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Chung 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 (SNMP-based MIB), the system management platform collecting call log data from a plurality of network devices ; and	Management and configuration of the MAC of one embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13) "Furthermore, the routing table includes information elements comprising a preference-order and a last-use timestamp. The last-use timestamp comprises a last update to entry timestamp and an entry creation timestamp, but the embodiment is not so limited. The preference-order defines an explicit selection priority that should be give to each table entry. The last use timestamp tracks the last time a call was routed to the entry. The last use timestamp, in tracking successfully completed calls and call attempts, is an important attribute in that it allows load balancing." (Col. 26, ll. 17-26)	

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
		embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13).	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	**SIP signaling:** "FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28)."	**Base System**— Chung discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, MAC 100). **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

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'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
			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 Chung to include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Chung 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:** The MAC supports analog voice streams using E&M, FXS, and FXO voice signal standards. (Col. 7, ll. 2-7) In operation, an analog voice port or digital voice port provides a voice signal to the Voice Signal Process module 202. The Voice Signal Process module 202 translates the voice signal to a call setup message. (Col. 7, ll. 29-32).	**Base System**— Chung discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, MAC 100). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an

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'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
			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 obvious to modify the base system of Chung to

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'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
			include these well-known claimed SIP components and apply the well-known techniques taught by Osterhout, for example, to enable the telephones in Chung to participate in SIP-based telephony systems.

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		202. The Voice Signal Process module 202 translates the voice signal to a call setup message. (Col. 7, ll. 29-32).	
	invoke the call processing application in response to the detected network signaling events or trigger points (call setup procedure),	The call setup procedure comprises calling the Voice Channel Manager 206 to allocate a DSP for connection to the receiving voice port via the PCM bus. Further, the call setup procedure signals the Voice Signal Process module 202 to enable the allocated DSP to start the DTMF sampling for the dialing digits. Upon collection of enough digits by consulting the dial-mapper 208, a setup message is provided to the Tandem Switch Module 210. The Tandem Switch Module 210, using the Dial-peer and Dial-mapper 208, locates the permanent victual connection (PVC) number of the remote extension in order to provide a setup message. (Col. 7, ll. 32-45)	
	the call processing application operating according to parameters defined in the service profiles	The MAC, in providing voice services, supports a wide range of call management configurations comprising configurations for dial on-net extensions, off-net numbers, and local calls, but the embodiment is not so limited. The MAC supports call set-up options comprising, but not limited to: on-net dialing; auto-dial, or private line automatic ringdown (PLAR); off-net dialing; and tandem switching. (Col. 15, l. 63-Col. 16, l. 3).	
	wherein the network device consists of one or more customer premise equipment modules (MAC 100 connects to a PBX 198).	"The MAC of an embodiment comprises two PIMs that are used for network access, the DVM and AVM. The DVM, functionally equivalent to the MTM, provides connectivity to a digital private branch exchange 198 or channel bank. The voice channels can be either mapped to time slots on the primary PCM bus for voice compression or mapped to time slots on another PCM bus through a cross-connect switch." (Col. 10, ll. 1-7)	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to	SIP user agent: "FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The	Base System—Chung discloses a network device for establishing a voice-over-packet network

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'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
	provide a SIP user agent to represent a telephone that uses the telephone line interface.	internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28)."	architecture (*e.g.*, MAC 100). **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

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			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 Chung to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Chung* 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 (SNMP-based MIB), the system management platform collecting call log data from a plurality of network devices ; and	Management and configuration of the MAC of one embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13). "Furthermore, the routing table includes information elements comprising a preference-order and a last-use timestamp. The last-use timestamp comprises a last update to entry timestamp and an entry creation timestamp, but the embodiment is not so limited. The preference-order defines an explicit selection priority that should be give to each table entry. The last use timestamp tracks the last time a call was routed to the entry. The last use timestamp, in tracking successfully completed calls and call attempts, is an important attribute in that it	

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		Management and configuration of the MAC of one embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13).	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	**SIP signaling:** "FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28)."	**Base System**—Chung discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, MAC 100). **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

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			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 Chung to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Chung* to participate in SIP-based telephony systems.
Claim 16	The method of claim 13, wherein the storage medium further stores processor-executable	**SIP proxy server:** The MAC supports analog voice streams using E&M, FXS, and FXO voice signal standards. (Col. 7,	**Base System**—Chung discloses a network device for establishing a voice-over-packet network

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
	Instructions to act as an SIP proxy server for devices using the telephone line interface and for devices using the computer data interface.	ll. 2-7) In operation, an analog voice port or digital voice port provides a voice signal to the Voice Signal Process module 202. The Voice Signal Process module 202 translates the voice signal to a call setup message. (Col. 7, ll. 29-32).	architecture (*e.g.*, MAC 100). **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

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
			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 Chung to include these well-known claimed SIP components and apply the well-known techniques taught by Wengrovitz, for example, to enable the telephones in *Chung* to participate in SIP-based telephony systems.

