

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

CISCO SYSTEMS INC.,

Petitioner

IPR2024-01282

U.S. Patent No. 7,738,368

**DECLARATION OF HENRY H. HOUEH, PH.D.,
UNDER 37 C.F.R. § 1.68 IN SUPPORT OF PETITION
FOR *INTER PARTES* REVIEW**

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I. INTRODUCTION

1. I, Henry H. Houh, have been retained by counsel for Cisco Systems Inc. (“Petitioner”) as a technical expert in connection with the proceeding identified above. I submit this declaration in support of Cisco’s Petition for *Inter Partes* Review (“IPR”) of U.S. Patent No. 7,738,368 (“the ’368 patent”).

2. I am being compensated for my work in this matter at an hourly rate. I am also being reimbursed for reasonable and customary expenses associated with my work and testimony in this matter. My compensation is not contingent on the outcome of this matter or the specifics of my testimony. I have no personal or financial stake or interest in the outcome of this proceeding.

3. I have been asked to provide my opinions regarding whether the subject matter of claims 1 and 2 (the “Challenged Claims”) of the ’368 patent would have been obvious to a person having ordinary skill in the art (“POSITA”) as of the earliest claimed priority date. It is my opinion that the Challenged Claims would have been obvious to a POSITA after reviewing the prior art, as discussed below.

4. In the preparation of this declaration, I have considered:

- (1) the ’368 patent, Ex.1001;
- (2) the prosecution history of the ’368 patent, Ex.1002;
- (3) U.S. Patent No. 7,643,414 (“Minhazuddin”), Ex.1005;
- (4) U.S. Patent Pub. No. 2004/0073690, (“Hepworth”), Ex.1006;

- (5) U.S. Patent No. 7,023,839 (“Shaffer”), Ex.1007;
- (6) U.S. Patent Pub. No. 2004/0032860, (“Mundra”) Ex.1008;
- (7) U.S. Patent No. 7,830,813 (“Lo”), Ex.1009;
- (8) Metro Ethernet Networks – A Technical Overview, Ex.1010;
- (9) Introduction to Cisco Network Design, Ex.1011;
- (10) Network Systems in CIM Environment, Ex.1012;
- (11) Avaya IP Office-Job Aid; Connecting an IP600 and IP Office via H323 IP Trunk, Ex.1013;
- (12) Avaya IP Office - Small Office Edition 2003, Ex.1014;
- (13) Merrill Lynch Selects Avaya IP Telephony Systems, Ex.1015;
- (14) Avaya IP Softphone Release 4 Ver. 1, Ex.1016;
- (15) Avaya Phone Manager 2.1, Ex.1017;
- (16) Avaya Definity[®] 4624 IP Telephone, Ex.1018;
- (17) VoIP for Dummies[®], Ex.1019;
- (18) U.S. Patent No. 7,657,021, Ex.1020;
- (19) What Is Real-Time Transport Control Protocol (RTCP), Ex.1021;
- (20) Combining Voice over IP with Policy-Based Quality of Service, Ex.1022;
- (21) Access Networks and Physical Media, Ex.1023;
- (22) Assessment of objective voice quality over best-effort networks, Ex.1024;
- (23) U.S. Patent No. 7,764,679 (“MeLampy”), Ex.1025;
- (24) M. Arango, et al., “Media Gateway Control Protocol (MGCP) Version 1.0,” IETF Request for Comments 2705 (October 1999); Ex.1026;
- (25) F. Andreasen, et al., “Media Gateway Control Protocol (MGCP) Version 1.0,” IETF Request for Comments 3435 (January 2003); Ex.1027;

- (26) B. Foster, et al., “Basic Media Gateway Control Protocol (MGCP) Packages,” IETF Request for Comments 3660 (December 2003);
Ex.1028;
- (27) U.S. Patent Pub. No. US2005/0276272 (“Arai”), Ex.1033;
- (28) U.S. Patent Pub. No. US2005/0232309 (“Kavaler”), Ex.1034;
- (29) U.S. Patent Pub. No. US2005/0190746 (“Xiong”), Ex.1035;
- (30) U.S. Patent Pub. No. US2006/0291643 (“Pfaff”), Ex.1036;
- (31) U.S. Patent Pub. No. US2006/0221912 (“Olivier”), Ex.1037;
- (32) U.S. Patent No. 7,881,712 (“Luo”), Ex.1038;
- (33) U.S. Patent No. 7,715,359 (“Lin”), Ex.1039;
- (34) U.S. Patent Pub. No. US2004/0083128 (“Buckingham”), Ex.1041;
- (35) U.S. Patent Pub. No. US2003/0093563 (“Young”), Ex.1042;
- (36) U.S. Patent Pub. No. US2006/0037072 (“Rao”), Ex.1043;
- (37) U.S. Patent Pub. No. US2004/0114581 (“Hans”), Ex.1044;
- (38) U.S. Patent No. 6,480,122 (“Oddy”), Ex.1045;
- (39) U.S. Patent No. 6,898,188 (“Hamami”), Ex.1046;
- (40) U.S. Patent No. 6,904,038 (“Moon”), Ex.1047;
- (41) U.S. Patent No. 6,442,032 (“Linares”), Ex.1048;
- (42) U.S. Patent Pub. No. US2006/0233158 (“Croak”), Ex.1049
- (43) U.S. Patent Pub. No. US2005/0053055 (“Horvath”), Ex.1050;
- (44) U.S. Patent Pub. No. US2005/0195949 (“Frattura”), Ex.1051;
- (45) U.S. Patent Pub. No. US2005/0207432 (“Velez-Rivera”), Ex.1052;
- (46) U.S. Patent Pub. No. US2005/0174993 (“Gass”), Ex.1053;
- (47) U.S. Patent Pub. No. US2004/0240432 (“Ving”), Ex.1054;
- (48) U.S. Patent Pub. No. US2004/0233898 (“Otsuka”), Ex.1055;
- (49) U.S. Patent Pub. No. US2002/0191557 (“Chow”), Ex.1056;
- (50) U.S. Patent Pub. No. US2003/0128670 (“Pinier”), Ex.1057;

- (51) U.S. Patent Pub. No. US2005/0015460 (“Gole”), Ex.1058;
- (52) U.S. Patent No. 6,757,363 (“Platt”), Ex.1059;
- (53) U.S. Patent Pub. No. US2002/0064178 (“Christie”), Ex.1060;
- (54) U.S. Patent No. 7,024,475 (“Abaye”), Ex.1061;
- (55) U.S. Patent Pub. No. US2006/0056630 (“Zimmer”), Ex.1062;
- (56) U.S. Patent Pub. No. US2004/0073597 (“Caveney”), Ex.1065;
- (57) U.S. Patent Pub. No. US2005/0078700 (“Thompson”), Ex.1068;
- (58) U.S. Patent Pub. No. US2006/0215680 (“Camagna”), Ex.1069;
- (59) U.S. Patent Pub. No. US2005/0282432 (“Murr”), Ex.1070;
- (60) U.S. Patent Pub. No. US2006/0198360 (“Biage”), Ex.1072;
- (61) U.S. Patent Pub. No. US2003/0086425 (“Bearden”), Ex.1073;
- (62) U.S. Patent Pub. No. US2003/0093513 (“Hicks”), Ex.1074;
- (63) U.S. Patent Pub. No. US2002/0094081 (“Medvinsky”), Ex.1075;
- (64) U.S. Patent No. 6,885,658 (“Ress”), Ex.1076;
- (65) U.S. Patent Pub. No. US2002/0101881 (“Sundaresan”), Ex.1077;
- (66) U.S. Patent Pub. No. US2004/0001493 (“Cloonan”), Ex.1078;
- (67) U.S. Patent Pub. No. US2001/0053208 (“Pinard”), Ex.1079;
- (68) U.S. Patent No. 6,873,604 (“Surazski”), Ex.1083;
- (69) Robert M. Gray, “The 1974 Origins of VoIP”, IEEE Signal Processing Magazine (July 2005), Ex.1084;
- (70) H. Schulzrinne, “Voice Communication Across the Internet: A Network Voice Terminal”, CS Technical Report 92-50 (July 1992), Ex.1085;
- (71) University of Southern California 1982 Annual Technical Report, Information Sciences Institute (July 1981 – June 1982), Ex.1086;
- (72) Clifford J. Weinstein, “Experience with Speech Communication in Packet Networks,” Ex.1087;

- (73) U.S. Patent Pub. No. US2001/0055331 (“Agazzi”), Ex.1088;
- (74) U.S. Patent Pub. No. US2005/0084269 (“Dallesasse”), Ex.1089;
- (75) U.S. Patent No. 6,934,258 (“Smith”), Ex.1091;
- (76) U.S. Patent Pub. No. US2004/0215812 (“Lu”), Ex.1092;
- (77) U.S. Patent Pub. No. US2004/0061505 (“Fitzpatrick”), Ex.1093;
- (78) U.S. Patent Pub. No. US2003/0179761 (“Dobbins”), Ex.1094;
- (79) U.S. Patent No. 6,914,900 (“Komatsu”), Ex.1095;
- (80) U.S. Patent Pub. No. US2004/0252701 (“Anandakumar”), Ex.1096;
- (81) U.S. Patent Pub. No. US2003/0063569 (“Kalliokulju”), Ex.1097; and
- (82) S. Bradner, “The Internet Standards Process - - Revision 3,” IETF Request for Comments 2026 (October 1996), Ex.1098.

5. In forming the opinions expressed below, I have considered: the documents listed above; the relevant legal standards, including the standard for obviousness; and my own knowledge and experience based upon my work in the field of wireless communications as described below, as well as any additional materials cited herein.

6. Unless otherwise noted, all emphasis in any quoted material has been added.

II. QUALIFICATIONS

7. The details of my background and education, and a listing of all publications that I have authored, are provided in my *Curriculum Vitae*, a copy of

which is submitted as Ex.1004. The following is a brief summary of my relevant qualifications and professional experience.

8. I received a Ph.D. in Electrical Engineering and Computer Science from the Massachusetts Institute of Technology (“MIT”) in 1998. I also received a Master of Science degree in Electrical Engineering and Computer Science in 1991, a Bachelor of Science Degree in Electrical Engineering and Computer Science in 1989, and a Bachelor of Science Degree in Physics in 1990, all from MIT.

9. During my college studies, I focused on communications and data networking. I took specialized courses including graduate courses in telecommunications networks, optical communications, and data networking. I, along with other graduate students in a networking research group, maintained both the computer workstations and the networking devices in the research group.

10. I have worked in data networking and distributed networking systems on several occasions. As part of my doctoral research at MIT from 1991-1998, I worked as a research assistant in the Telemedia Network Systems (“TNS”) group at the Laboratory for Computer Science. The TNS group built a high-speed gigabit ATM network and applications which ran over the network, such as remote video capture (including audio), processing and display on computer terminals. In addition to helping design the core network components (such as the ATM switch), designing and building the high-speed ATM links, and designing and writing the

device drivers for the interface cards, I also set up the group's web server, which at the time was one of the first several hundred web servers in existence. Our high-speed data network carried multimedia data including video and audio data within ATM cells.

11. The TNS group was the first group to initiate a remote video display over the World Wide Web. Vice President Al Gore visited our group in 1996 and received a demonstration of—and remotely drove—a radio-controlled toy car with a wireless video camera mounted on it; the video was encoded by TNS-designed hardware, streamed over the TNS-designed network and displayed using TNS-designed software.

12. I authored or co-authored twelve papers and conference presentations on our group's research. I also co-edited the final report of the gigabit networking research effort with Professor David Tennenhouse and Senior Research Scientist David Clark. David Clark is generally considered to be one of the fathers of the Internet Protocol and served as Chief Protocol Architect for the Internet. With its focus on networking, the group, including myself, set up and maintained the network and computer systems. These systems included the networking on the workstations and desktops, the distributed file system, desktops and workstations, setting up and maintaining the distributed file system (Network File System) and

the authentication system (Network Information Service, formerly known as Yellow Pages).

13. I defended and submitted my Ph.D. thesis, titled “Designing Networks for Tomorrow’s Traffic,” in January 1998. As part of my thesis research, I analyzed local-area and wide-area flows to show a more efficient method for routing packets in a network, based on traffic patterns at the time. My thesis involved analyzing flows of data in the network and the routing efficiencies gained by labeling the flows at the edge of the network, and routing the data based on the label instead of the IP destination address. My analysis included different methods of setting up the granularity of the flows for the most efficient use of routing resources. This type of label switching became popular with Multiprotocol Label Switching (MPLS) later implemented in various commercial routers. I gathered a large amount of network traffic, used publicly available network traces, and broke the traffic into different types of flows using various parameters for classifying the flows. My flow analysis is applicable to the flow classification at the ingress to an MPLS domain. My thesis also addressed real-time streamed audio and video. The network traffic that I analyzed was IP protocol traffic, including UDP and TCP.

14. I have been involved in Voice over Internet Protocol (“VoIP”) data streaming technologies since 1997 and have specific experience in designing and testing VoIP networks.

15. From 1997 to 1999, I worked at NBX Corporation, which was acquired by 3Com Corporation in 1999. During this time, I was a Senior Scientist and Engineer working in IP Telephony. NBX delivered the world's first fully featured business telephone system to run over a data network, the NBX100. NBX was one of the first business phone systems to be configurable via a web interface. Users and administrators had access to varying levels of configuration for the phone system.

16. As part of my work at NBX, I designed the core audio reconstruction algorithms for the telephones which depacketized the voice data and reconstructed the audio. In addition, I designed the voice data packet transmission algorithms. I created a system to capture and analyze network packets sent by devices in the NBX system for aid in testing and debugging. I also designed and validated the core packet transport protocol used by the phone system. In addition, I designed and oversaw the development of the underlying transport protocol used by the NBX100 phone system for reliable packet transport, used in the system to communicate from the network-based telephones to the call control unit. I wrote NBX's first demonstration IP software stack, which added the capability for utilizing the NBX100 phone system on an IP network. I also specified and prototyped the phone system's support for Differentiated Services. NBX first demonstrated a phone in the NBX100 system working over the Internet in 1998 at a trade show in Las Vegas,

for which I acquired, set up, and configured Virtual Private Network tunnels to carry the traffic. I was later the lead architect in designing NBX's next-generation highly scalable system. After NBX was acquired by 3Com, I did some work with 3Com's cable equipment division, including demonstrating a working NBX IP phone system over 3Com's cable equipment infrastructure using an early version of DOCSIS at a trade show in 1999. The NBX100 was the market's leading business phone system to run on a data network for several years following its introduction. During that time, I became more familiar with the various standards relevant to Internet telephony as well as the problems which designers of commercial telephony operations were faced with in implementing VoIP.

17. I, along with two of NBX's founders, were awarded U.S. Patent No. 6,967,963 titled "Telecommunication Method for Ensuring On-Time Delivery of Packets Containing Time-Sensitive Data," for some of the work we did while at NBX.

18. After NBX, I worked at Teradyne, a test tool company primarily focused on semiconductors. Teradyne had recently acquired Hammer, a company that specialized in load and functional testing for telecommunications systems. The Hammer product is well known as a telecom test tool. Teradyne spun out Hammer and several other internal divisions into an independent company called Empirix. I

became Chief Technologist of the Hammer division of Empirix. Empirix was a leader in VoIP network testing and monitoring.

19. At Empirix, I laid out a new multi-year product vision for data network testing, secured internal funding for the effort, and led a team to deliver a new technology platform to the market in February 2001. This new product, PacketSphere, initially emulated network behavior so that wide-area VoIP connections could be tested in a lab. The PacketSphere allowed the packet loss, jitter, and reordered packet rates to be adjusted to emulate the behavior of a real-world network inside the test lab. A later release allowed PacketSphere to generate high volumes of VoIP calls, including media streams, and to monitor the quality of VoIP voice streams. PacketSphere was also used in the Storage Area Network area. Later, the core technology was added to other Empirix products such as Empirix's Hammer XMS to monitor thousands of VoIP media streams in real time to determine their quality. PacketSphere was Empirix's most successful new platform introduction at the time. Companies purchased the PacketSphere product to emulate an Internet Protocol network to see the effects of deploying their product on the Internet prior to launch. PacketSphere received several industry awards.

20. During my time at Empirix, I presented lectures on VoIP and data network testing to companies including Lucent Labs (formerly AT&T Bell Labs). I was also invited to present several guest lectures in a software engineering course at

MIT. Since then, I have also participated twice as a unit lecturer in an experimental course that was taught by an Institute Professor (the highest award that a MIT Professor can achieve) and sponsored by the Chairman of the MIT Corporation (MIT's board of trustees).

21. From 2004 to 2008, I was employed by BBN Technologies Corp., a technology research and development company located in Cambridge, Massachusetts. BBN Technologies is a world-renowned company with expertise in acoustics, speech recognition, and communications technology. BBN Technologies staff have pioneered many internetworking technologies and Internet applications and have built some of the world's largest government and commercial data networks.

22. My duties and responsibilities at BBN Technologies generally included commercialization of the technologies developed by BBN Technologies, which included spinning off companies and growing commercial businesses in-house. More particularly, I was involved in utilizing the award-winning AVOKE STX speech recognition technology to create the public audio/video search engine EveryZing (formerly known as PodZinger) which was spun out into a stand-alone company now known as RAMP, Inc. PodZinger won the 2006 MITX Technology Award for best Web 2.0 Application and was also named the 2006 Forbes Favorite Video & Audio Search Engine, beating out Google, Yahoo, and other companies.

After managing the creation of the initial prototype system, PodZinger built out a full streaming audio and video search solution when I was the Vice President of Operations and Technology there. I was also involved in the Boomerang Mobile Shooter Detection project as the Vice President of Engineering for the program. The Boomerang system was deployed to Iraq and Afghanistan and was credited with saving many lives.

23. In 2012, I opened Einstein's Workshop, a 7,000 square foot facility for teaching science, technology, engineering, and math to children. I installed and configured the telephone system, designed and programmed the website, and designed and configured the network, which has grown to roughly 100 computers, multiple WiFi access points, firewalls, multiple wireless networks, and multiple facilities. We also created an educational 3D Computer-Aided Design program, which we spun-out into a separate company, BlocksCAD. BlocksCAD has received grants from DARPA DSO and the USDA and has participated in the LearnLaunch, MIT Play Labs, and MassChallenge accelerator programs. BlocksCAD currently has over 150,000 registered users and is used in schools throughout the US.

24. From 1989 to 1990, I worked at AT&T Bell Laboratories on optical computers. This work generated six peer-reviewed papers, and multiple U.S. and European patent applications in which I was named as a co-author or inventor. I

also interned at AT&T Bell Laboratories in 1987 and 1988. Additional relevant experience in the field of computer networking is listed in my *Curriculum Vitae*.

25. I am a named inventor on several patents and published patent applications that are related to VoIP technology including: U.S. Patent No. 6,967,963, entitled “Telecommunication Method for Ensuring On-Time Delivery of Packets Containing Time-Sensitive Data”; U.S. Patent Application Publication No. 20020015387, entitled “Voice Traffic Packet Capture and Analysis Tool for a Data Network”; U.S. Patent Application Publication No. 20020016708, entitled “Method and Apparatus for Utilizing a Network Processor as Part of a Test System”; U.S. Patent Application Publication No. 20020016937, entitled “Method and Apparatus for Utilizing a Network Processor as Part of a Test System”; and U.S. Patent No. 7,590,542, entitled “Method of Generating Test Scripts Using a Voice-Capable Markup Language.”