Chung in view of Girard-SIP

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
		202. The Voice Signal Process module 202 translates the voice signal to a call setup message. (Col. 7, ll. 29-32).	
	invoke the call processing application in response to the detected network signaling events or trigger points **(call setup procedure)**,	The call setup procedure comprises calling the Voice Channel Manager 206 to allocate a DSP for connection to the receiving voice port via the PCM bus. Further, the call setup procedure signals the Voice Signal Process module 202 to enable the allocated DSP to start the DTMF sampling for the dialing digits. Upon collection of enough digits by consulting the dial-mapper 208, a setup message is provided to the Tandem Switch Module 210. The Tandem Switch Module 210, using the Dial-peer and Dial-mapper 208, locates the permanent victual connection (PVC) number of the remote extension in order to provide a setup message. (Col. 7, ll. 32-45)	
	the call processing application operating according to parameters defined in the service profiles	The MAC, in providing voice services, supports a wide range of call management configurations comprising configurations for dial on-net extensions, off-net numbers, and local calls, but the embodiment is not so limited. The MAC supports call set-up options comprising, but not limited to: on-net dialing; auto-dial, or private line automatic ringdown (PLAR); off-net dialing; and tandem switching. (Col. 15, l. 63-Col. 16, l. 3).	
	wherein the network device consists of one or more customer premise equipment modules **(MAC 100 connects to a PBX 198)**.	"The MAC of an embodiment comprises two PIMs that are used for network access, the DVM and AVM. The DVM, functionally equivalent to the MTM, provides connectivity to a digital private branch exchange 198 or channel bank. The voice channels can be either mapped to time slots on the primary PCM bus for voice compression or mapped to time slots on another PCM bus through a cross-connect switch." (Col. 10, ll. 1-7)	
Claim 6	The network device of claim 1, wherein the instructions further cause the network device to	**SIP user agent:** "FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The	**Base System**—Chung discloses a network device for establishing a voice-over-packet network

Chung in view of Girard-SIP

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
	provide a SIP user agent to represent a telephone that uses the telephone line interface.	internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28)."	architecture (*e.g.*, MAC 100). **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: "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.

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	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 Chung to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Chung* 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 (**SNMP-based MIB**), the system management platform collecting call log data from a plurality of network devices ; and	Management and configuration of the MAC of one embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13). "Furthermore, the routing table includes information elements comprising a preference-order and a last-use timestamp. The last-use timestamp comprises a last update to entry timestamp and an entry creation timestamp, but the embodiment is not so limited. The preference-order defines an explicit selection priority that should be give to each table entry. The last use timestamp tracks the last time a call was routed to the entry. The last use timestamp, in tracking successfully completed calls and call attempts, is an important attribute in that it allows load balancing." (Col. 26, ll. 17-26)	

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
		embodiment is designed to be compatible with existing network router management systems. As such, three types of configuration interfaces are provided, wherein the configuration interfaces comprise a command line interface, a HTTP-based configuration server, and a SNMP-based MIB ... The SNMP MIB allows management of the MAC from SNMP mangers, but the embodiment is not so limited. (Col. 17, ll. 6-13).	
Claim 15	The method of claim 13, wherein the routing of telephone calls includes SIP signaling.	**SIP signaling:** "FIG. 2 is a voice processing subsystem 200 of a MAC of an embodiment of the present invention. The internetwork software operating system that provides kernel services, network services and routing capability comprises the voice processing subsystem 200... (Col. 7, ll. 24-28)."	**Base System**—Chung discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, MAC 100). **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

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
			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 Chung to include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Chung* 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:** The MAC supports analog voice streams using E&M, FXS, and FXO voice signal standards. (Col. 7, ll. 2-7) In operation, an analog voice port or digital voice port provides a voice signal to the Voice Signal Process module 202. The Voice Signal Process module 202 translates the voice signal to a call setup message. (Col. 7, ll. 29-32).	**Base System**—Chung discloses a network device for establishing a voice-over-packet network architecture (*e.g.*, MAC 100). **Known Technique**— A person having ordinary skill in the art in 2001 would have been well-aware of SIP technology since it became an

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
			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 obvious to modify the base system of Chung to

APPENDIX K4

Chung in view of Girard-SIP

'519 Claim	Claim Limitations	Chung U.S. Patent 6,584,108	Secondary References
			include these well-known claimed SIP components and apply the well-known techniques taught by Girard, for example, to enable the telephones in *Chung* to participate in SIP-based telephony systems.

APPENDIX L1

Oran

'519 Claim	Claim Limitations	Oran U.S. Patent 6,275,574
Claim 9	A network device **(gateway 18, 28)** comprising: a broadband network interface **(IP interface 38)**; a plurality of communication interfaces, including a telephone line interface **(telephony interface 36)** and a computer data interface **(interface that integrates dial plan mapper 20 to a variety of existing IP hosts)**;	The invention relates to telephone systems and more particularly to a dial plan mapper used for routing telephone calls to different telephone networks. Col. 1, ll. 4-5. The dial plan mapper 20 can be integrated into a variety of existing IP hosts such [as] VoIP routers, Voice over Frame Relay (VoFR) routers, and H.320/H.323 gateways. Other types of telephony systems connected to the IP network 22 may also use the dial plan mapper 20 and also come within the scope of this invention. Col. 3, ll. 16-18. "Telephones 14 are coupled either directly or through a PSTN 16, PBX 30, etc. telephony interface 36 to the session application 34. The telephony interface 36 consists of whatever circuitry abstracts the signaling part of the telephony hardware attached to the gateway." Col. 3, ll. 30-35.
	a processor **(processor in gateway 18, 28)**;	"FIG. 2A is a detailed diagram of one of the gateways 18 or 28 in the telephone system shown in FIG. 1. The gateways 18 and 28 include a session application 34 that responds to telephone calls. For example, session applications may comprise an integrated voice response system or a phone call routing system. These session applications are generally well known and therefore, not described in further detail." Col. 3, ll. 24-30.
	a machine-readable storage medium that stores processor-a machine-readable storage medium that stores processor-executable instructions to provide SIP agents **(dial plan mapper 20)**	The dial plan mapper 20 can be integrated into a variety of existing IP hosts such [as] VoIP routers, Voice over Frame Relay (VoFR) routers, and H.320/H.323 gateways. Other types of telephony systems connected to the IP network 22 may also use the dial plan mapper 20 and also come within the scope of this invention. Col. 3, ll. 16-18.
	the instructions causing the network device to provide a SIP user agent to represent a non-SIP telephone that uses the telephone line interface **(dial plan mapper 20)**, and the instructions further causing the network device to implement a SIP proxy server **(session application 34)** that mediates all SIP communications over the broadband network interface involving the non-SIP telephone.	FIG. 2A is a detailed diagram of one of the gateways 18 or 28 in the telephone system shown in FIG. 1. The gateways 18 and 28 include a session application 34 that responds to telephone calls. Col. 3, ll. 24-27. Telephones 14 are coupled either directly or through a PSTN, PBX 30, etc. telephony interface 36 to the session application 34. The telephony interface 36 consists of whatever circuitry abstracts the signaling part of the telephony hardware attached to the gateway. The IP network 22 is coupled through an IP