III. LEVEL OF ORDINARY SKILL IN THE ART

26. I understand there are multiple factors relevant to determining the level of ordinary skill in the pertinent art, including (1) the levels of education and experience of persons working in the field at the time of the invention; (2) the sophistication of the technology; (3) the types of problems encountered in the field; and (4) the prior art solutions to those problems.

27. A POSITA in the field of the '368 patent, as of the earliest claimed priority date of November 10, 2005, would have been someone knowledgeable and familiar with packet-based communication network techniques available at the time and VoIP technologies. Such a POSITA would have a bachelor's degree in computer science, computer engineering, electrical engineering, or equivalent training, and approximately two years of experience working in the field of network communications, including experience with VoIP. Additional work experience can substitute for specific educational background, and vice versa.

28. For purposes of this Declaration, in general, and unless otherwise noted, my statements and opinions, such as those regarding my own experience and what a POSITA would have understood or known generally (and specifically related to the references I consulted herein), reflect the knowledge that existed in the relevant field as of the priority date of the '368 patent.

IV. RELEVANT LEGAL STANDARDS

29. I am not an attorney. In preparing and expressing my opinions and considering the subject matter of the '368 patent, I am relying on certain legal principles that counsel has explained to me.

30. I understand that prior art to the '368 patent includes patents and printed publications in the relevant art that predate the priority date of the '368

patent. For purposes of this Declaration, I am applying the earliest claimed priority date of November 10, 2005, as the priority date of the '368 patent.

31. I have been informed by Cisco's counsel that a claimed invention is unpatentable under 35 U.S.C. § 103 if the differences between the claimed invention and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a POSITA. I have also been informed by Cisco's counsel that the obviousness analysis considers factual inquiries, including the level of ordinary skill in the art, the scope and content of the prior art, and the differences between the prior art and the claimed subject matter.

32. I have been further informed by Cisco's counsel that there are several recognized rationales for combining references or modifying a reference to show obviousness. These rationales include: (a) combining prior art elements according to known methods to yield predictable results; (b) simple substitution of one known element for another to obtain predictable results; (c) use of a known technique to improve a similar device (method, or product) in the same way; (d) applying a known technique to a known device (method, or product) ready for improvement to yield predictable results; (e) choosing from a finite number of identified, predictable solutions, with a reasonable expectation of success; and (f) some teaching, suggestion, or motivation in the prior art that would have led a POSITA to modify the prior art or to combine prior art teachings to arrive at the claimed invention.

33. Also, I have been informed and understand that obviousness does not require physical combination/bodily incorporation, but rather consideration of what the combined teachings would have suggested to a POSITA at the time of the alleged invention.

V. BACKGROUND

34. I discuss in this section general background information about VoIP telephony. It is my opinion that the information I discuss in this “BACKGROUND” section would have been background knowledge to a POSITA.

35. Using the internet for voice communications dates back to the earliest days of the internet, and in fact, predates the internet. Researchers created voice-over-packet communications demonstrations on the research predecessor network predecessor to the internet, the ARPANET:

In the initial experiments on the ARPANET, the basic feasibility of speech communication on a store-and-forward packet network was demonstrated. Techniques were developed for reconstitution of speech from packets, and protocols were developed for call setup and for speech transport.... Key developments to date associated with the wideband experiments have been 1) techniques for internetting via voice/data gateways from a variety of local access networks (packet cable, packet radio, and circuit-switched) to a long-haul broadcast satellite network and 2) compact implementations of packet voice terminals with full protocol and voice capabilities.

Ex.1087, 963.

36. As early as the 1960's, there were multiple speech compression algorithms, which at the time were commonly called vocoders, short for "voice encoder." Ex.1084, 88. Early voice gateways supported multiple forms of voice encoding. For example, researchers described in 1982 a "Switched Telephone Network" Interface that supported "low-data-rate external vocoders in addition to the PCM [pulse coded modulation] codec on-board." Ex.1086, 16-17.

37. Over time, speech or audio compression algorithms came to be more commonly referred to as codecs (short for "coder-decoder"). By at least 1992, voice-over-internet terminals supported a variety of audio codecs, including some (like G.711 and G.723) that remained in use in the early 2000's. *See* Ex.1085, 3-4; Ex.1074, [0057]; Ex.1053, [0010]. Voice terminals also supported changing between codecs during a call (i.e., a "dynamic change in audio encoding"). Ex.1085, 5.

38. Like the internet, voice over internet protocol communications technology became commercialized over time. By the time the '368 patent application was filed in late 2005, there were many models of VoIP phones¹

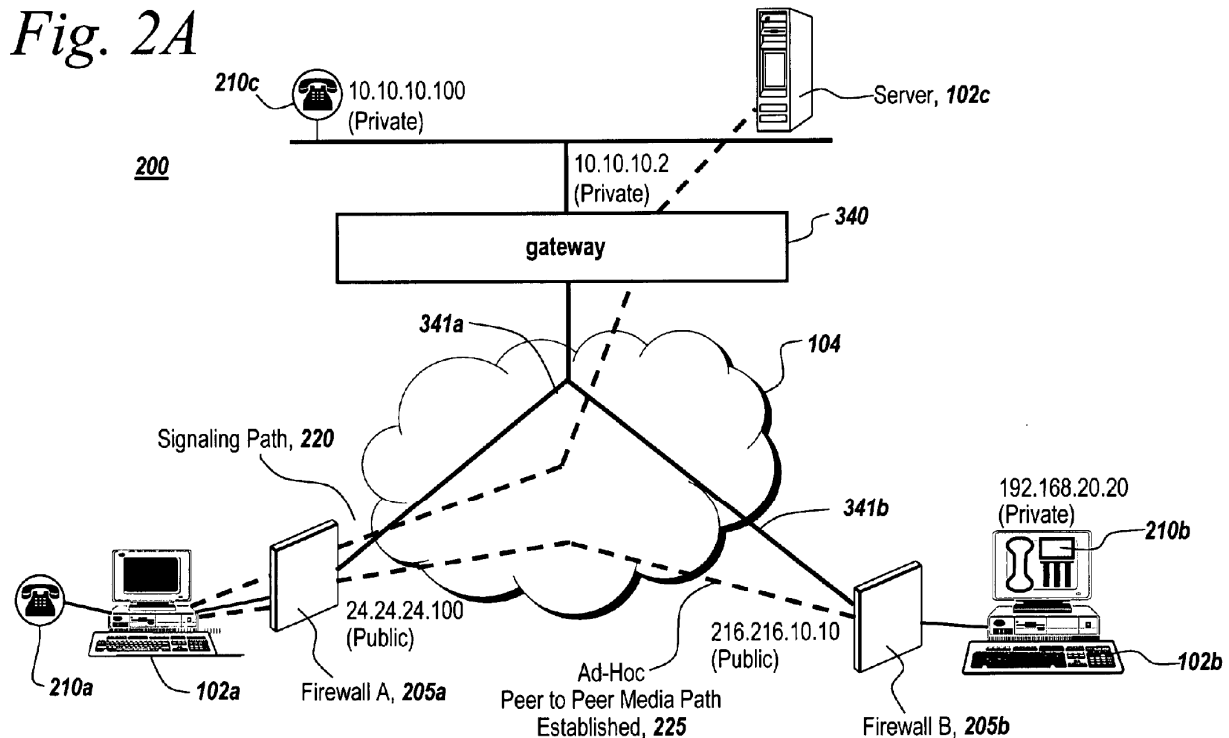
¹ VoIP phones were sometimes referred to by other terms, such as "IP phone," "digital VoIP telephone," or "IP telephony device." *See* Ex.1008, [0017]-[0020], Fig. 2 (referring to "digital VoIP telephone 20," "IP telephony device 20," and "IP phone 20"). These terms are largely synonymous and interchangeable.

commercially available. *See* Ex.1019, 142 (“Many makes and models of VoIP telephones are available.”). For example, companies such as Cisco Systems, Avaya, and Nortel Networks (among others) sold IP phones. *See* Ex.1041, [0021] (“a VoIP phone commercially available from Nortel”); Ex.1042, [0091] (“MGCP, SIP, and H.323 VoIP phones 950, VoIP client adapters 935 such as Cisco Systems, Carrier Access Corporation, Sylantro Systems or Zhone Technologies integrated access devices”); Ex.1005, 4:64-5:4 (“IP hardphones (such as the Avaya™ IP Hardphone), IP Softphones (such as the Avaya™ IP Softphone”); Ex.1054, [0003] (“...IP telephones such as those manufactured by Cisco, Siemens, Polycom, and Alti-IP...”).

39. VoIP phones were available as either wired or wireless models.

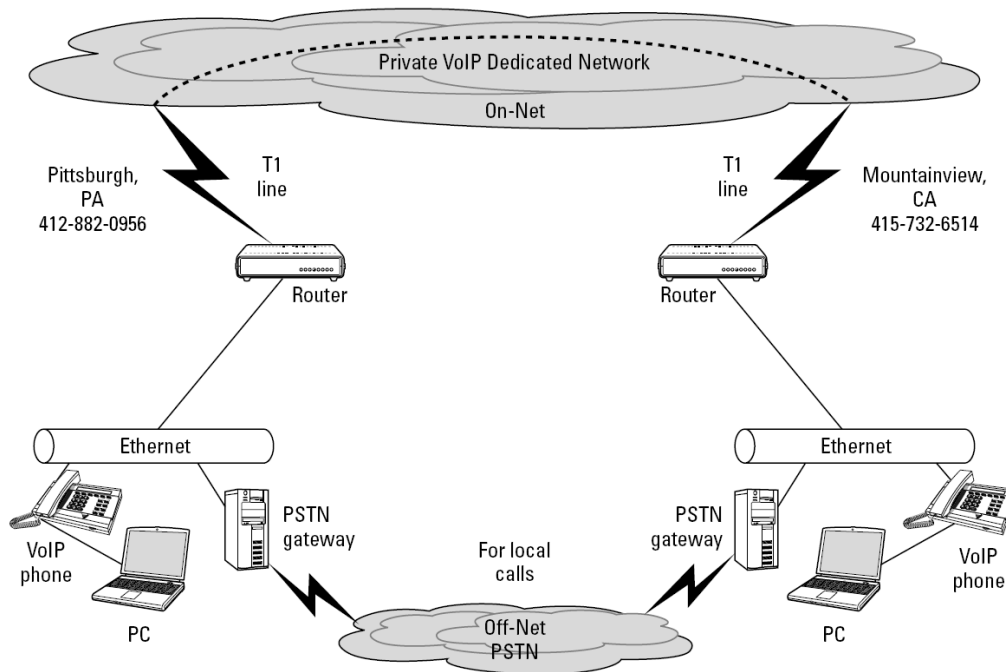
Ex.1019, 142. VoIP phones could also be generally grouped into two categories: (1) “hard” phones, which were physical devices that looked and operated much like a traditional desk telephone, and (2) “soft” phones, which were software for a personal computer that would allow the computer to make and receive voice calls. *See* Ex.1019, 142-149 (discussing hard and soft VoIP phones, including illustrating an example hard VoIP phone at Fig. 10-2 and a “soft phone running on a Windows PC” at Fig. 10-3); Ex.1005, 4:64-5:4. For example, a prior art patent application figure, shown below, illustrates a “telecommunication device 210a, such as a hard IP phone” and a “telecommunication device 210b, such as a soft IP phone.”

Ex.1043, [0140]. The hard IP phone 210a is illustrated as a typical phone icon (evoking the design of the classic “Model 2500” analog telephone), while the soft IP phone 210b is shown as software displayed on the monitor of a personal computer:



Ex.1043, Fig. 2A.

40. A VoIP hard phone would typically be designed to connect to an Ethernet network, and thus, would have an RJ-45 connection port. *See* Ex.1019, 142 (“The RJ-45 connector on a [VoIP] hard phone is an Ethernet port used to connect the phone to your network.”). For example, the figure below from “VoIP for Dummies” shows two VoIP hard phones, each connected to an Ethernet network:



Ex.1019, 27 (Fig. 2-3); *see also* 38 (Fig. 2-10), 74 (“VoIP runs on Ethernet LANs...”).

41. Numerous other prior art references further confirm that it was well-known for VoIP phones—including commercially available VoIP phones—to have an RJ-45 connection port for connecting to an Ethernet network, and that this was a well-known and typical arrangement:

- **“Many currently available IP telephones** such as those manufactured by Cisco, Siemens, Polycom, and Alti-IP **have an Ethernet jack, e.g. RJ45, to connect the IP telephone to the LAN connection,** and an additional RJ45 jack to connect the IP telephone and a computer to the corporate LAN connection.” Ex.1054, [0003].

- “With reference to FIG. 1 initially, shown is a configuration where a telephony server 10 is connected to a LAN with **IP phone sets 12 and 14 connected to the LAN** and computer 16. Also illustrated is a USB phone set 18 connected to a computer 20 in turn connected to the LAN. **This is a typical arrangement which is well identified in the art.**” Ex.1079, [0027].
- “In yet another example, **the VoIP communicator 10 may be connected to the POTS and VoIP networks via** separate hardware interfaces 110, 120 (e.g., **a RJ-45 interface 120 for the VoIP network** and a RJ-11 interface 110 for the POTS network). In this example, the VoIP communicator 10 may include additional **well-known technologies for connection to the VoIP network via Ethernet**, cable, high Speed modem (e.g., 56 K or higher), and/or other known techniques.” Ex.1044, [0018].
- “In the case of **Ethernet LAN connected telephones (often referred to as IP Phones)**, an international standard is being developed for providing power to such telephones from a central location via an additional “spare” pair of wires (the 7/8 pair of wires in **a standard RJ-45 jack configuration**). This powering arrangement is made possible by the fact that standard Ethernet LAN cables have four pairs of wires, of which only two are used by most common data signaling protocols (e.g. 10BaseT and 100BaseT).” Ex.1045, 1:20-28.

- “The IP telephony system, generally referenced 30, comprises, a **LAN infrastructure represented by an Ethernet switch 32**, a router, one or more telephones 36, workstations 34, a gateway 42, a gatekeeper 46, a PBX 33 with a LAN interface port and a Layer 3 switch 38. The key components of an IP telephony system 30 are the modified desktop, gatekeeper and gateway entities. For the desktop, users may have **an Ethernet phone 36 that plugs into an Ethernet RJ-45 jack....**” Ex.1046, 1:65-2:6, Fig. 2 (showing IP telephone 36 connected to Ethernet switch 32).
- “IP telephones and other IP telephony devices 52” connect to “LAN 50 [that] uses a communication protocol (such as Ethernet).” Ex.1047, Fig. 1, 4:62-63.

42. Ethernet networks were common, with Ethernet as “the most pervasive if not the dominant, network technology for LANs.” Ex.1088, [0005]; *see also* Ex.1010, 3 (“Ethernet has been the dominant technology in enterprise and campus LANs for many years.”). An Ethernet network would have one or more Ethernet switches, and each switch would have multiple RJ-45 connection ports for coupling to other network devices. *See* Ex.1048, Fig. 1A, 2:63-3:8 (“ethernet switch module 102” including “industry standard RJ-45 electrical connectors”). To be sure, other types of Ethernet network interfaces existed, such as the XENPAK standard used for Gigabit Ethernet connections using fiber optic cables. *See, e.g.*, Ex.1089, [0045].

But the dominant form of Ethernet used with local area networks and IP phones used RJ-45 connection ports.

43. It was common to connect multiple communications devices, including IP telephones, to a LAN, especially in the context of a corporate network. Many prior art patents depict such arrangements. For example, Fig. 1 of Ex.1083 “shows an **IP telephony system 100 of the prior art**. System 100 includes a business system 102 configured to provide IP telephony in an enterprise environment. Business system 102 may include **a network 104, such as a corporate Ethernet LAN, to which a plurality of IP telephones 106 may be operatively coupled to network 104 using hardware and software standard in the art.**”:

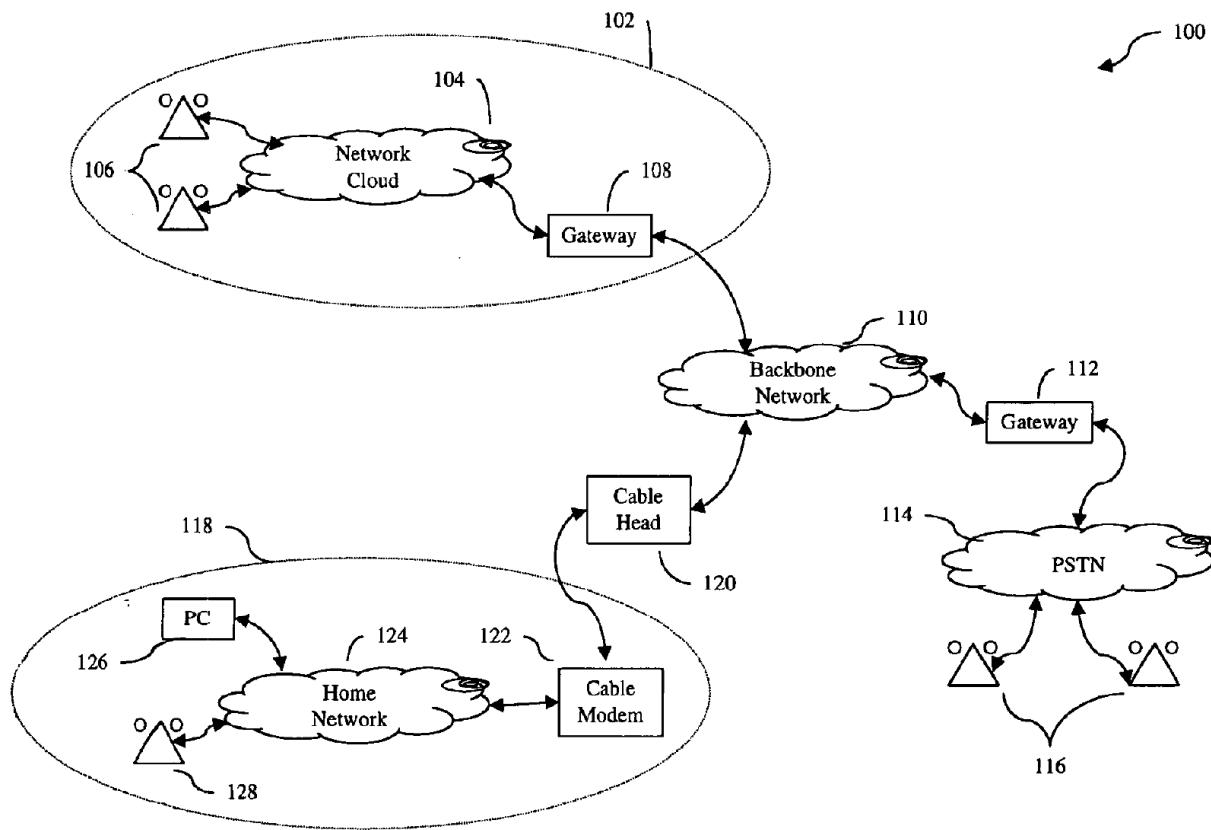


FIG. 1
Prior Art

Ex.1083, Fig.1.