Oran

'519 Claim	Claim Limitations	Oran U.S. Patent 6,275,574
		interface 38 to the session application 34. The session application 34 provides call translation between the telephone interface and the IP interface 38. Col. 3, ll. 31-39. FIG. 4 shows a call coming through a local telephone 14 or a telephony interface such as an FXS, FXO, PBX trunk, etc. The session application 34 begins listening for a dial string when informed of a telephone off-hook condition by a telephony driver in telephony interface 36.... Each time a digit 40 arrives, it is appended to an accumulated dial string 42, and the dial plan mapper 20 returns either a match or an indication that the dial string cannot possibly match an entry... The dial plan mapper queries the configuration database 32 for the longest match 44. The match pattern 44 ... is used as an index to identify associated configuration information 46. The session application 34 uses the returned session protocol, and session target from the configuration information 46 to initiate a session with a destination for the presented dial string 42, as phone 15 on IP network 22. Col. 4, ll. 43-65. The session application can also process Session [Initiation] Protocol (SIP)invitations, and if the invited user is reachable from one of the local telephony interfaces, the user is called and an attempt is made to connect him to the conference. Col. 6, ll. 14-18. See FIG. 6 showing SIP as one of the available session protocol options. FIG. 6 is a table showing the different inputs and outputs of the dial plan mapper 20. Col. 7, ll. 4-5 ... Session _ protocol indicates what call-establishment procedure to use for communicating with the call destination. Col. 7, ll. 38-39.
Claim 10	The network device of claim 9, wherein the computer data interface passes IP data.	"The dial plan mapper allows normal circuit-switched telephones to be used with VoIP and allows existing dialing conventions to be used unchanged. As far as the dial plan mapper is concerned, there is no difference between a call originated locally on a phone connected to a host containing the dial plan mapper and a call originated through a PBX or PSTN switch connected to the host via either analog or digital trunks. When calls arrive over the IP network, the dial plan mapper maps from the destination telephone number provided in the session protocol (usually an E.164 number) to the proper local interface (local FXS line or

Oran

'519 Claim	Claim Limitations	Oran U.S. Patent 6,275,574
		outbound trunk) for completing the call over the legacy voice network." Col. 2, ll. 20-25.
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	The dial plan mapper 20 can be integrated into a variety of existing IP hosts such [as] VoIP routers, Voice over Frame Relay (VoFR) routers, and H.320/H.323 gateways. Other types of telephony systems connected to the IP network 22 may also use the dial plan mapper 20 and also come within the scope of this invention. Col. 3, ll. 16-18.
Claim 12	The network device of claim 9, wherein the network device is contained in a single physical enclosure.	Gateway 18 is contained in a single physical enclosure. Figure 2A.

'519 Claim	Claim Limitations	Oran U.S. Patent 6,275,574	Secondary References
		protocol options. FIG. 6 is a table showing the different inputs and outputs of the dial plan mapper 20. Col. 7, ll. 4-5 ... Session _ protocol indicates what call-establishment procedure to use for communicating with the call destination. Col. 7, ll. 38-39.	
Claim 11	The network device of claim 9, wherein the plurality of interfaces includes a video streaming device interface.	The dial plan mapper 20 can be integrated into a variety of existing IP hosts such [as] VoIP routers, Voice over Frame Relay (VoFR) routers, and H.320/H.323 gateways. Other types of telephony systems connected to the IP network 22 may also use the dial plan mapper 20 and also come within the scope of this invention. Col. 3, ll. 16-18.	To the extent that Oran 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 Oran to provide an interface for video

'519 Claim	Claim Limitations	Oran U.S. Patent 6,275,574	Secondary References
			conferencing data, for example.

'519 Claim	Claim Limitations	Inbar U.S. Patent 6,885,660
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 (**Master Server 6**), the system management platform collecting call log data from a plurality of network devices ; and	Master-Server 6 is a system management platform with layers for managing calls. Figure 3; Col. 6, ll. 44-51 "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. 54-61.
	distributing the plurality of network devices (**IP Centers 2**) that each include a telephone line interface, a computer data interface, a broadband network interface terminating a link from the shared packet network, (**IP Centers 2 include a telephone line interface, a computer data interface, and a broadband network interface terminating a link from Internet 5**)	IPCenter 2 includes a number of interfaces for communication devices including a telephone line interface, a computer data interface, and a broadband network interface. "The IPCenter preferably has a plurality of interfaces and connections. A WAN interface 19 may be used to connect to the Internet using cable, satellite, cellular, wireless, power lines or dial up modem, and a LAN interface 20 may connect the unit to a local area network such as LAN 3c of FIG. 2. The IPCenter unit preferably has a number of interfaces 21 for individual kinds of communication devices e.g. audio, telephone, video and data and may additionally comprise a number of physical interfaces 22 for card readers, IrDA, appliances I/O, display and others physical devices." Figure 4; Col. 7, ll. 9-19.
	a processor (**CPU 13**);	CPU 13 (Figure 4).
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**Memory 15 and software to control telephone calls**),	IPCenter 2 includes memory 15 and software to control telephone calls. "The IPCenter further comprises software to perform and manage the various IPCenter activities, as well as memory 15, a security system 16, backup unit 17, and a converter unit 18, which is able to carry out D/A and A/D conversions." Figure 4; Col. 7, ll. 5-9.
	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 (**IPCenter 2 routes telephone calls in a peer-to-peer**	IPCenter 2 routes telephone calls in a peer-to-peer fashion and sends call log data to Master Server 6.

'519 Claim	Claim Limitations	Inbar U.S. Patent 6,885,660
	fashion and sends data to Master Server 6).	"The IPCenter of the above embodiments is thus able to use IP addresses received from the central unit to support End-to-End or peer to peer communication. Subscribers are thereby enabled to make direct connections to each other, thus avoiding the need to make use of Intermediate servers." Col. 7, ll. 20-24. "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. 54-61.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"The IPCenter preferably has a plurality of interfaces and connections. A WAN interface 19 may be used to connect to the Internet using cable, satellite, cellular, wireless, power lines or dial up modem." Figure 4; Col. 7, ll. 9-12.
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	"The present embodiments provide a peer to peer configuration for communication using any kind of media. The configuration uses querying to obtain an IP address from a central server, and then allows for a direct peer to peer connection to be formed using the IP address obtained." Col. 5, ll.6-12.
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface.	"Preferably, said subscriber electronic devices are any one of a group comprising: a magnetic card reader, a smart card reader, a security sensor, a meter, an electronic utility, a video camera, a television, a Wireless device including a Bluetooth or other wireless local loop device, a telephone, a fax machine, a cellular telephone, a personal digital assistant, a portable computer and a desktop computer." Col. 2, ll. 54-60.

Kung

'619 Claim	Claim Limitations	Kung U.S. Patent 6,917,610
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 (**IP central station 200**), the system management platform collecting call log data from a plurality of network devices ; and	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.
	distributing the plurality of network devices (**BRG 300**) that each include a telephone line interface, a computer data interface, a broadband network interface terminating a link from the shared packet network, (**BRG 300 includes a telephone line interface, a computer data interface, and a broadband network interface**)	BRG 300 includes a number of interfaces for communication devices including a telephone line interface, a computer data interface, and a broadband network interface. "Again referring to FIG. 1, the broadband network 1 may include one or more customer premises equipment (CPE) units 102. The customer premise equipment 102 may be variously configured. In one example, the customer premise equipment 102 may include one or more local control devices such as a broadband residential gateway (BRG) 300. Although the broadband residential gateway is preferably disposed in a residence for many aspects of the invention, in exemplary embodiments, it may also be disposed in a business or other location. The broadband residential gateway 300 may be variously configured to provide one or more integrated communication interfaces to other devices within the customer premise equipment 102 such as televisions (TV), personal computers (PC), plain old telephone system (POTS) phone(s), video phones, IP enabled phones, and other devices. For example, the broadband residential gateway 300 may provide one or more telephone port connections (e.g., plain old telephone system), Ethernet connections, coaxial connections, fiber distributed data interface (FDDI) connections, wireless local area network (LAN) connections, firewire connections, and/or other connections to a plurality of devices such as plain old telephones, IP based phones, television converters, e.g., cable television (CATV) set top devices, televisions, digital televisions, high definition televisions (HDTV), video phones, and other devices.." Figure 1; Col. 4, ll. 34-59; Figure 3.
	a processor (**processor of broadband residential computer (BRG 300) or distributed processing controller 306**);	"e.g., broadband residential computer (BRG), personal computer (PC), etc." Col. 2, ll. 7-9. "distributed processing controller 306 which may be a microprocessor" Col. 18, ll. 56-47.

Kung

'619 Claim	Claim Limitations	Kung U.S. Patent 6,917,610
	a machine-readable storage medium storing processor-executable instructions to control telephone calls (**memory 322**),	"memory 322 (e.g., DRAM, RAM, flash, and/or other memory)" Col. 18, ll. 46-47.
	the instructions causing each network device to route telephone calls in a peer-to-peer fashion (**BRG 300 routes calls in a peer-to peer fashion over IP Network 120**) over the shared packet. and to send call log data to the system management platform (**IP central station 200**).	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; Col. 35, l. 37 - col. 36, l. 10.
Claim 14	The method of claim 13, wherein for each device the broadband network interface terminates a link from the shared packet network.	"The IP network 120 and/or ATM network 185 may include one or more routers and/or other devices to route, for example, telephony calls, multimedia calls, signaling messages, administrative messages, programming messages and/or computer data between the various devices in the broadband network 1 such as the head-end hub 115, the public switched telephone network 160, the private branch exchange (PBX) 146, as well as the other devices discussed above. In preferred embodiments, the information traveling in the IP network 120 may be packetized and formatted in accordance with one of the Internet protocols. The IP network 120 may also include gateways to interface with the various other networks and/or devices." Figure 1; Col. 6, ll. 3-15.