44. In another example, Fig. 1 of Ex.1059 shows a “system 10 includ[ing] a **network 12, which is of a known type** commonly used to transfer digital information between computers. Information is sent through the network in the form of packets or message units, which each include digital data and control information. In the disclosed embodiment, the **network 12 is a local area network (LAN) which conforms to a standard known in the industry as the Ethernet standard....** The system 10 further includes a **plurality of telephones, six of**

which are shown at 21-26. Each of the telephones 21-26 is operationally coupled to the packet-switched network 12,...”:

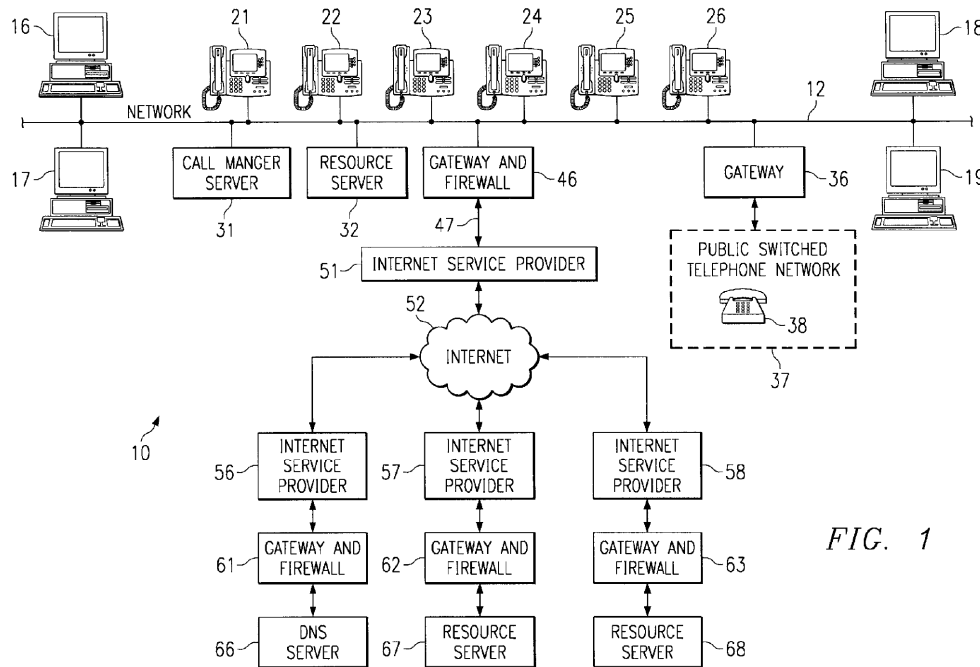


FIG. 1

Ex.1059, Fig. 1.

45. Connecting IP phones to an Ethernet switch allowed the switch to power the IP phones using “Power-over-Ethernet” technology. In the 1990’s there were various proprietary approaches to this, but the technology was subsequently standardized by the Institute of Electrical and Electronics Engineers as part of its suite of Ethernet standards, and more particularly, as IEEE 802.3af in 2003. See Ex.1068, [0004]-[0006] (“IEEE Std 802.3af... provides an integrated power and data network connection...”). Power-over-Ethernet provided a variety of benefits, including eliminating the need for an IP phone to have a local power source, such as an adjacent electrical outlet. See Ex.1068, [0005] (“Such an arrangement removes

the need for independent power and LAN connections”); [Ex.1069, [0041] (“This eliminates the need for client devices 12-16 to have separate power connections 18 to electrical outlets...”). With the IP phones on an Ethernet network powered by an Ethernet switch, backup power for emergency services during a power outage would be provided with a single backup power supply at the switch. *See* Ex.1065, Fig. 8, [0055] (“In an electrical outage... voice services are maintained for emergency situations”). Thus, Power-over-Ethernet “ultimately lowers the cost of attaching a device to the LAN.” Ex.1068, [0005]. Because of these benefits, it was common and desirable for IP phones to receive both power and data from an Ethernet switch. *See* Ex.1070, [0002] (“It is desirable in today's marketplace, given the building infrastructure, to provide power over ethernet cable....”).

46. An IP phone is an example of a VoIP endpoint, which refers to a device that terminates a voice over internet protocol session. *See* Ex.1049, [0007], [0014] & Fig. 2 (illustrating “VoIP endpoints” that “typically comprise IP phones”). A VoIP endpoint might not be the terminus of the voice call itself, however. For example, a VoIP endpoint could be a terminal adapter, which would provide “interworking functions” to allow connection to a VoIP-incompatible device, such as an ordinary “analog phone.” Ex.1049, [0014]. In other words, while not every VoIP endpoint is an IP phone, every IP phone is a VoIP endpoint. Numerous prior art documents confirm this understanding, including the following:

- “VoIP endpoints may include gateways, IP phones, software clients on a PC or server endpoints.” Ex.1050, [0025].
- “Examples of VoIP telephony endpoint device 24 include: VoIP telephones; VoIP handsets; hybrid computer/VoIP devices; and VoIP video phones, for example.” Ex.1051, [0022]; *see also id.* [0003] (“VoIP endpoint devices (e.g., telephones, handsets, videophones, etc.)”).

47. VoIP endpoints—including IP phones—implement one or more codecs to encoding and decoding audio information. As I discussed above, the use of codecs in VoIP endpoints calling dates to at least the 1980’s. *See* Ex.1085, 3-4. The use of one or more codecs by an IP phone was well-known, as evidenced by the following prior art descriptions:

- “FIG. 2 is a block diagram illustrating a stand-alone VoIP telephone apparatus 201. The **VoIP telephone apparatus 201 includes the following components:** an analog telephone interface adaptor 202 with an RJ11 connection 203 to an analog telephone extension; **an encoder and decoder (codec) module 204** responsive to a dedicated VoIP telephone handset 211 and responsive to the analog telephone interface adaptor 202; a time division multiplexer 205 incorporated into the encoder/decoder module 204; a digital signal processing circuit 206 responsive to the encoder/decoder module 204; a telephone keypad 207 responsive to caller input; a control processor 208; a

visual display window 209 responsive to an external analog telephone interface; a digital Internet protocol (IP) interface 214; and a RJ45 connection 215 to a data switch.” Ex.1054, [0025], Fig. 2.

- “**With IP phones**, voice signals input from the handset 208 are processed through the audio processing unit 207, **CODEC processing for audio processing is performed by a CODEC unit 243**, and voice signals are transmitted/received as digital data through encoding/decoding conforming to an encoding format such as ITU-T Recommendation G.711 or G.729, or the like.” Ex.1055, [0063].
- “IP Phone 114—**IP telephone with** digital processing capability to support **multiple codecs** and communications protocols, echo cancellation with an Ethernet interface.” Ex.1056, [0044].
- “Today, IP telephony is based on the use of a vocoder working at fixed bit rates usually defined in some recommendation e.g. ITU-T G.711 (48 kbps to 64 kbps) or G.723.1 with a low bit rate 5.3 to 6.3 kbps output quality, or even G.729 at 8 kbps using conjugate structure-algebraic code excited linear predictive methods (kpbs stands for kilo bit per second). **When multiple coding algorithms have been embedded inside the user equipment like the IP phone** or inside an IP gateway letting available for the respective vocoder, a codec (coder-decoder) negotiation phase is started during an

earlier phase of the building up of a telecommunications as defined e.g. in ITU-T H.323 recommendation. Capability exchanges are running between both ending points i.e. between the two equipment usually being the user equipment between which the telecommunications shall take place. The final choice of the algorithm to be used as encoding level is taken according to the common vocoder found in the capability lists at the respective end points.” Ex.1057, [0003].

- US2005/0207432 describes “codecs” as one of the “settings” that could be managed for an “IP phone 101. Ex.1052, [0071].

48. The quality of a VoIP call was commonly assessed based on measurements of packet delay, number of lost packets, and jitter. For example, prior art patent applications describe how “[t]he quality of a VoIP call is assesse[d] by monitoring the quality of the corresponding IP packet traffic using conventional IP packet quality measures such as, for example, **jitter, packet loss and packet delay.**” Ex.1072, Abstract. This was, in part, because it was known that “applications delivering voice packets are especially sensitive to delay, jitter, and packet loss.” Ex.1073, [0026]; *see also* [0041] (“suitable packet delay characteristics (as well as jitter and packet loss) prove to be of special importance in successful implementation of VoIP applications”). Older analog telephone systems had used a “Mean Opinion Score (MOS)” to evaluate call quality. Ex.1074, [0004].

A MOS score was a subjective evaluation of the voice call quality. Ex.1074, [0004]-[0005]. Engineers had developed technology for measuring “one-way delay, packet loss and jitter” and using these values to “generate an estimated subjective rating, such as a Mean Opinion Score (MOS).” Ex.1074, [0061]. Thus, even when other techniques for evaluating call quality were developed, measurements of packet delay, packet loss, and jitter remained important for evaluating VoIP call quality.

49. As I discussed above, IP phones would typically connect to an Ethernet network. It was (and is) common to illustrate a network as a cloud, which was intended to represent a variety of networking devices such as routers, switches, interconnecting cables, and sometimes servers, too. For example, in networking drawings, “network infrastructure is depicted in the conventional manner using network clouds.” Ex.1062, [0054]. A network cloud could “represent[] any number of network elements such as switches, enhanced platforms, and servers.” Ex.1060, [0040]. As another example, a “‘data network cloud’ may refer to one or more communications networks, channels, links, or paths, and systems or devices (such as routers or switches)....” Ex.1061, 4:5-11.

50. Network diagrams often illustrate communication links between components as a line. It was understood, however, that the connected components would each have an interface for sending data, receiving data, or (most commonly)

for both. While the specific type of interface would vary depending on the attributes of communication link (e.g., whether wired or wireless, optical or electrical, standards-based or proprietary), there would always be an interface of some kind to facilitate the component's connection to the communication link. As one patent application explained, "Each system attached to the network cloud 102 includes an **appropriate conventional network interface** arrangement (not shown) for communicating over the network 102, or through the switches 106 and 108." Ex.1058, [0026]. In the context of IP phones, for example, it was generally known that a "telephone 21 also includes a network interface circuit 142, which interfaces... to the Ethernet network." Ex.1059, 5:56-59; *see also* Ex.1056, [0044] (defining "IP Phone" as requiring "an Ethernet interface.").

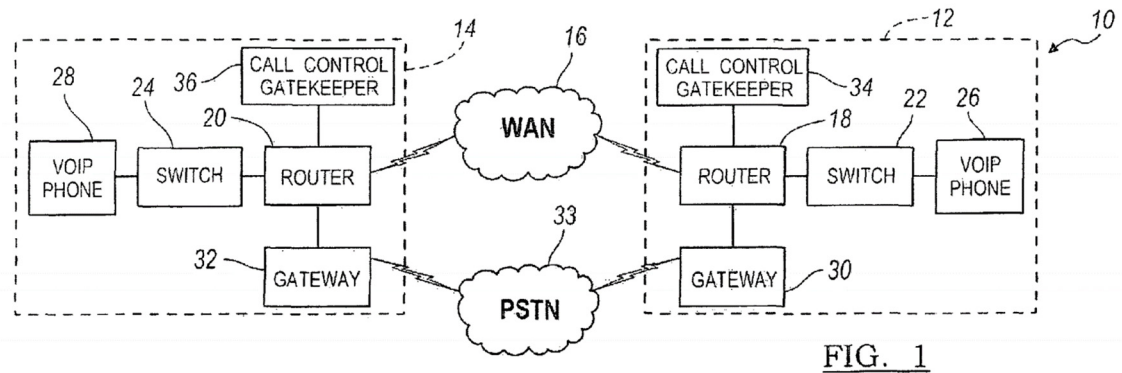
VI. THE '368 PATENT

A. Summary of the '368 Patent

51. The '368 patent "generally relates to voice over internet protocol ('VoIP') telephone systems." Ex.1001, 1:6-8, Abstract.

52. The '368 patent discloses a system for changing the coder-decoder ("codec") of a VoIP telephone call to a lower bandwidth codec. With reference to Figure 1 shown below, the '368 patent describes a system that includes a wide area network ("WAN") 16 connected to two local area networks ("LAN") 10, 14, with each LAN including a call control gatekeeper 34, 36, a router 18, 20, a switch 22,

24, and a VoIP phone 26, 28. The system also includes a Public Switched Telephone Network (“PSTN”) 33 connected to two gateways 30, 32. Ex.1001, 2:41-67.



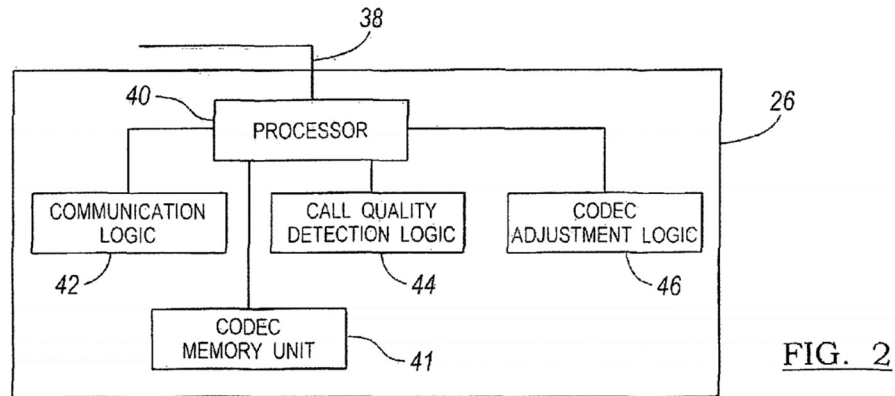
Ex.1001, FIG. 1.

53. The '368 patent provides an embodiment for adjusting the codecs during a VoIP call where a gatekeeper receives call quality information from the VoIP telephones and if the call quality is low instructs the phones to renegotiate to a lower bandwidth codec:

Alternatively, the gatekeepers 34 and 36 may be configured to receive a call quality signal from their respective VoIP telephone 26 and 28. When one of the call control gatekeepers 34 and 36 receive a call quality signal indicating that voice packets are being dropped, the call control gatekeeper 34 or 36 will instruct the VoIP telephone 26 and 28 to renegotiate with the other VoIP telephone 26 or 28 to renegotiate the telephone call and select a codec with a lower sampling rate.

Ex.1001, 3:58-65; *see also* Ex.1001, 4:47-55.

54. The '368 patent's VoIP telephone 26 is illustrated below at Figure 2:



Ex.1001, FIG. 2.

55. In the above figure, the VoIP telephone 26 includes an interface 38 for connecting to the network, a processor 40, a codec unit 41, and various logic components 42, 44, and 46. Ex.1001, 3:66-4:28.

56. Challenged Claim 1 of the '368 patent is reproduced below:

1. A system comprising:
 - a VoIP service provider system having a call control gatekeeper, a first interface and a second interface;
 - the first interface connected to a first LAN having an interface to a first VoIP telephone;
 - the second interface connected to a second LAN having an interface to a second VoIP telephone;
 - the first and second VoIP telephones being configured to communicate using a first codec;
 - the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone and send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec; and
 - wherein the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones during a telephone call between the first and second VoIP telephones.

B. Prosecution History

57. In a first Office Action, the Examiner rejected independent claims 1, 8, and 15 in view of Kalliokulju et al., US 2003/0063569, and the remaining claims 2-7, 9-14, and 16-18, were indicated as allowable if rewritten in independent form. Ex.1002, 52-65. In the subsequent Response, Applicants canceled rejected independent claims and amended the dependent claims to be written in independent form. Ex.1002, 66-77. Dependent claim 2 (issued as independent claim 1) was amended as follows:

2. (Currently Amended) ~~The system of claim 1,~~ A system comprising:
 a VoIP service provider system having a call control gatekeeper, a first
 interface and a second interface;
 the first interface connected to a first LAN having an interface to a first VoIP
 telephone;
 the second interface connected to a second LAN having an interface to a
 second VoIP telephone;
 the first and second VoIP telephones being configured to communicate using
 a first codec;
 the call control gatekeeper configured to receive a call quality signal from the
 first VoIP telephone and send an adjustment signal based on the call quality
 signal to the first and second VoIP telephones instructing the first and second
 VoIP telephones to communicate using a second codec; and
 wherein the adjustment signal based on the call quality signal is sent to the
 first and second VoIP telephones during a telephone call between the first and
 second VoIP telephones.

58. In a subsequent Office Action, the Examiner indicated that claims 2-7 and 9-14 were allowed and rejected pending 16-18 claims as anticipated by Kakimoto, JP2005-260649. Ex.1002, 90-93. The Applicants, in response, canceled rejected claims 16-18, and the Examiner issued a Notice of Allowance, allowing pending claims 2-7 and 9-14 to issue as claims 1-12. Ex.1002, 102-110, 117-120.

59. Notably, as I demonstrate below, there is nothing novel about claim 1 of the '368 patent since the element that which the Examiner identified as allowable was known in the art and it would have been obvious to combine the prior art as claimed.

VII. CLAIM CONSTRUCTION

60. It is my understanding that in order to properly evaluate the '368 patent, the terms of the claims must first be interpreted.

61. It is my understanding that for purposes of *Inter Partes* Review, the claim terms must be construed according to their ordinary and customary meaning as would have been understood by one of ordinary skill in the art. I have also been informed that claim terms only need to be construed to the extent necessary to resolve the obviousness inquiry. It is my opinion that for purposes of applying the prior art presented herein to evaluate the patentability of the claims no term requires express construction.

VIII. DETAILED UNPATENTABILITY ANALYSIS

62. I have been asked to provide my opinion as to whether the Challenged Claims of the '368 patent would have been obvious in view of the prior art. The discussion below provides a detailed analysis of how the prior art references I reviewed teach the limitations of the Challenged Claims of the '368 patent.

63. As part of my analysis, I have considered the scope and content of the prior art and any potential differences between the claimed subject matter and the prior art. I conducted my analysis as of the earliest claimed priority date of the '368 patent: November 10, 2005. I have also considered the level of ordinary skill in the pertinent art as of that date.

64. I describe in detail below the scope and content of the prior art, as well as any differences between the claimed subject matter and the prior art, on an element-by-element basis for claims 1 and 2 of the '368 patent. This analysis supports my opinion that the differences between the Challenged Claims and the prior art discussed herein are such that the subject matter as a whole would have been obvious at the time of the filing of the '368 patent to a person having ordinary skill in the art to which the subject matter pertains. I note that my analysis and proposed prior art combinations rely on the teachings of the references and not on physical incorporation of the elements.

65. Additionally, as part of my analysis, I have reviewed and appropriately cite to certain “Request for Comments” (RFC) documents published by the Internet Engineering Task Force (IETF), in particular, RFC3435 and RFC3660. RFCs were well-known at the priority date of the ’368 patent, and it is my opinion that they were “printed publications” for the purposes for which they are cited throughout. *See, e.g.*, Ex.1098, p. 6, (“RFCs can be obtained from a number of Internet hosts...”), p. 8 (IETF documents are “readily available to a wide audience”), p. 26 (IETF participants “shall publicly announce...every activity” relating to the standardization process). Many other prior art patents cited to RFC3435 and RFC3660. *See, e.g.*, Ex.1033, [0005]; Ex.1034, [0039]; Ex.1035, [0039]; Ex.1036, [0016]; Ex.1037, [0056]; Ex.1039, 4:49-53 (all citing RFC3435); *see also* Ex.1038, 6:38-41; Ex.1039, 4:49-53 (citing RFC3660).

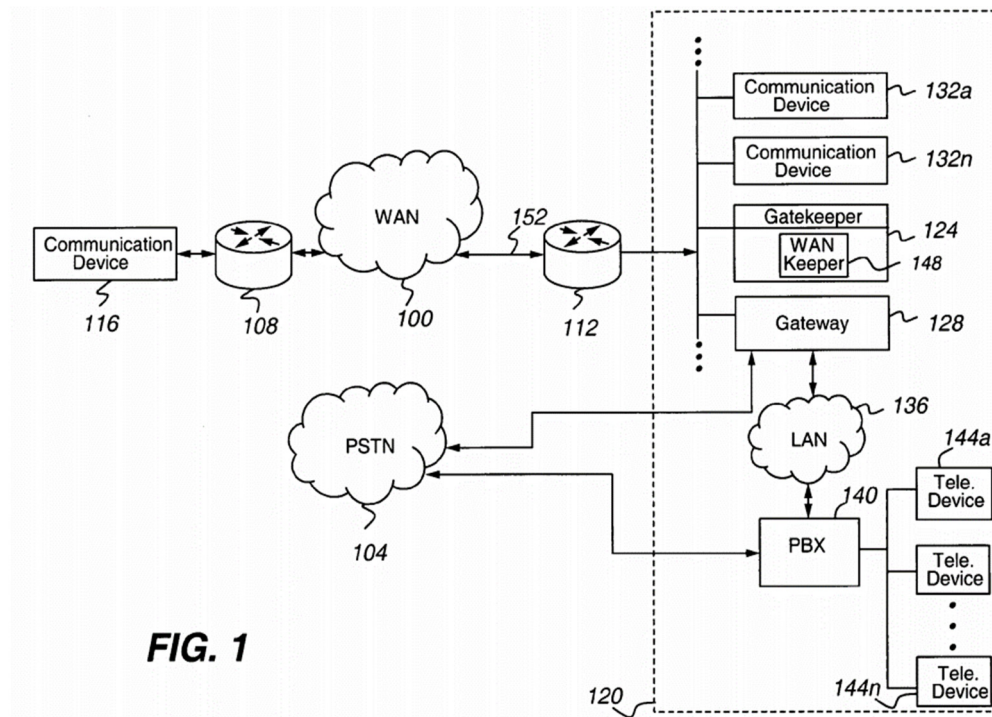
66. I will now describe, in the grounds below, on an element-by-element basis, how the prior art teaches all elements of claims 1 and 2. Unless otherwise noted, all emphasis in any quoted material has been added.

A. Ground 1: Claims 1 and 2 are obvious over Minhazuddin, Hepworth, and Shaffer

67. The combination of Minhazuddin, Hepworth, and Shaffer renders obvious claim 1 as discussed below.

1. Summary of Minhazuddin

68. Like the '368 patent, U.S. Patent No. 7,643,414 to Minhazuddin ("Minhazuddin," Ex.1005) generally discloses a VoIP system. Ex.1005, 1:5-8, 3:16-65, Abstract. Minhazuddin's exemplary system is illustrated at Figure 1, reproduced below:



Ex.1005, FIG. 1.

Referring to FIG. 1, a simplified communications network is depicted. The network comprises a packet-switched Wide Area Network or WAN 100, a circuit-switched network 104, local edge routers 108 and 112, packet-switched communication device 116, and enterprise network 120. The enterprise network 120 includes a gatekeeper 124, a gateway 128, packet switched communication devices 132a-n, a Local

Area Network or LAN 136, a Private Branch Exchange or PBX 140, and telecommunication devices 144*a-n*.

Ex.1005, 4:36-46, FIG. 1.

The gatekeeper 124 includes a WAN keeper 148 to perform dynamic call admission control as described in detail below. A preferred gatekeeper is a modified form of the Avaya™ IP600, S8700, S8300, S8500, or S8100 media servers.

Ex.1005, 5:7-11.

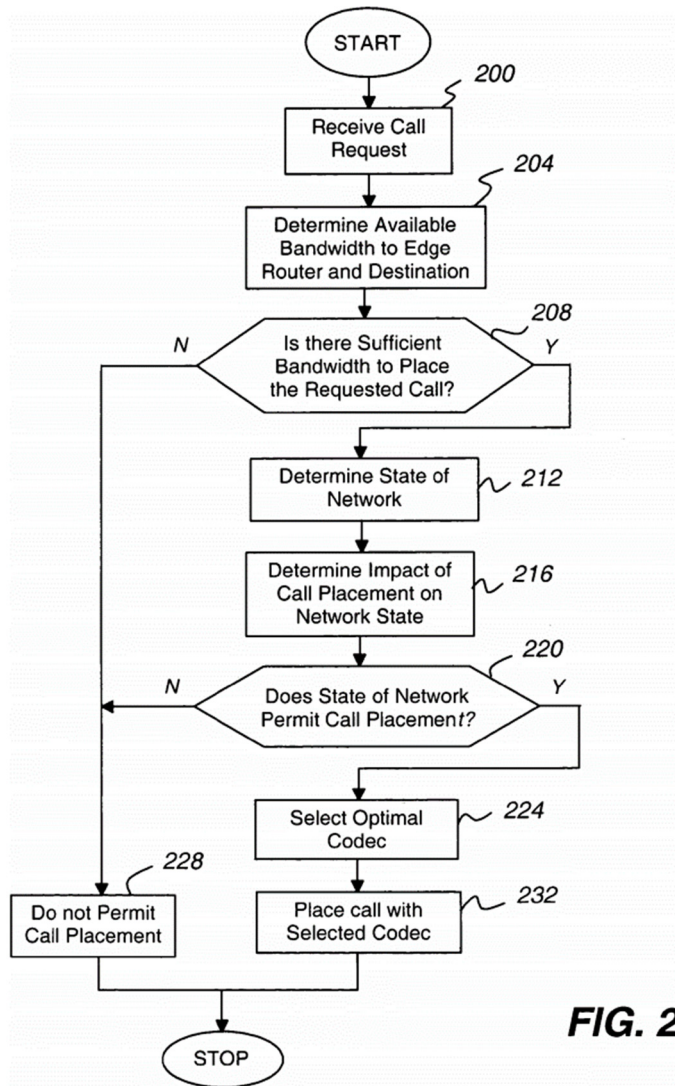
The WAN keeper 148 performs dynamic call admission control by determining, based on measured QoS metrics and/ or available (unused) bandwidth in the link 152 extending from the local edge router 112 to the next (nonlocal) router (not shown) in the first WAN hop and/ or in the various links between the gatekeeper 124 and the destination communication device 116, and whether or not the call can be placed using any codec....The WAN keeper 148 monitors the end-to-end QoS characteristics or metrics link(s) (or network state) by any suitable technique including receiving feedback signals from one or more response-time-response probes, by polling selected network components, using ICMP ping packets, and/ or SNMP/ RTCP packets.

Ex.1005, 5:40-58.

69. Minhazuddin's communication devices 116, 123*a-n* initially communicate using a first selected codec:

If a codec is identified, the WAN keeper proceeds to step 224 and selects that codec as the codec to be used in negotiating with the other endpoint.

Ex.1005, 7:52-63; Fig. 2.



Ex.1005, FIG. 2.

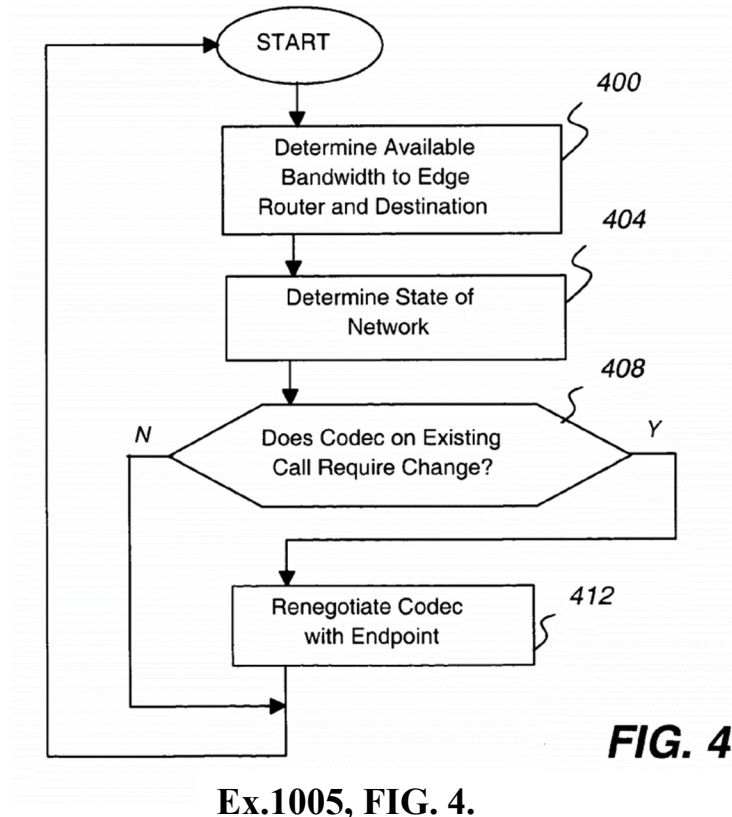
70. Minhazuddin discloses that gatekeeper 124 includes WAN keeper 148 that monitors QoS information (e.g., packet delay or latency, jitter, packet loss, etc.)

to determine if the first codec should be changed (via renegotiation) to a second codec that uses less bandwidth:

When the utilized bandwidth is above the maximum threshold or the unused bandwidth is below the maximum threshold, the WAN keeper 148 institutes one or more measures to control bandwidth utilization. In another configuration, when one or more QoS metrics exceeds selected thresholds...the bandwidth control measures are also implemented.

These bandwidth control measures are implemented in step 412. In one measure, the codecs used on existing VoIP calls are renegotiated to a lower bit rate codec (that will use less bandwidth)... In one technique, during call setup multiple possible codecs are negotiated between the endpoints, with the highest bit rate codec being used initially in the call. In the event bandwidth control measures require implementation, a lower bit rate codec (that was previously negotiated) will thereafter be used between the endpoints until bandwidth utilization level(s) and/or the QoS metric drops below selected threshold(s).

Ex.1005, 8:23-52; 3:55-57 (“The QoS metrics can be any suitable metric and provide a measure of the state of the network. Metrics include packet delay or latency, jitter, packet loss, RSVP status and the like.”).



2. Summary of Hepworth

71. U.S. Publication No. 2004/0073690 to Hepworth et al. (Ex.1006, “Hepworth”) also relates to VoIP communications over data networks. Ex.1006, [0002]-[0003].

72. Hepworth, in the context of Figure 1, discloses router interfaces for connecting to a network (e.g., LAN) and VoIP endpoints:

FIG. 1 depicts a VoIP architecture 100 according to an embodiment of the present invention. The architecture includes a first, second, third, and fourth end points 104, 108, 112, and 116, a router 120 having associated interfaces 124a-c, and switch or server 128. The various

components are interconnected by first, second, and third subnets 132, 136, and 140, respectively, which collectively form a local area network. For illustration purposes, the first and third endpoints 104 and 112 are configured as IP hard-phones, while the second and third endpoints 108 and 116 are configured as IP soft-phones. As will be appreciated, the various end points can be any communication device, such as a plain old telephone, a soft phone, an IP hard phone, a personal or laptop computer, a PDA, a pager, and a wireless phone. It is further to be understood that the architecture can include any number of subnetworks and routers, depending on the application, and that the subnets can form a network topology other than a LAN, such as a wide-area network, an enterprise network, and a metropolitan-area network.

Ex.1006, [0036]; *see also* Ex.1006, [0045].

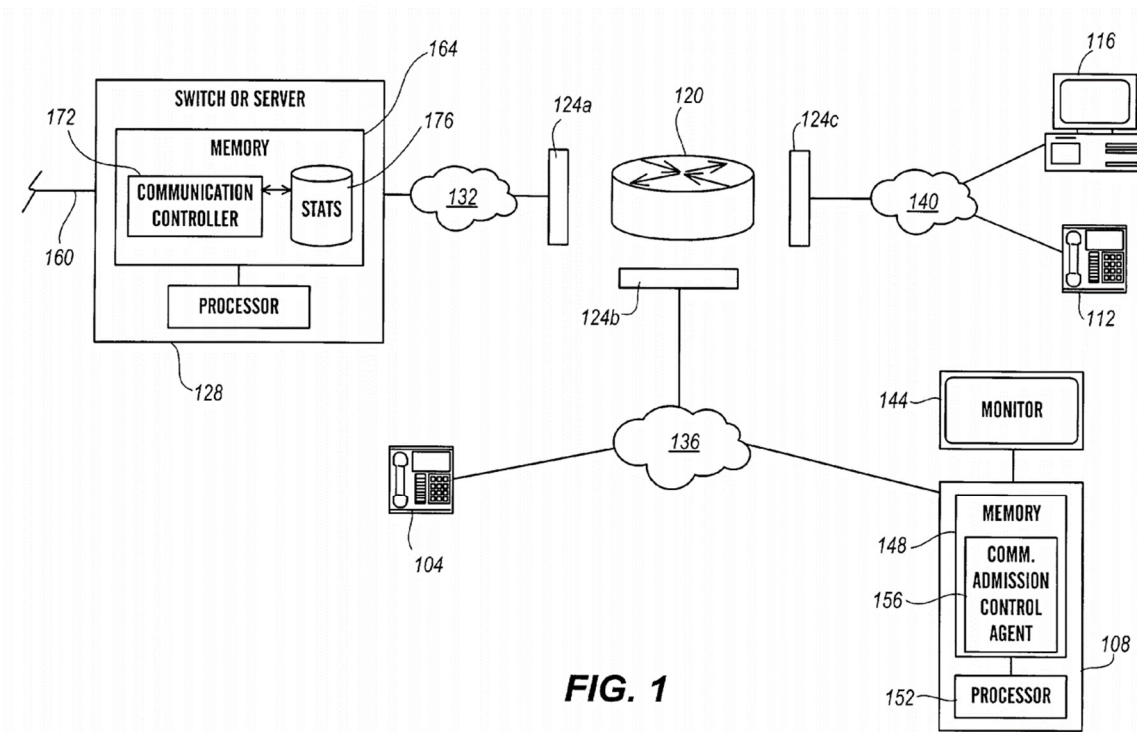


FIG. 1

Ex.1006, FIG. 1.

73. Hepworth further discloses that each VoIP telephone endpoint measures and collects QoS information (e.g., jitter, packet loss, round-trip time, number of dropped packets, bandwidth utilization, etc.) of a VoIP telephone call. Ex.1006, [0006] (“...key statistics about the media packets being sent and received by each end point such as jitter, packet loss, round-trip time, etc.) are collected by the end points.”), [0021] (“The bandwidth information can include...round trip time, jitter buffer delay, jitter, packet loss...a number of dropped packets by a router, router bandwidth utilization...”), [0044] (“[T]est RTP/RTCP packets can be sent between the two endpoints to measure one or more of the bandwidth information noted above, such as jitter, packet delay, and packet loss.”). Based on the collected QoS information, the VoIP telephone endpoints perform call admission control in a decentralized manner. Ex.1006, [0010], [0041].

74. Although Hepworth generally focuses on decentralized call admission control, Hepworth also contemplates implementation of conventional network architectures that utilize a centralized session monitor (e.g., Avaya VoIP Monitoring ManagerTM) that receives a signal with QoS information (e.g., jitter, packet loss, number of dropped packets, round-trip time, etc.) from VoIP endpoints to perform call admission control:

Media packets are transmitted between endpoints during a session according to RTP while additional performance information governing the communication link (e.g., key statistics about the media packets

being sent and received by each end point such as jitter, packet loss, round-trip time, etc.) are collected by the end points and transmitted to a session monitor according to RTCP. The network monitor can be, for example, VoIP Monitoring Manager™ or VMon™ by Avaya, Inc.

Ex.1006, [0006].

In another alternative embodiment, the switch or server 128 uses one or more of the endpoints (whether or not involved in a session) as network nodes to collect periodically any of the bandwidth information noted above. This embodiment can be used with the network architecture of the present invention or a conventional network architecture.

Ex.1006, [0053].

3. Summary of Shaffer

75. U.S. Patent No. 7,023,839 to Shaffer et al. (Ex.1007, “Shaffer”) generally pertains to a system that facilitates communication of real-time voice data over packet-switched networks (e.g., Internet and LAN). Ex.1007, 1:15-18, 3:14-27.

76. Shaffer discloses using a network interface that connects a telephony device to a LAN to facilitate telephony-over-LAN communications:

A logical diagram of an H.323 interface to LAN 101 is shown in FIG. 2, according to an embodiment of the present invention. The interface includes a known network terminal/device 10 utilizing the ITU-T H.323 protocol, and a packet network interface 13 that is coupled to network terminal 10. Network interface 13 couples the H.323 device to LAN 101. H.323 terminals/devices and equipment carry real-time voice, video and/or data. It should be noted that H.323 is an umbrella

recommendation that sets standards for multimedia communications, including telephony-over-LAN communications. The network can include packet-switched Transmission Control Protocol/Internet Protocol (TCP/IP) and Internet Packet Exchange (IPX) over Ethernet, Fast Ethernet and Token Ring networks.

Ex.1007, 3:14-27, FIGS. 1, 2.

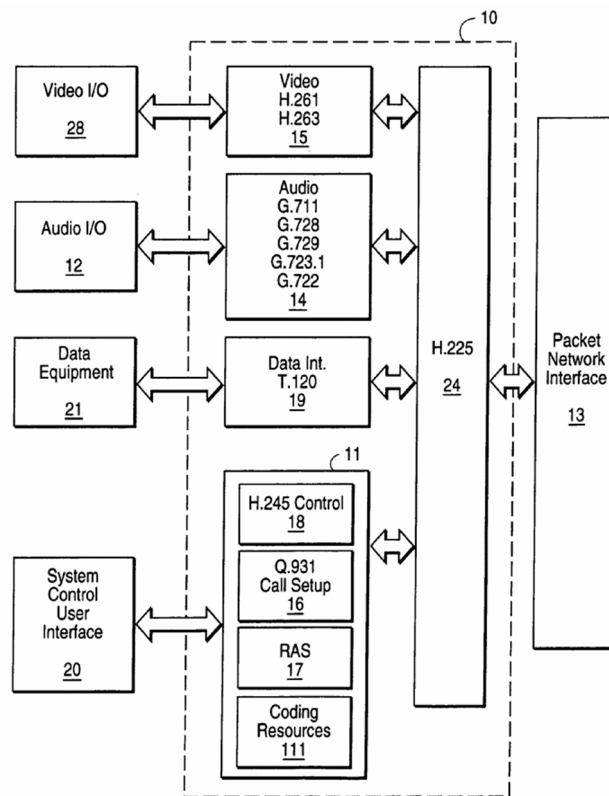


FIG. 2

Ex.1007, FIG. 2.