'519 Claim	Claim Limitations	Kung U.S. Patent 6,917,610
Claim 17	The method of claim 13, wherein the shared packet network uses IP protocols.	"The IP network 120 and/or ATM network 185 may include one or more routers and/or other devices to route, for example, telephony calls, multimedia calls, signaling messages, administrative messages, programming messages and/or computer data between the various devices in the broadband network 1 such as the head-end hub 115, the public switched telephone network 160, the private branch exchange (PBX) 146, as well as the other devices discussed above. In preferred embodiments, the information traveling in the IP network 120 may be packetized and formatted in accordance with one of the Internet protocols. The IP network 120 may also include gateways to interface with the various other networks and/or devices." Figure 1; Col. 6, ll. 3-15.
Claim 19	The method of claim 13, wherein the plurality of network devices each further include a video streaming device interface (**BRG 300 has a video phone interface**).	BRG 300 includes a number of interfaces for communication devices including a telephone line interface, a computer data interface, and a broadband network interface. "Again referring to FIG. 1, the broadband network 1 may include one or more customer premises equipment (CPE) units 102. The customer premise equipment 102 may be variously configured. In one example, the customer premise equipment 102 may include one or more local control devices such as a broadband residential gateway (BRG) 300. Although the broadband residential gateway is preferably disposed in a residence for many aspects of the invention, in exemplary embodiments, it may also be disposed in a business or other location. The broadband residential gateway 300 may be variously configured to provide one or more integrated communication interfaces to other devices within the customer premise equipment 102 such as televisions (TV), personal computers (PC), plain old telephone system (POTS) phone(s), video phones, IP enabled phones, and other devices. For example, the broadband residential gateway 300 may provide one or more telephone port connections (e.g., plain old telephone system), Ethernet connections, coaxial connections, fiber distributed data interface (FDDI) connections, wireless local area network (LAN) connections, firewire connections, and/or other connections to a plurality of devices such as plain old telephones, IP based phones, television converters, e.g., cable television (CATV) set top devices, televisions, digital televisions, high definition televisions (HDTV), video phones, and other devices.." Figure 1; Col. 4, ll. 34-59.

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Appendix C3	Claims 13-19 are obvious over Osterhout in view of Inbar
Appendix C4	Claims 13-19 are obvious over Osterhout in view of Kung
Appendix D1	Claims 1-19 are anticipated by Wengrovitz
Appendix D2	Claims 7 and 8 are obvious over Wengrovitz in view of Chung
Appendix D3	Claims 2, 11, and 19 are obvious over Wengrovitz in view of Osterhout
Appendix D4	Claim 18 is obvious over Wengrovitz in view of Osterhout
Appendix D5	Claims 13-19 are obvious over Wengrovitz in view of Inbar
Appendix D6	Claims 13-19 are obvious over Wengrovitz in view of Kung
Appendix E1	Claims 1-5, 7, 8, 13, 14, and 17-19 are anticipated by Baratz
Appendix E2	Claims 9-12 are obvious over Baratz in view of Osterhout
Appendix E3	Claims 9-12 are obvious over Baratz in view of Wengrovitz
Appendix E4	Claims 9-12 are obvious over Baratz in view of Girard
Appendix E5	Claims 6, 15, and 16 are obvious over Baratz in view of Osterhout
Appendix E6	Claims 6, 15, and 16 are obvious over Baratz in view of Wengrovitz
Appendix E7	Claims 6, 15, and 16 are obvious over Baratz in view of Girard
Appendix E8	Claims 2, 11, and 19 are obvious over Baratz in view of Osterhout
Appendix E9	Claims 7 and 8 are obvious over Baratz in view of Chung
Appendix E10	Claim 5 is obvious over Baratz in view of Czajkowski
Appendix F1	Claims 1-5, 7, 8, 13, 14, and 17-19 are anticipated by Czajkowski
Appendix F2	Claims 9-12 are obvious over Czajkowski in view of Osterhout
Appendix F3	Claims 9-12 are obvious over Czajkowski in view of Wengrovitz
Appendix F4	Claims 9-12 are obvious over Czajkowski in view of Girard
Appendix F5	Claims 6, 15, and 16 are obvious over Czajkowski in view of Osterhout
Appendix F6	Claims 6, 15, and 16 are obvious over Czajkowski in view of Wengrovitz
Appendix F7	Claims 6, 15, and 16 are obvious over Czajkowski in view of Girard
Appendix F8	Claims 7 and 8 are obvious over Czajkowski in view of Chung
Appendix F9	Claims 13, 14, and 17-19 are obvious over Czajkowski in view of Inbar
Appendix F10	Claims 13, 14, and 17-19 are obvious over Czajkowski in view of Kung
Appendix G1	Claims 1-5, 7, 8, 13, 14, and 17-19 are anticipated by Gerszberg
Appendix G2	Claims 9-12 are obvious over Gerszberg in view of Osterhout
Appendix G3	Claims 9-12 are obvious over Gerszberg in view of Wengrovitz
Appendix G4	Claims 9-12 are obvious over Gerszberg in view of Girard