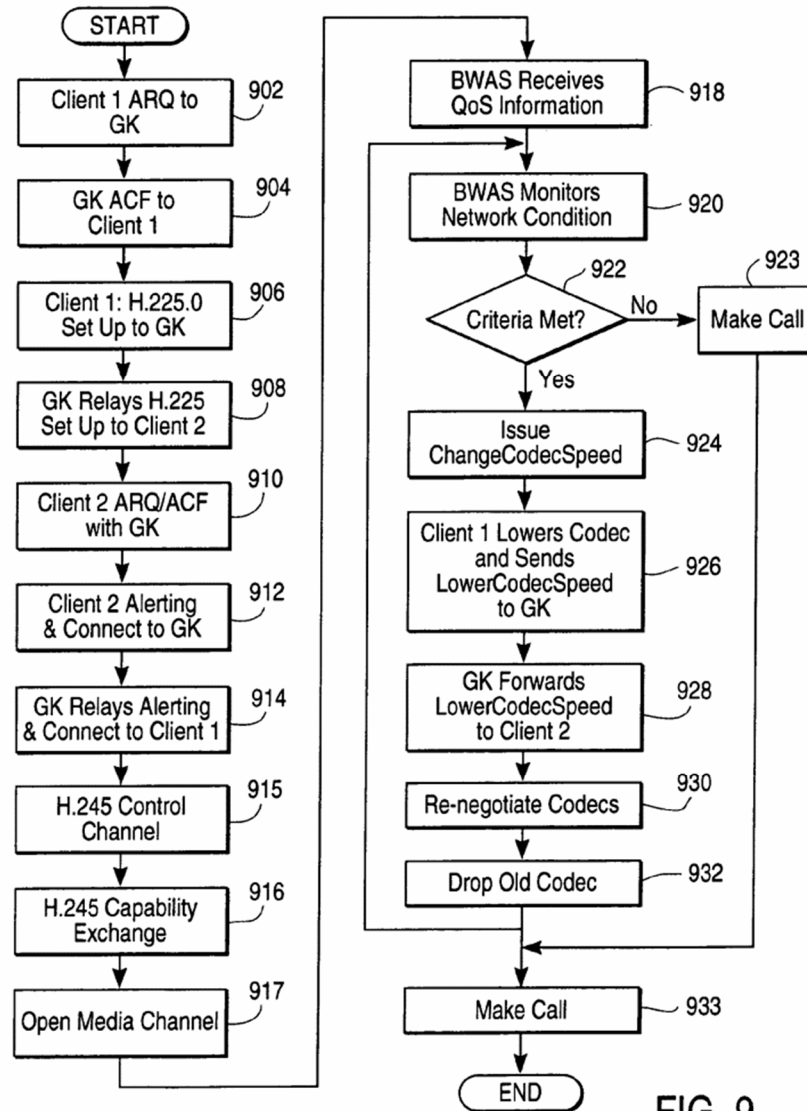
77. Shaffer further teaches that a gatekeeper includes a bandwidth allocation server (“BWAS”) that monitors the network and if a threshold is met, it sends a signal to adjust codecs to use less bandwidth:

A bandwidth adjustment server or bandwidth allocation server (BWAS) is provided which monitors system bandwidth usage, sends requests to user terminals to identify their coding capabilities, and directs each of the user terminals to adjust their coding algorithms based on system bandwidth usage. If system bandwidth usage is high, the BWAS requires the user terminals to employ a less bandwidth-intense coding algorithm; similarly, when system bandwidth usage is low, the BWAS will allow the user terminals to employ higher bandwidth-use coding algorithms.

The BWAS is configured with a first threshold identified as the threshold for reducing the coder/decoder (codec) speeds of the idle endpoints. The BWAS monitors system traffic, or communicates with other system monitors to determine system bandwidth usage. The BWAS sends a message to the user terminals, requiring them to identify their coding capabilities and the specific hierarchy used by them. Once this information is returned to the BWAS, **the BWAS sends another message requiring the user terminals to lower their bandwidth usage by selecting a lower speed codec.**

Ex.1007, 1:51-2:5.

78. Notably, Shaffer teaches the reason for allowance since the BWAS signal initiates “**renegotiation of codec usage while calls are ongoing.**” Ex.1007, 8:49-50; *see also* Ex.1007, 3:4-12, 8:49-67, 9:43-46, 10:17-36, 11:20-42, 12:7-47, FIGS. 4-10.



Ex.1007, FIG. 9.

FIG. 9

4. Reasons to Combine Hepworth with Minhazuddin

a. Hepworth is Analogous Art

79. Hepworth and Minhazuddin both disclose VoIP communications and are therefore analogous art to the '368 patent which pertains to VoIP communications. Ex.1001, 1:6-8, Abstract; Ex.1005, 1:5-8, 3:16-65; Ex.1006, [0002]-[0003]. Thus, given the similar subject matter, a POSITA considering

Minhazuddin would have naturally considered Hepworth, which merely provides additional teachings directly applicable to Minhazuddin.

b. Motivation to combine with Hepworth

80. A POSITA would have been motivated to combine the teachings of Hepworth and Minhazuddin to produce numerous predictable and beneficial results.

i. An interface

81. Minhazuddin discloses that its edge router has an “interface.” Ex.1005, 8:6-9. Minhazuddin, however, provides limited interface details. Hepworth provides implementation details regarding a router’s interfaces as contemplated by Minhazuddin. For instance, Hepworth discloses that the router has different associated interfaces for connecting to a LAN and a WAN, among others. Ex.1006, [0036], FIG. 1. It would have been obvious to a POSITA to consider and apply the router interface details of Hepworth, when implementing Minhazuddin’s local edge router interface in order to achieve the results that Minhazuddin is already describing; namely, connecting the various edge routers to different networks, e.g., LAN, WAN. That is, it would have been obvious to a POSITA, in view of Hepworth, to utilize an interface with Minhazuddin’s local edge routers, to provide a means for connecting to the LAN and supporting communications across the WAN. Ex.1005, 4:55-63, 6:57-60, 9:33-41.

82. The combination of Hepworth with Minhazuddin merely represents a simple combination of known elements (e.g., an interface per Hepworth with Minhazuddin's local edge routers) to yield predictable results (e.g., facilitate communication across the networks).

ii. Receive a call quality signal

83. Minhazuddin discloses that WAN keeper 148 (part of gatekeeper 124) "monitors the end-to-end QoS characteristics or metrics link(s) (or network state) by any suitable technique," including "measured QoS metrics [such as] packet delay, jitter, packet loss," among others. Ex.1005, 5:53-64.

84. Hepworth teaches that a well-known suitable technique to monitor network QoS includes a session monitor receiving a signal with measured QoS information (e.g., jitter, packet loss, number of dropped packets, round-trip time, etc.) from VoIP endpoints that collect this information. Ex.1006, [0006], [0021], [0053]. It would have been obvious to a POSITA to consider and apply Hepworth's QoS information signaling details, when implementing Minhazuddin's system in order to achieve the results that Minhazuddin is already describing, i.e., monitoring network QoS by any suitable technique to determine if the currently used codec should change. Ex.1005, 5:53-66, 8:14-67.

85. Additionally, it would have been obvious to a POSITA for Minhazuddin's VoIP communication devices to measure and collect the QoS

information and send this information to the WAN keeper 148 (part of gatekeeper 124) because it would free up the WAN keeper 148 to perform other administrative call control tasks. Ex.1005, 5:40-64. As another benefit, because the QoS measurement is distributed to the communication devices, failure of one communication device would not significantly impact the monitoring of network QoS. Additionally, utilizing a more distributed approach to measuring network QoS would have been beneficial because the network QoS information could be cross-correlated and cross-referenced to identify problem areas on the network that need to be resolved. For example, if one VoIP communication device has a low QoS measurement and another VoIP communication device has a higher QoS measurement then the WAN keeper 148 would be able to cross-correlate the QoS information and instruct the VoIP communication devices to use a codec that has a bandwidth requirement that meets low QoS requirements.

86. The proposed combination merely represents the application of a known technique (e.g., sending a signal with measured network QoS information, per Hepworth, to Minhazuddin's WAN keeper 148 that monitors network QoS) to yield predictable and beneficial results (e.g., providing the WAN keeper 148 sufficient information to perform admission control and determine if a codec should change).

87. Also, the combination of Hepworth with Minhazuddin merely represents a simple combination of known elements (e.g., sending a signal with measured network QoS information, per Hepworth to Minhazuddin's WAN keeper 148) to yield predictable results (e.g., providing the WAN keeper 148 with network QoS information that it seeks to obtain using any suitable technique).

c. Reasonable expectation of success

88. I note that the results would have been predictable and there would have been a reasonable expectation of success in the combination since Minhazuddin and Hepworth address the same technology, as analyzed above in the analogous art section. Also, the results would have been predictable and there would have been a reasonable expectation of success in the combination since interfaces for LAN were standardized components well-known in the art. Ex.1010, 1 ("Ethernet interfaces are available on a plethora of data communication/telecommunication devices. Standards-compliant interfaces are available for 10/100/1000 Mbps and the standard for 10 Gbps Ethernet was ratified in the IEEE in 2002"); *see also* Ex.1014, 44 (disclosing standard interfaces); Ex.1020, 4:55-63 (disclosing that network devices "...conventionally include[] interfaces to external communication links...").

89. The reasonable expectation of success is also supported by the fact that Minhazuddin contemplates using any suitable technique to monitor network QoS

(Ex.1005, 5:53-58) and Hepworth describes such a suitable technique, including how the VoIP telephones (communication devices) measure, collect, and send the QoS information (Ex. [0006], [0044]). A POSITA would have possessed the skills required to make the proposed combination, including being able to implement the combined teachings of Hepworth and Minhazuddin in corresponding hardware and software.

5. Reasons to Combine Shaffer with Minhazuddin

a. Shaffer is Analogous Art

90. Shaffer discloses voice communication over packet-switched networks (e.g., Internet and LAN) and is therefore analogous art to the '368 patent which also communicates voice data over such packet-switched networks. Ex.1001, 1:6-8, Abstract; Ex.1007, 1:15-18, 3:14-27. Minhazuddin similarly addresses voice communication over packet-switched networks. Ex.1005, 4:64-5:3. Accordingly, a POSITA considering Minhazuddin would have found Shaffer's communication teachings relevant.

b. Motivation to combine with Shaffer

91. A POSITA would have been motivated to combine the teachings of Shaffer and Minhazuddin to produce numerous predictable and beneficial results.

i. a first LAN having an interface to a first VoIP telephone and a second LAN having an interface to a second VoIP telephone

92. Minhazuddin discloses packet-switched communication devices (e.g., 116, 123*a-n*), which may be implemented as Avaya IP telephones that connect to LAN networks for packetized live voice communications. Ex.1005, 3:62-65, 4:36-5:3. It would have been obvious to a POSITA for each communication device implemented as Avaya IP telephones would have a corresponding network interface that is suitable for LAN connections. *See* Ex.1019, 13 (“All IP phones have one important thing in common: a built-in network interface card (NIC), just like a computer uses. The NIC is critical for any network device because it provides the device with a physical address and a way to communicate over the network.”), 245 (“A VoIP-enabled telephone that has an RJ-45 LAN interface port to connect it to the Ethernet LAN.”).

93. Shaffer also discloses a packet-switched communication device (referred to as a network terminal/device 10), which may be a telephony device, that communicates “real-time voice” over LAN. Ex.1007, 3:14-27, Shaffer provides additional implementation details of how the telephony device connects to the LAN, by disclosing that a “packet network interface 13 [] is coupled to network terminal 10” and the LAN. Ex.1007, 3:14-27, 4:14-22, FIGS.1, 2.

94. It would have been obvious to a POSITA, in view of Shaffer, to utilize a network interface to provide a means for connecting each of Minhazuddin’s communication devices (e.g., 116, 123*a-n*) to the LAN, thereby facilitating

packetized live voice communications between the devices. Ex.1006, 7:26-33, 7:54-57; Ex.1019, 13 (“All IP phones have one important thing in common: a built-in network interface card (NIC), just like a computer uses. The NIC is critical for any network device because it provides the device with a physical address and a way to communicate over the network.”), 245 (“A VoIP-enabled telephone that has an RJ-45 LAN interface port to connect it to the Ethernet LAN.”).

95. The combination of Shaffer with Minhazuddin merely represents a simple combination of known elements (e.g., a LAN network interface per Shaffer with Minhazuddin’s communication devices and LANs) to yield predictable results (e.g., facilitate communication between the communication devices over LANs).

ii. Sending a signal to change codecs

96. Minhazuddin discloses that WAN keeper 148 (implemented in gatekeeper 124) monitors QoS metrics and if a threshold is met, “the codecs used on existing VoIP calls are renegotiated to a lower bit rate codec (that will use less bandwidth).” Ex.1005, 8:42-52, FIG. 4; *see also* Ex.1005, 8:23-41(“when one or more QoS metrics exceeds selected thresholds...the bandwidth control measures are also implemented”), Abstract (“...change an existing live voice communication from the first codec to the second codec...”). Minhazuddin provides limited disclosure as to how the communication devices are informed to renegotiate to a lower bit rate codec that uses less bandwidth. Shaffer provides additional details,

disclosing that BWAS (incorporated in a gatekeeper) monitors QoS and if a threshold is met, a signal is sent to the communicating endpoints instructing them to renegotiate codecs on existing calls to use less bandwidth. Ex.1007, 1:51-2:14, 3:4-12, 10:17-38, 11:20-42, FIG. 4.

97. It would have been obvious to a POSITA, in view of Shaffer, to implement Minhazuddin such that when QoS metrics reach a threshold, the WAN keeper 148 (implemented in gatekeeper 124) sends a signal to its communication devices to renegotiate codecs on existing calls to use less bandwidth, as Shaffer teaches. The signal would be beneficial because it would inform the communication devices that they need to renegotiate codecs on existing live communications, thereby achieving Minhazuddin's goal of renegotiating the codec to be used in the live voice communication to a lower bit codec. Ex.1005, 8:43-52, 12:31-34. The renegotiation would reduce the chances of network QoS issues such as voice delay, packet loss, among others. Ex.1005, 3:55-57. Moreover, because network bandwidth for existing communications has been reduced, new calls may be supported on the network, thereby "maximi[zing] the number of voice sessions carried...and provide substantial increases in trunking efficiency." Ex.1005, 4:13-16.

98. The proposed combination merely represents the application of a known technique (e.g., sending a signal per Shaffer to a Minhazuddin's

communication devices) to yield predictable and beneficial results (e.g., informing the communication devices to renegotiate to a lower bit codec that uses less bandwidth, thereby freeing up bandwidth for new calls and maximizing the number of voice sessions carried).

99. Also, the combination of Shaffer with Minhazuddin merely represents a simple combination of known elements (e.g., sending a signal per Shaffer to Minhazuddin's communication devices) to yield predictable results (e.g., informing the communication devices to renegotiate to a lower bit codec that uses less bandwidth).

c. Reasonable expectation of success

100. I note that the results would have been predictable and there would have been a reasonable expectation of success in the combination since network interfaces were standardized components well-known in the art. Ex.1010, 1 (“Ethernet interfaces are available on a plethora of data communication/telecommunication devices. Standards-compliant interfaces are available for 10/100/1000 Mbps and the standard for 10 Gbps Ethernet was ratified in the IEEE in 2002”); *see also* Ex.1014, 44 (disclosing standard interfaces); Ex.1020, 4:55-56 (disclosing that network devices “...conventionally include[] interfaces to external communication links...”).

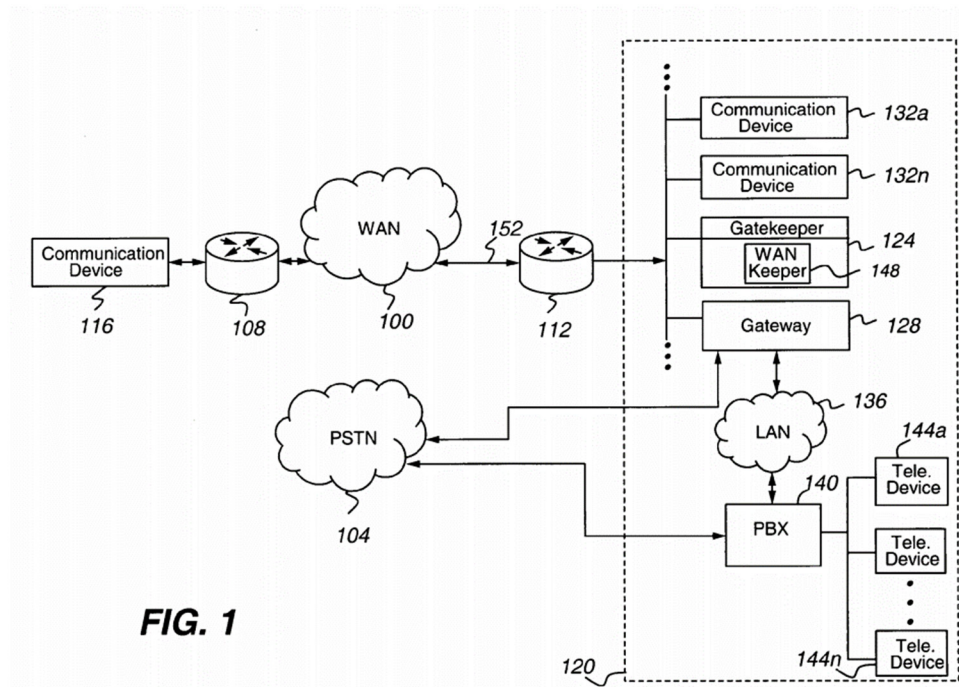
101. Also, there would have been a reasonable expectation of success in sending a signal. Shaffer describes that BWAS (incorporated in a gatekeeper) sends the signal to renegotiate codecs and Minhazuddin already uses a gatekeeper and renegotiates codecs. Shaffer simply provides additional implementation details with respect to how to renegotiate codecs, including how the gatekeeper informs communication devices to renegotiate codecs, as analyzed above. A POSITA would have possessed the skills required to make the proposed combination, including being able to send a signal utilizing corresponding hardware and software.

6. Claim 1

a. [1.0] *A system comprising:*

102. To the extent limiting, Minhazuddin renders obvious a system as recited in the preamble.

103. Specifically, Minhazuddin, in the context of Figure 1, reproduced below, discloses “methods, processes, **systems** and/or apparatus.” Ex.1005, 10:9-12.



Ex.1005, FIG. 1.

104. The system of Figure 1 has the following architecture:

Referring to FIG. 1, a simplified communications network is depicted. The network comprises a packet-switched Wide Area Network or WAN 100, a circuit-switched network 104, local edge routers 108 and 112, packet-switched communication device 116, and enterprise network 120. The enterprise network 120 includes a gatekeeper 124, a gateway 128, packet switched communication devices 132a-n, a Local Area Network or LAN 136, a Private Branch Exchange or PBX 140, and telecommunication devices 144a-n.

Ex.1005, 4:36-46, FIG. 1.

105. Thus, Minhazuddin's system renders obvious "[a] system," as recited in the preamble.

b. **[1.1] a VoIP service provider system having a call control gatekeeper, a first interface and a second interface;**

106. Minhazuddin alone and in combination with Hepworth renders obvious a VoIP service provider system having a call control gatekeeper, a first interface and a second interface.

107. First, Minhazuddin discloses a VoIP system that provides “**call admission control ...[for] VoIP.**” Ex.1005, 3:58-65. Minhazuddin’s VoIP service provider system comprises a gatekeeper 124, a first local edge router 108, and a second local edge router 112:

Referring to FIG. 1, a simplified communications network is depicted. The network comprises a packet-switched Wide Area Network or WAN 100, a circuit-switched network 104, **local edge routers 108 and 112**, packet-switched communication device 116, and enterprise network 120. The enterprise network 120 includes a **gatekeeper 124**, a gateway 128, packet switched communication devices 132a-n, a Local Area Network or LAN 136, a Private Branch Exchange or PBX 140, and telecommunication devices 144a-n.

Ex.1005, 4:36-46, FIG. 1.

108. The gatekeeper 124 includes a WAN keeper 148 that performs dynamic call admission control:

The gatekeeper 124 provides a variety of functions, including call set-up, call control, authentication, authorization, accounting, and address mapping or lookup, translation and resolution services. **The**

gatekeeper 124 includes a WAN keeper 148 to perform dynamic call admission control as described in detail below. A preferred gatekeeper is a modified form of the Avaya™ IP600, S8700, S8300, S8500, or S8100 media servers.

Ex.1005, 5:4-11.

The **WAN keeper 148 performs dynamic call admission control** by determining, based on measured QoS metrics and/ or available (unused) bandwidth in the link 152 extending from the local edge router 112 to the next (nonlocal) router (not shown) in the first WAN hop and/or in the various links between the gatekeeper 124 and the destination communication device 116, and whether or not the call can be placed using any codec....The WAN keeper 148 monitors the end-to-end QoS characteristics or metrics link(s) (or network state) by any suitable technique including receiving feedback signals from one or more response-time-response probes, by polling selected network components, using ICMP ping packets, and/or SNMP/ RTCP packets.

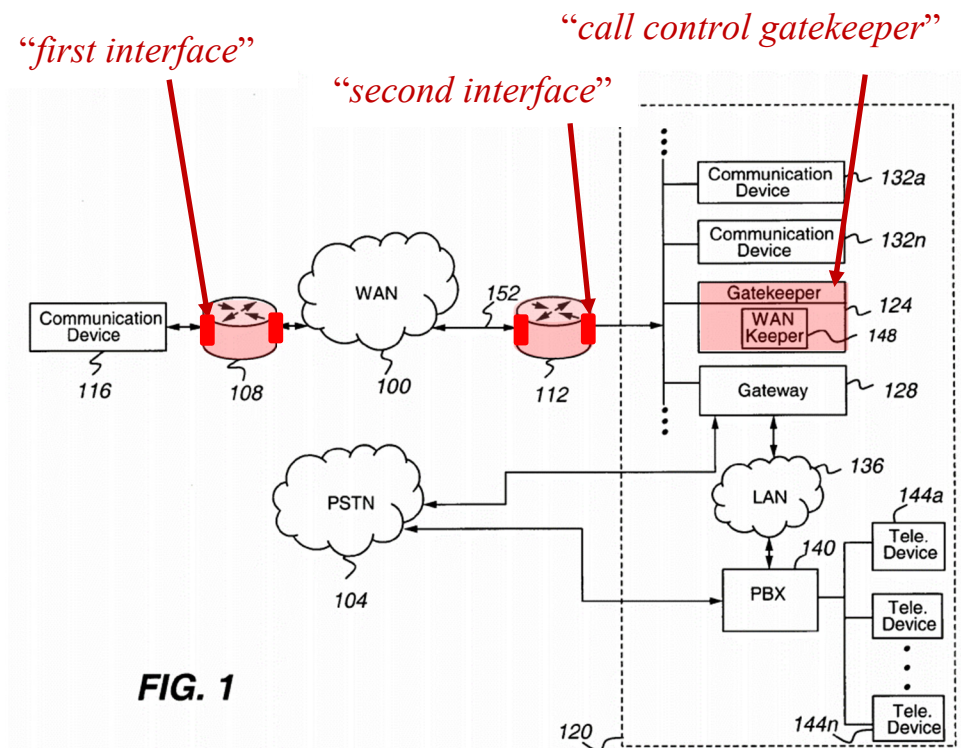
Ex.1005, 5:40-58. Minhazuddin's gatekeeper 124, with the included WAN keeper 148 that provides call admission control, corresponds to the "*call control gatekeeper*."

109. Minhazuddin further discloses that an edge router has an "**interface**."

Ex.1005, 8:6-9. It would have been obvious to a POSITA for each of Minhazuddin's local edge routers 108 and 112 to include an "*interface*" to provide a physical connection to different networks, as it was common and well-known in the art. Indeed, having multiple network interfaces is central to the purpose and

function of a network router. For example, it would have been obvious to a POSITA for local edge routers 108 and 112 to include an interface for connecting to WAN 100 and another interface for connecting to local network where various communication devices reside.

110. Accordingly, as shown below in Figure 1, Minhazuddin's system that provides VoIP services ("a VoIP service provider system") includes a gatekeeper 124 that performs admission control ("a call control gatekeeper"), a first local edge router 108 with a first interface ("a first interface"), and a second local edge router 112 with a second interface ("a second interface").



Ex.1005, FIG. 1 (annotated).

111. I note that Minhazuddin's disclosed system that provides VoIP services (e.g., gatekeeper 124, first and second local edge routers 108, 112) is within the scope of the '368 patent's system, which "include[s] any collection of systems or sub-systems." Ex.1001, 6:13-17. Additionally, Minhazuddin's gatekeeper 124's location on the LAN side of local edge router 112 is consistent with the '368 patent's embodiment of Figure 1 (shown on the face of the patent) where the call control gatekeeper 34 is located behind router 18 (and part of first LAN 12). *See* Ex.1001, FIG. 1. Minhazuddin also contemplates that gatekeeper 124 may be located in different network devices. *See* Ex.1005, 5:29-35, 9:4-7.

112. Thus, Minhazuddin's system for providing VoIP services, which has a gatekeeper 124 with a WAN keeper 148 that performs admission control, a first local edge router 108 with a first interface, and a second local edge router 112 with a second interface, renders obvious "*a VoIP service provider system having a call control gatekeeper, a first interface and a second interface.*"

113. Second, to the extent argued that an "*interface*" is not rendered obvious by Minhazuddin, Hepworth teaches a "VoIP architecture" that includes a "router 120 having associated **interfaces** 124a-c" for connecting to a LAN or a WAN. Ex.1006, [0036]; *see also* Ex.1006, [0045] ("...**routers and one or more of their associated interfaces**..."). Hepworth's router interfaces are illustrated at Figure 1, reproduced below:

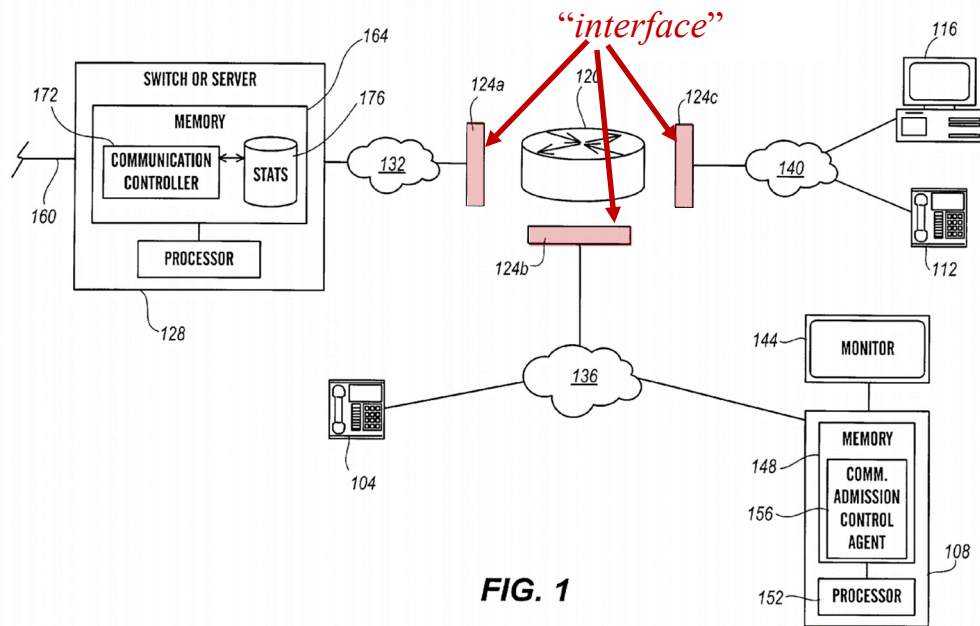
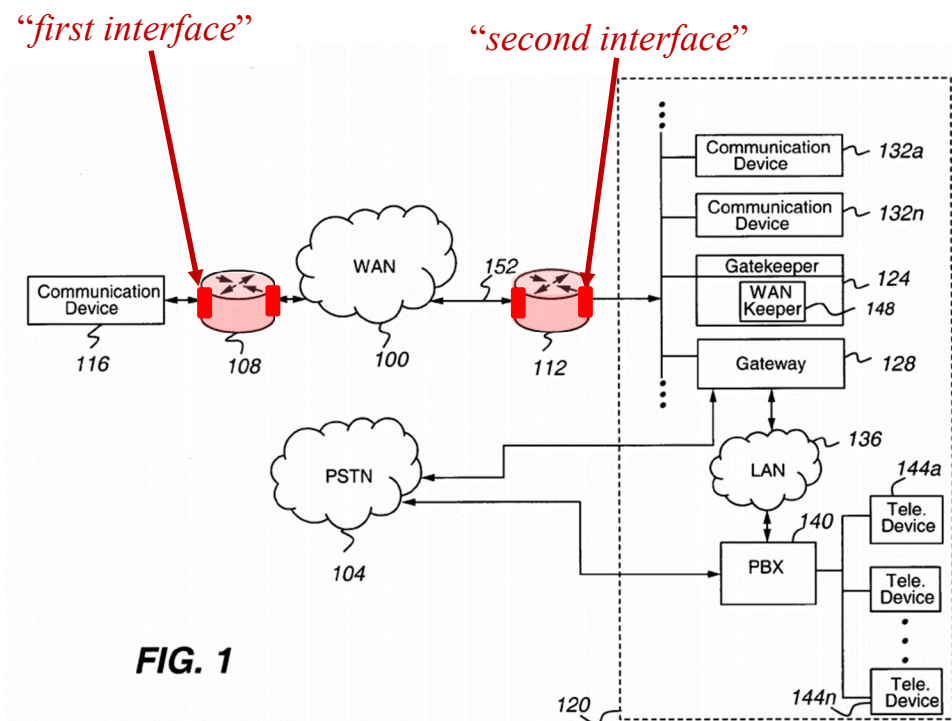


FIG. 1
Ex.1006, FIG. 1 (annotated).

114. It would have been obvious to a POSITA for Minhazuddin's local edge routers 108 and router 112, illustrated in Figure 1, to include associated interfaces, per Hepworth, to facilitate physical connections to respective networks. *See also* Reasons to Combine Hepworth and Minhazuddin.

115. In implementing the proposed combination, Minhazuddin's local edge routers 108 and 112 would have included interfaces to connect to different networks, as illustrated below in annotated Figure 1:



Ex.1005, FIG. 1 (annotated).

116. Thus, Minhazuddin in combination with Hepworth discloses a VoIP service provider system that includes a gatekeeper 124 with a WAN keeper 148 that performs admission control, a local edge router 108 with a first interface, and local edge router 112 with a second interface, which renders obvious “a VoIP service provider system having a call control gatekeeper, a first interface and a second interface,” as claimed.

c. **[1.2] the first interface connected to a first LAN having an interface to a first VoIP telephone;**

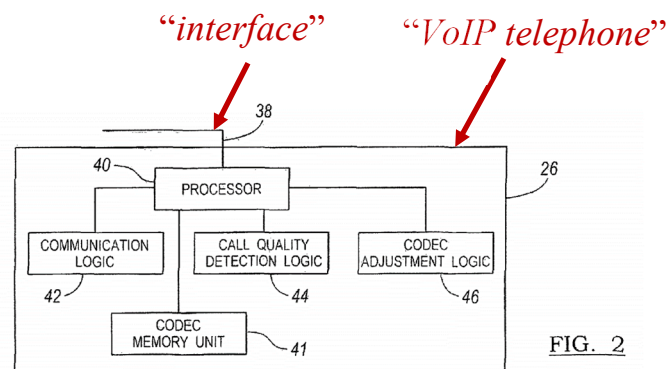
117. In numerous ways, the combination of Minhazuddin, Hepworth, and Shaffer render obvious that the first interface is connected to a first LAN having an interface to a first VoIP telephone.

118. **First**, as discussed at limitation [1.1], Minhazuddin alone and in combination with Hepworth renders obvious that local edge router 108 has an interface, which corresponds to the “*first interface*.”

119. **Second**, in numerous ways, the prior art renders obvious “*a first LAN having an interface to a first VoIP telephone*.”

i. **Minhazuddin alone renders obvious this limitation**

120. As an initial matter, I note that an exemplary LAN “*interface*” in the ’368 patent corresponds to a network link for connecting the VoIP telephone to the LAN. *See* Ex.1001, 3:66-4:1, FIG. 2 (“**interface 38**”).

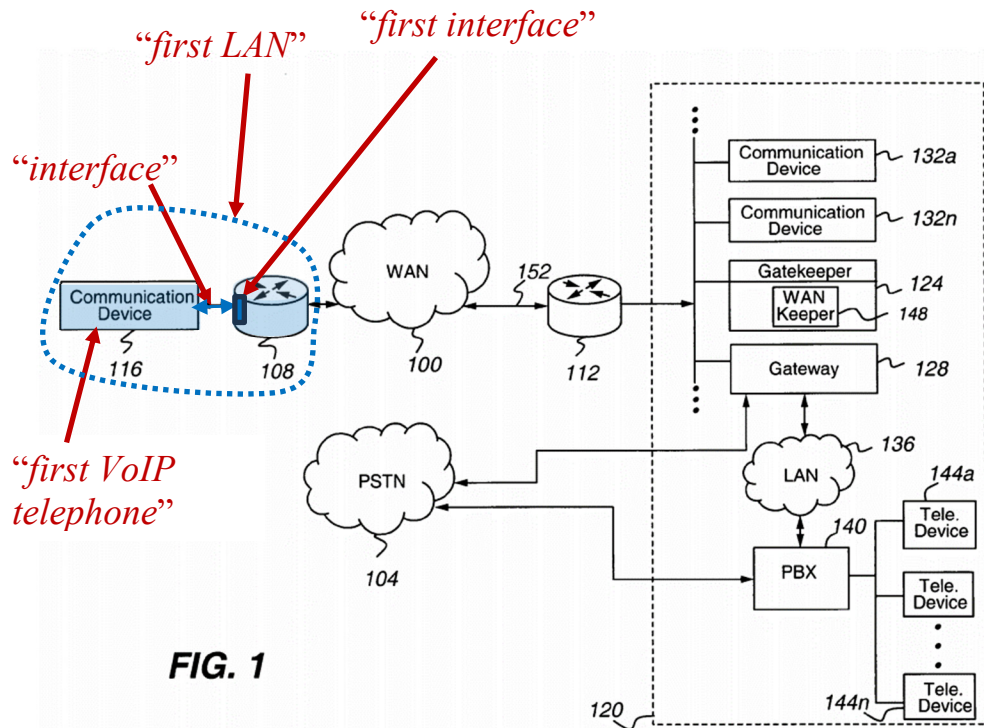


Ex.1001, FIG. 2 (annotated).

121. Minhazuddin in the context of Figure 1 discloses that destination “**communication device 116**” “can be any packet-switched communication device for live voice communications, such as IP hardphones (such as the Avaya™ IP Hardphone), IP Softphones (such as the Avaya™ IP Softphone).” Ex.1005, 4:64-5:3. Minhazuddin’s communication device 116 (e.g., an Avaya VoIP telephone) corresponds to the “*first VoIP telephone*.”

122. A POSITA would have found it obvious for communication device 116 (e.g., an Avaya VoIP telephone) to be part of a LAN for communication, as was well known in the art. *See* Ex.1016, 44 (“...the **Avaya IP Telephone configuration, you will need: access to your company's LAN...**”); *see also* Ex.1013, 2, Fig. 1 (illustrating the IP phone connected to a VLAN); Ex.1017, 15 (disclosing that Avaya IP phone connected to LAN); Ex.1018, 65 (disclosing that if the Avaya IP phone does not operate, unplug the Ethernet cable).

123. Accordingly, it would have been obvious to a POSITA for Minhazuddin’s communication device 116 (e.g., an Avaya VoIP telephone) (“*first VoIP telephone*”), illustrated at Figure 1, to be connected to a network link (“*interface*”) of a LAN (“*first LAN*”) that is connected to the local edge router 108 interface (“*first interface*”):



Ex.1005, FIG. 1 (annotated).

124. Thus, in this first way, Minhazuddin alone renders obvious *“the first interface connected to a first LAN having an interface to a first VoIP telephone.”*

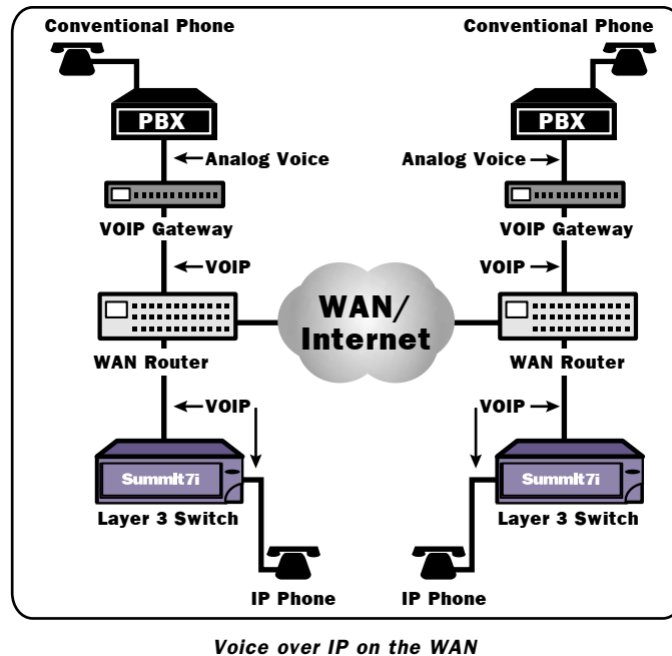
125. **Additionally**, Minhazuddin explains that various embodiments “include[] providing devices and processes in the absence of items not depicted and/or described herein or in various embodiments hereof, including in the absence of such items as may have been used in previous devices or processes.” Ex.1005, 10:15-21. Minhazuddin also recognizes that its disclosed “embodiments and configurations are neither complete nor exhaustive” and that **“other embodiments of the invention are possible utilizing, alone or in combination, one or more of the features set forth above or described in detail below.”** Ex.1005, 4:19-23.

126. Accordingly, although Minhazuddin's embodiment at Figure 1 illustrates only one enterprise network 120, it would have been obvious to a POSITA to implement communication device 116 within an enterprise network having a configuration similar to that of enterprise network 120, including a plurality of communication devices, a gatekeeper, a gateway, etc. This combination would merely represent "utilizing... one or more of the features set forth" in enterprise network 120, as Minhazuddin itself suggests.

127. Such an implementation would have been recognized as desirable because it would allow for a company to support numerous communication devices at an additional enterprise location and beneficially provide access to numerous features, including conferencing and malicious call tracing. Ex.1015, 2 ("...provide access to more than 600 distinct telephony features - from meet-me conferencing to malicious call tracing and emergency call override capabilities.")