Office Action Appendix
4-2-2009

Control Number 95/000,344 Part of Paper No. 20090402

Appendix	Description
Appendix G5	Claims 6, 15, and 16 are obvious over Gerszberg in view of Osterhout
Appendix G6	Claims 6, 15, and 16 are obvious over Gerszberg in view of Wengrovitz
Appendix G7	Claims 6, 15, and 16 are obvious over Gerszberg in view of Girard
Appendix G8	Claims 7 and 8 are obvious over Gerszberg in view of Chung
Appendix H1	Claims 1-19 are anticipated by Janning
Appendix H2	Claims 7 and 8 are obvious over Janning in view of Chung
Appendix H3	Claim 18 is obvious over Janning in view of Osterhout
Appendix H4	Claims 13-19 are obvious over Janning in view of Inbar
Appendix H5	Claims 13-19 are obvious over Janning in view of Kung
Appendix I1	Claims 1-5, 7, 8, 13, 14, and 17-19 are anticipated by Nodoushani
Appendix I2	Claims 9-12 are obvious over Nodoushani in view of Osterhout
Appendix I3	Claims 9-12 are obvious over Nodoushani in view of Wengrovitz
Appendix I4	Claims 9-12 are obvious over Nodoushani in view of Girard
Appendix I5	Claims 6, 15, and 16 are obvious over Nodoushani in view of Osterhout
Appendix I6	Claims 6, 15, and 16 are obvious over Nodoushani in view of Wengrovitz
Appendix I7	Claims 6, 15, and 16 are obvious over Nodoushani in view of Girard
Appendix I8	Claims 2, 11, and 19 are obvious over Nodoushani in view of Osterhout
Appendix I9	Claims 7 and 8 are obvious over Nodoushani in view of Chung
Appendix J1	Claims 1-5, 7, 8, 13, 14, and 17-19 are anticipated by Chow
Appendix J2	Claims 9-12 are obvious over Chow in view of Osterhout
Appendix J3	Claims 9-12 are obvious over Chow in view of Wengrovitz
Appendix J4	Claims 9-12 are obvious over Chow in view of Girard
Appendix J5	Claims 6, 15, and 16 are obvious over Chow in view of Osterhout
Appendix J6	Claims 6, 15, and 16 are obvious over Chow in view of Wengrovitz
Appendix J7	Claims 6, 15, and 16 are obvious over Chow in view of Girard
Appendix K1	Claims 1-5, 7, 8, 13, 14, and 17-19 are anticipated by Chung
Appendix K2	Claims 6, 15, and 16 are obvious over Chung in view of Osterhout
Appendix K3	Claims 6, 15, and 16 are obvious over Chung in view of Wengrovitz
Appendix K4	Claims 6, 15, and 16 are obvious over Chung in view of Girard
Appendix L1	Claims 9-12 are anticipated by Oran
Appendix L2	Claim 11 is obvious over Oran in view of Osterhout
Appendix M1	Claims 13, 14, 17, and 19 are anticipated by Inbar
Appendix N1	Claims 13, 14, 17, and 19 are anticipated by Kung