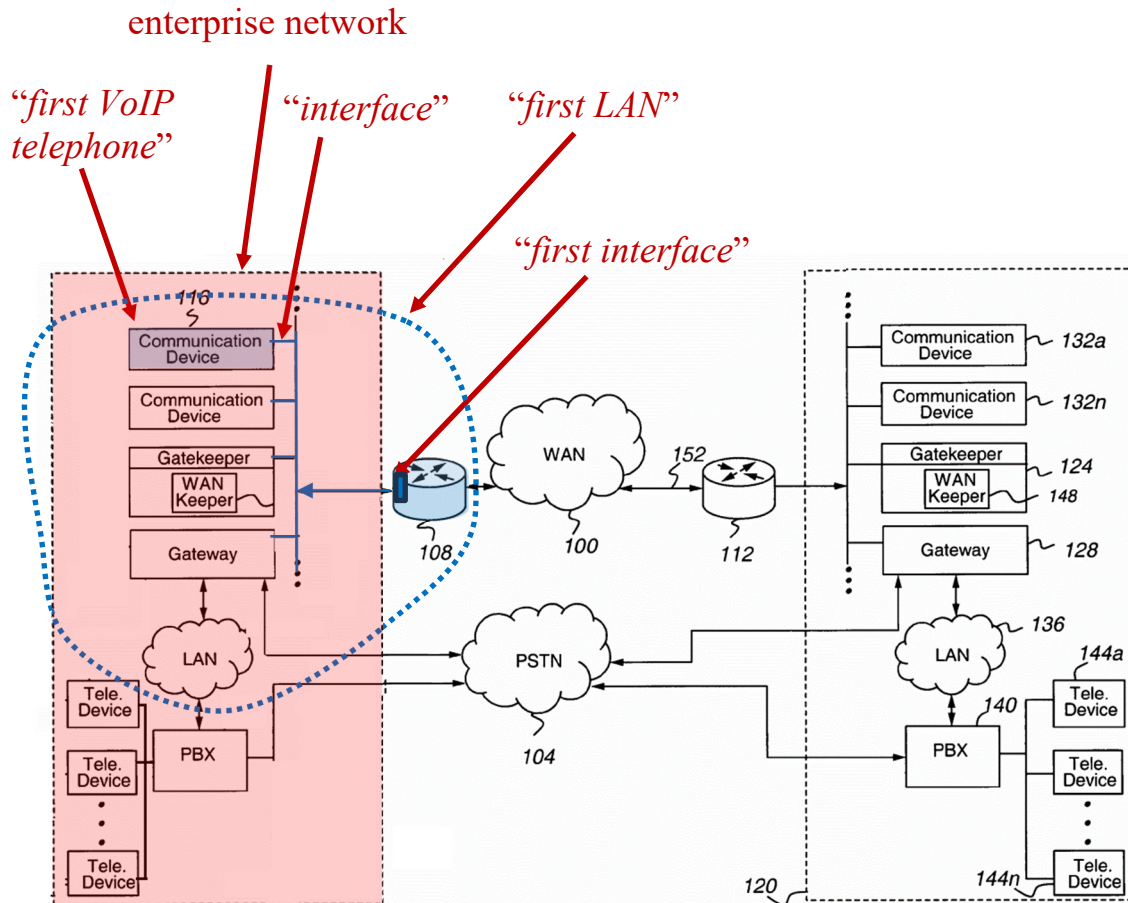
128. The proposed implementation of Minhazuddin's system would be consistent with general knowledge in the art, where a company utilizes two enterprise networks, with a first enterprise network in one location and another enterprise network in another location. As a first example, Merrill Lynch as of July 2003, had utilized Avaya IP telephony systems to implement a first enterprise network for 5800 employees in New Jersey and a second enterprise network for 1800 employees in Japan. Ex.1015, 2; *see also* Ex.1010, 1, Fig. 1; Ex.1011, 1. As a

second example, Extreme Networks had made available as of August 7, 2004, infrastructure solutions for a business with two enterprise networks, as illustrated below:



Ex.1022, p. 5

129. Accordingly, it would have been obvious to a POSITA to utilize another enterprise network that includes a LAN and other communication devices (analogous to enterprise network 120), as illustrated below at Minhazuddin's modified Figure 1:



Ex.1005, FIG. 1 (modified and annotated to show an additional enterprise network).

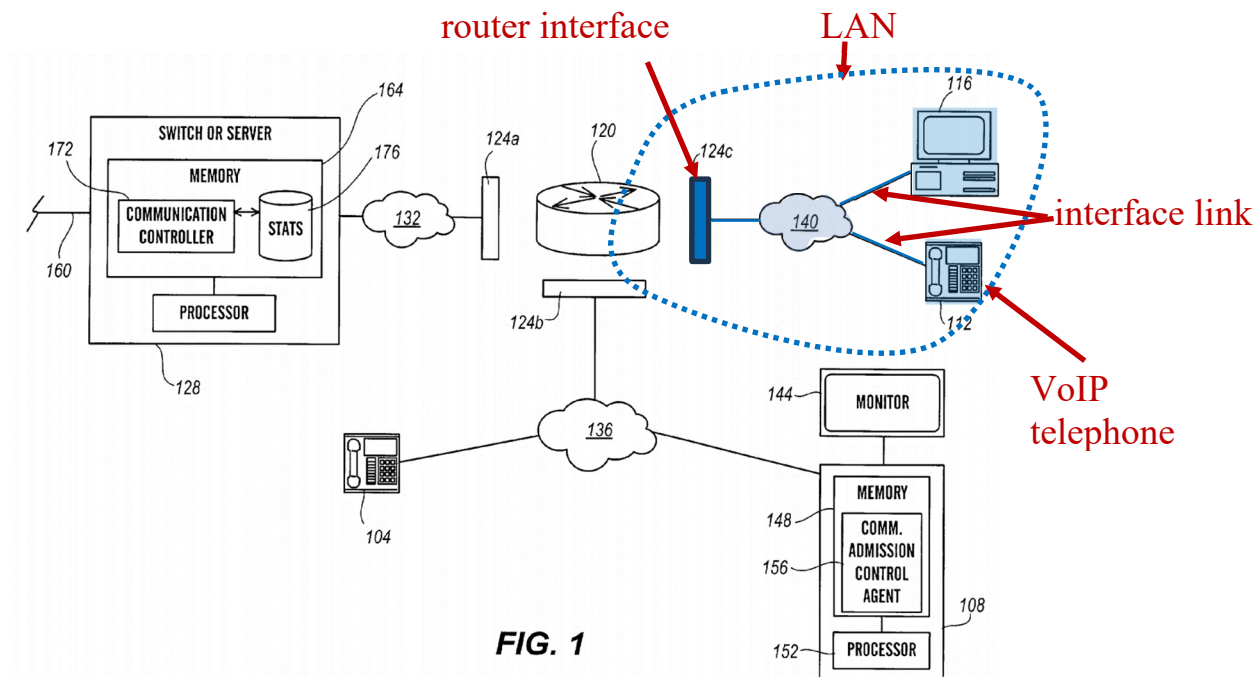
130. I note that implementing a gatekeeper (e.g., an Avaya media server, Ex.1005, 5:9-11) at each location, per the above illustration, is consistent with topology disclosed in other Avaya patents. See e.g., Ex.1020, 4:46-67, FIG. 1. Also, the gatekeeper would serve to control the gateway in its LAN. Ex.1020, 6:45-62.

131. Thus, in this second way, Minhazuddin alone renders obvious “*the first interface connected to a first LAN having an interface to a first VoIP telephone.*”

ii. Minhazuddin and Hepworth render obvious this limitation

132. To the extent argued that “*a first LAN having an interface to a first VoIP telephone*” is not rendered obvious by Minhazuddin, the further combination with Hepworth and Shaffer renders this obvious.

133. Hepworth teaches that LAN subnet 140 has a network interface link connected to IP hard-phone 112. Ex.1006, [0036]. Hepworth illustrates this configuration at Figure 1, reproduced below:



Ex.1006, FIG. 1 (annotated).

134. It would have been obvious to a POSITA, in view of Hepworth, to implement Minhazuddin's communication device 116 in a LAN having a plurality of network links for connecting a plurality of communication devices to the LAN.

See also Reasons to Combine Hepworth and Minhazuddin.

135. Thus, Minhazuddin in combination with Hepworth discloses that the local router 108 interface is connected to a LAN that has a network link for connecting to communication device 116 (e.g., an Avaya VoIP telephone), which renders obvious “*the first interface connected to a first LAN having an interface to a first VoIP telephone,*” as claimed.

iii. Minhazuddin, Hepworth, and Shaffer render obvious this limitation

136. To the extent argued that the claimed “*interface*” is not a network link, the further combination with Hepworth and Shaffer discloses that a “network interface” connects a network terminal (e.g., telephony device) to a LAN. Ex.1007, 3:16-19 (“The interface includes a known network terminal/device 10 utilizing the ITU-T H.323 protocol, and **a packet network interface 13 that is coupled to network terminal 10. Network interface 13 couples the H.323 device to LAN 101.**”); *see also* Ex.1007, 11:21-26 (“...one or more **telephony devices coupled to said packet switched network...**”), FIGS.1, 2.

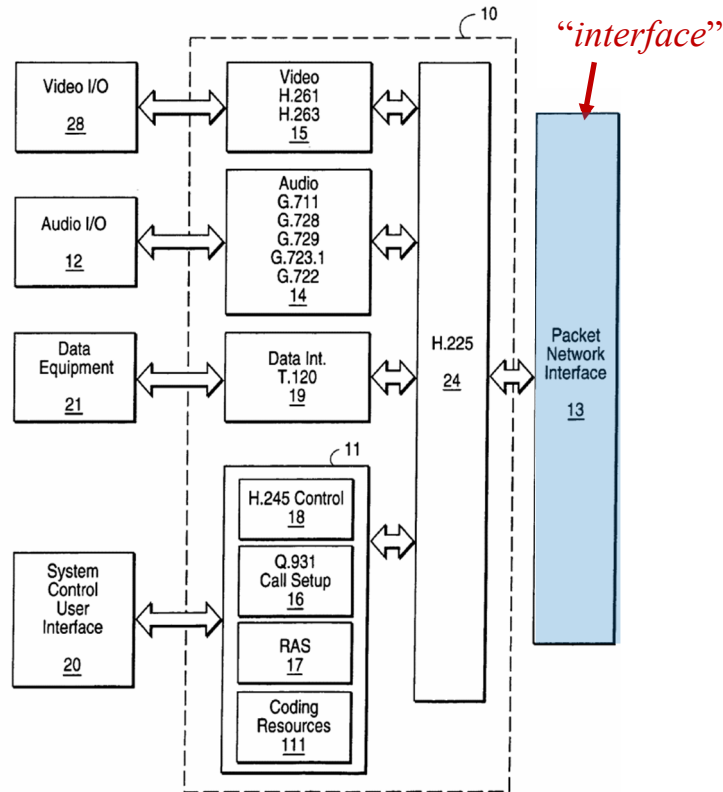


FIG. 2

Ex.1007, FIG. 2 (annotated).

137. It would have been obvious to a POSITA, in view of Shaffer, to utilize a network interface in Minhazuddin's system to facilitate coupling communication devices (each a network terminal), including communication device 116, to a LAN so that packets may be communicated over the LAN. *See also* Reasons to Combine Shaffer and Minhazuddin.

138. Thus, additionally, Minhazuddin in combination with Hepworth and Shaffer discloses that the local router 108 interface is connected to a LAN having a network interface to communication device 116 (e.g., an Avaya VoIP telephone),

which renders obvious “*the first interface connected to a first LAN having an interface to a first VoIP telephone.*”

d. [1.3] *the second interface connected to a second LAN having an interface to a second VoIP telephone;*

139. Minhazuddin, Hepworth, and Shaffer render obvious that the second interface is connected to a second LAN having an interface to a second VoIP telephone.

140. First, as discussed at limitation [1.1], Minhazuddin alone and in combination with Hepworth renders obvious that local edge router 112 has an interface, which corresponds to the “*second interface.*”

141. Second, Minhazuddin teaches that the local edge router 112 is connected to enterprise network 120 that comprises a LAN:

Referring to FIG. 1, a simplified communications network is depicted. The network comprises a packet-switched Wide Area Network or WAN 100, a circuit-switched network 104, **local edge routers 108 and 112**, packet-switched communication device 116, and **enterprise network 120**. The enterprise network 120 includes a gatekeeper 124, a gateway 128, packet switched communication devices 132*a-n*, a **Local Area Network or LAN 136**, a Private Branch Exchange or PBX 140, and telecommunication devices 144*a-n*.

Ex.1005, 4:36-46, FIG. 1.

142. It would have been obvious to a POSITA for Minhazuddin’s local edge router 112 to correspond to a junction point that separates WAN 100 and the LAN

of enterprise network 120. This is consistent with the general understanding in the art that a router of the enterprise network represents a LAN/WAN junction point. Ex.1011, 3 (“Routers within enterprise networks represent the LAN/WAN junction points of a network.”); Ex.1019, 121 (“If your organization has already migrated to IP telephony, the LAN side of the enterprise (or the LAN side of each site location in a multilocation company) is running a standard Ethernet-based network.”), 152 (Figure10-5); Ex.1023, 4 (“In enterprise access networks, a local area network (LAN) is used... Ethernet technology is currently by far the most prevalent access technology in enterprise networks.”); Ex.1010, 3 (“Ethernet has been the dominant technology in enterprise and campus LANs for many years.”), 4 (“One of the advantages of Ethernet in an enterprise domain is the ability to logically partition distinct user groups over the same physical network, using the virtual LAN (VLAN) concept.”); Ex.1011, 1 (“...enterprise network that consists of many local-area networks (LANs).”); Ex.1012, 153 (“The enterprise network system is usually built using interconnected LANs.”).

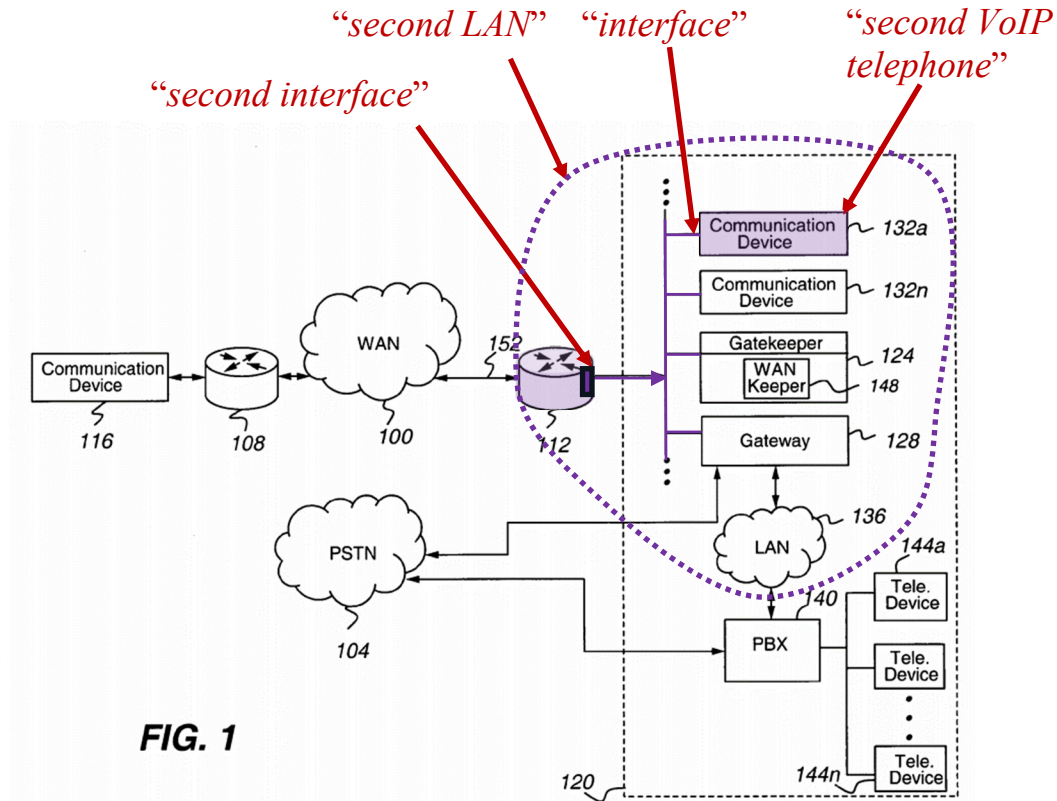
143. As such, Minhazuddin’s description that the local edge router 112 is connected to enterprise network 120 renders obvious that the “*second interface*” (see limitation [1.1]) is “*connected to a second LAN.*”

144. Minhazuddin further teaches that enterprise network 120 includes a “**communication device 132a**,” which “can be any packet-switched

communication device for live voice communications, such as IP hardphones (such as the Avaya™ IP Hardphone), IP Softphones (such as the Avaya™ IP Softphone).” Ex.1005, 4:64-5:4. Minhazuddin’s communication device 132a, e.g., an Avaya VoIP telephone, corresponds to a “*second VoIP telephone*.”

145. A POSITA would have understood that Minhazuddin’s communication device 132a (e.g., an Avaya VoIP telephone) connects to a LAN for communication. *See* Ex.1016, 44 (“To use Avaya IP Softphone in the Avaya IP Telephone configuration, you will need: access to your company’s LAN (via dialup networking or a cable modem), which provides the signaling path between your PC running Avaya IP Softphone and your company’s telephone system.”); *see also* Ex.1013, 2, Fig. 1 (illustrating the IP phone connected to a VLAN); Ex.1018, 65 (disclosing that if the Avaya IP phone does not operate, unplug the Ethernet cable).

146. Minhazuddin’s Figure 1 illustrates that local edge router 112 is connected, via a network link (“*second interface*”), to a LAN (“*second LAN*”) of enterprise network 120:



Ex.1005, FIG. 1 (annotated).

147. Furthermore, to the extent argued that the claimed “interface” is not a network link, consistent with the analysis at limitation [1.2], it would have been obvious to a POSITA to utilize a network interface, in view of Shaffer, to facilitate coupling Minhazuddin’s communication devices (each a network terminal), including communication device 132a, to a LAN so that packets may be communicated over the LAN. *See also* Reasons to Combine Shaffer and Minhazuddin.

148. Thus, Minhazuddin in combination with Hepworth and Shaffer discloses that the local router 112 interface is connected to a LAN that has an

interface to communication device 132a (e.g., an Avaya VoIP telephone), which renders obvious “*the second interface connected to a second LAN having an interface to a second VoIP telephone,*” as claimed.

e. **[1.4] *the first and second VoIP telephones being configured to communicate using a first codec;***

149. Minhazuddin renders obvious that the first and second VoIP telephones are configured to communicate using a first codec.

150. First, as discussed at limitations [1.2]-[1.3], Minhazuddin discloses communication devices 116 and 132a (e.g., each an Avaya VoIP telephone), which respectively correspond to “*the first and second VoIP telephones.*”

151. Second, Minhazuddin teaches that the communication devices communicate using a first selected codec:

In decision diamond 220, the WAN keeper 148 determines if the adjusted metrics are acceptable in view of the VoIP requirements or in view of selected thresholds. The thresholds can be the same as or related to the voice communication application requirements. If not, the WAN keeper 148 selects the codec having the next lower bandwidth requirement and repeats step 216 for that codec. The process is repeated until an acceptable codec is identified. If no codec can be identified, the WAN keeper 148 proceeds to step 228. **If a codec is identified, the WAN keeper proceeds to step 224 and selects that codec as the codec to be used in negotiating with the other endpoint.**

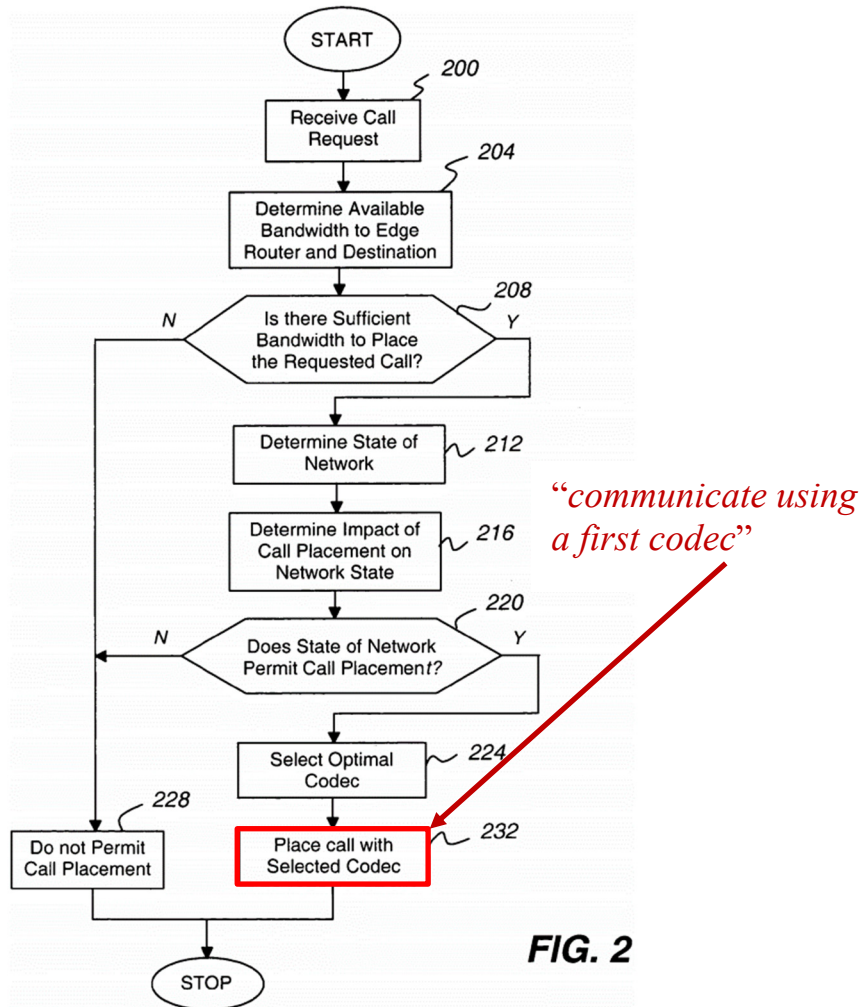
Ex.1005, 7:52-63.

In step 224, the call is permitted to proceed with the selected codec.

In the event that the other endpoint is not configured to handle the selected codec, the WAN keeper selects the codec having the next lower bandwidth requirement until a codec is found that can be processed by the other endpoint. In the event that no such codec exists and a codec having a higher bandwidth requirement than the selected codec must be selected, the WAN keeper 148 can proceed to step 228 and not permit placement of the call.

Ex.1005, 7:64-8:5.

152. As shown below at Figure 2, at step 232, Minhazuddin's communication devices communicate using the "selected codec."



Ex.1005, FIG. 2 (annotated).

153. Based on the above disclosure, a POSITA would have understood that Minhazuddin’s communication devices, including communication device 116 and communication device 132a, are “configured to communicate” with each other using the selected codec because a call is permitted to proceed with that codec. The selected codec corresponds to the claimed “first codec.”

154. Thus, Minhazuddin’s disclosure that communication device 116 (e.g., an Avaya VoIP telephone) and communication device 132a (e.g., an Avaya VoIP

telephone) are configured to communicate using the selected codec, renders obvious “the first and second VoIP telephones being configured to communicate using a first codec,” as claimed.

f. [1.5.1] the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone and

155. Minhazuddin in combination with Hepworth renders obvious that the call control gatekeeper is configured to receive a call quality signal from the first VoIP telephone.

156. First, as discussed at limitation [1.1], Minhazuddin discloses a gatekeeper 124 (that includes WAN keeper 148), which corresponds to “the call control gatekeeper.” Further, as discussed at limitation [1.2], Minhazuddin discloses communication device 116 (e.g., an Avaya VoIP telephone), which corresponds to “the first VoIP telephone.”

157. Second, Minhazuddin discloses that WAN keeper 148 (part of gatekeeper 124) monitors QoS information (e.g., packet delay, jitter, packet loss, etc.) by any suitable technique:

The WAN keeper 148 monitors the end-to-end QoS characteristics or metrics link(s) (or network state) by any suitable technique including receiving feedback signals from one or more response-time-response probes, by polling selected network components, using ICMP ping packets, and/or SNMP/ RTCP packets. The measured QoS metrics include packet delay, jitter, packet loss, the availability of

Differentiated Services Code Point or DSCP or other capability to assign Voice packets a priority, and RSVP status (e.g., an error message in response to a PATH and/or RESV message, and successful reservation setup. As will be appreciated, packet loss refers to packets that are not delivered to a destination.

Ex.1005, 5:53-64.

158. Hepworth supplements Minhazuddin by teaching a suitable technique which includes a session monitor (e.g., Avaya VoIP Monitoring Manager™) receiving a signal with QoS information (e.g., jitter, packet loss, number of dropped packets, round-trip time, etc.) from VoIP endpoints:

The Real-Time Control Protocol or RTCP, a companion protocol to RTP, is used by applications to monitor the delivery of RTP streams. Media packets are transmitted between endpoints during a session according to RTP while additional **performance information governing the communication link (e.g., key statistics about the media packets being sent and received by each end point such as jitter, packet loss, round-trip time, etc.) are collected by the end points and transmitted to a session monitor according to RTCP.** The network monitor can be, for example, VoIP Monitoring Manager™ or VMon™ by Avaya, Inc.

Ex.1006, [0006]; *see also* Ex.1006, [0021] (“The bandwidth information can include...round trip time, jitter buffer delay, jitter, packet loss...a number of dropped packets by a router, router bandwidth utilization...”), [0044] (“[T]est RTP/RTCP packets can be sent between the two endpoints to measure one or more

of the bandwidth information noted above, such as jitter, packet delay, and packet loss.”), [0053] (“In another alternative embodiment, the switch or **server 128 uses one or more of the endpoints (whether or not involved in a session) as network nodes to collect periodically any of the bandwidth information noted above.**”).

This embodiment can be used with the network architecture of the present invention or a conventional network architecture.”).

159. A POSITA would have understood that Hepworth’s endpoints measure network QoS and send a signal with the monitored QoS information to the session monitor, as was known in the art. *See* Ex.1021, 1 (“RTCP carries statistical and control data, while RTP delivers the data. RTCP statistics typically include the number of bytes sent, packets sent, lost packets and round trip delay between endpoints.”); Ex.1024, 587 (“RTP and its Real time Transmission Control Protocol (RTCP) have several properties that can be utilized to improve VoIP calls. The RTCP will frequently send information on the arrived frames, lost frames and jitter to the transmitter, so this information can be and also has been used for adjusting, for example, the size of the jitter buffer.”).

160. It would have been obvious to a POSITA, in view of Hepworth, to implement Minhazuddin’s communication devices, including communication devices 116 and 132*a*, to collect QoS information (e.g., jitter, packet loss, round-trip time, number of dropped packets, router bandwidth utilization, etc.) and

transmit a signal with the QoS information to WAN keeper 148. This QoS information would allow for making call admission control decisions, including whether to allow a call, which codec to use, or whether ongoing calls should change to a lower codec. *See e.g.*, Ex.1005, 8:23-52; *see also* Reasons to Combine Hepworth and Minhazuddin.

161. The proposed implementation would have been recognized by a POSITA as consistent with Minhazuddin's disclosure of monitoring QoS information "by any suitable technique including receiving feedback signals." Ex.1005, 5:53-58.

162. Thus, Minhazuddin in combination with Hepworth discloses that the WAN keeper 148 is configured to receive a signal with QoS information (e.g., jitter, packet loss, round-trip time, number of dropped packets, bandwidth utilization, etc.) from communication device 116, which renders obvious "*the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone,*" as claimed.

- g.** **[1.5.2] *send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec; and***

163. Minhazuddin in combination with Hepworth and Shaffer renders obvious that the call control gatekeeper is configured to send an adjustment signal

based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec.

164. First, as discussed at limitations [1.2]-[1.3], Minhazuddin discloses communication devices 116 and 132a (e.g., each an Avaya VoIP telephone), which respectively correspond to “*the first and second VoIP telephones.*” Further, as discussed at limitation [1.5.1], Minhazuddin in combination with Hepworth discloses that WAN keeper 148 receives a signal with QoS information (e.g., jitter, packet loss, number of dropped packets, round-trip time, etc.), which corresponds to “*the call quality signal.*”

165. Second, Minhazuddin discloses that WAN keeper 148 (implemented in gatekeeper 124) utilizes QoS information to determine if selected thresholds are exceeded and if so, it implements measures to control bandwidth:

In decision diamond 408, the WAN keeper 148 determines whether there is sufficient bandwidth and/or whether the state of the network can support additional calls. In one configuration, this determination is made by comparing the unused or available bandwidth level(s) with a maximum threshold.

Maximum Threshold=Allocated Link VoIP Bandwidth-Bandwidth required for one VoIP call,
where the allocated link VoIP bandwidth is the bandwidth dedicated to VoIP calls.

When the utilized bandwidth is above the maximum threshold or the unused bandwidth is below the maximum threshold, the WAN keeper 148 institutes one or more measures to control bandwidth utilization. In another configuration, when one or more QoS metrics exceeds selected thresholds...the bandwidth control measures are also implemented.

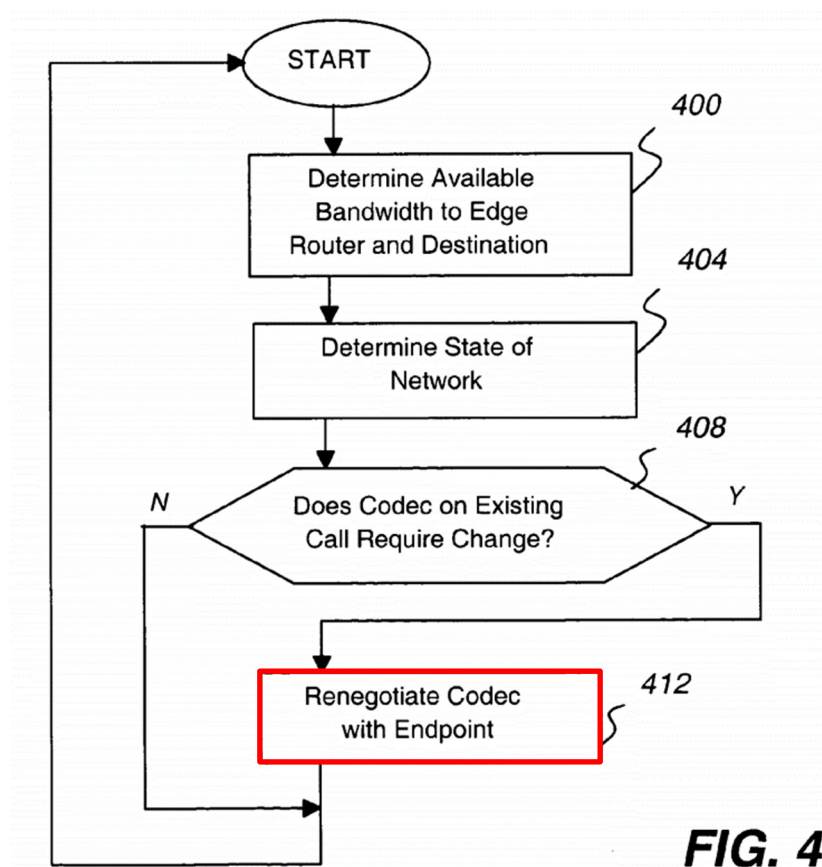
Ex.1005, 8:23-41.

166. The implemented bandwidth control measures include having the VoIP endpoints (also referred to a communication devices) renegotiate to a lower bit rate codec on existing VoIP calls:

These bandwidth control measures are implemented in step 412. In one measure, **the codecs used on existing VoIP calls are renegotiated to a lower bit rate codec (that will use less bandwidth)**...In one technique, during call setup multiple possible codecs are negotiated between the endpoints, with the highest bit rate codec being used initially in the call. In the event bandwidth control measures require implementation, a lower bit rate codec (that was previously negotiated) will thereafter be used between the endpoints until bandwidth utilization level(s) and/or the QoS metric drops below selected threshold(s).

Ex.1005, 8:42-52. The lower bit codec that uses less bandwidth corresponds to the claimed “*second codec*.”

167. As shown below at Figure 2, at step 232, Minhazuddin’s endpoints, e.g., communication device 116 and communication device 132*a*, renegotiate to a lower bit rate codec:



Ex.1005, FIG. 4 (annotated).

168. It would have been obvious to a POSITA to notify Minhazuddin's endpoints, e.g., communication device 116 and communication device 132a to renegotiate to a lower bit rate codec, by having WAN keeper 148 send a signal to both communication endpoints directing them to change codecs.

169. Furthermore, Shaffer discloses a "telephony-over-LAN (local area network) system." Ex.1007, 1:16-18; *see also* Ex.1007, 3:14-24. Shaffer's system includes a bandwidth allocation server ("BWAS") that may "be incorporated into [a] gatekeeper." Ex.1007, 3:4-10. The BWAS monitors QoS information and, if a

threshold is met, directs telephony endpoints to renegotiate to a codec that uses less bandwidth:

A bandwidth adjustment server or bandwidth allocation server (BWAS) is provided which monitors system bandwidth usage, sends requests to user terminals to identify their coding capabilities, and **directs each of the user terminals to adjust their coding algorithms based on system bandwidth usage. If system bandwidth usage is high, the BWAS requires the user terminals to employ a less bandwidth-intensive coding algorithm**; similarly, when system bandwidth usage is low, the BWAS will allow the user terminals to employ higher bandwidth-use coding algorithms.

Ex.1007, 1:51-61.

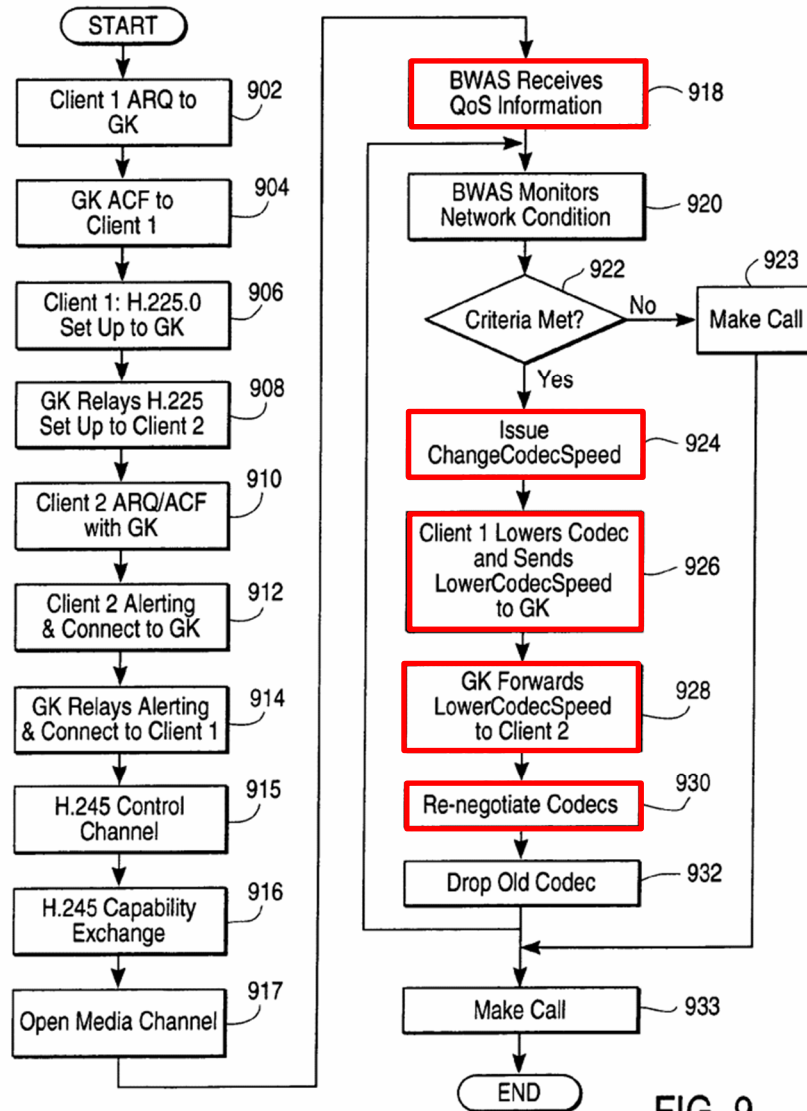
The BWAS according to one embodiment monitors band width usage, and if there is a disparity between the band width allocated to new connections versus ongoing ones or an increase in data traffic, **the BWAS sends a Lower Codec Speed message to all active H.323 entities. This causes the H.323 entities to renegotiate their codecs.**

Ex.1007, 2:9-14.

Then, in a step 918, the BWAS 109 receives the QoS information regarding the call that is being established. In a step 920, the BWAS monitors the condition of the network. In a step 922, if the particular change codec criteria have been met, then in a step 924, the BWAS 109 causes the gatekeeper GK to issue a ChangeCodecSpeed command to the relevant calling party endpoint(s), in this example, endpoint Client 1. The ChangeCodecSpeed command includes a “higher” or “lower”

parameter, as appropriate. Then, in a step 926, the endpoint Client 1 adjusts its codec speed and sends a LowerCodecSpeed command (or, if appropriate, a HigherCodecSpeed command) to the gatekeeper GK, which identifies the existing call whose codec is to be renegotiated. In a step 928, this command is forwarded to the endpoint Client 2. The codec renegotiation then takes place over the H.245 control channel, in a step 930. Once the renegotiation has occurred, the previously-used codec(s) are dropped, in a step 932, the system cycles back to step 920 (after an optional configurable delay), and the call is made (step 933). If, in step 922, the criteria had not been met, the communication would have been established without codec renegotiation, in step 923.

Ex.1007, 10:17-38; *see also* Ex.1007, 11:20-42 (“...a bandwidth allocation server configured to **cause a renegotiation of which of said coding algorithms said one or more telephony devices communicates** with while said one or more telephony devices are communicating using a predetermined coding algorithm...”); Ex.1007, 12:21-47, FIGS. 4-9.



Ex.1007, FIG. 9.

FIG. 9

170. It would have been obvious to a POSITA, when implementing Minhazuddin's bandwidth control measures of renegotiating to a lower bit rate codec), for WAN keeper 148 to send a signal to the endpoints (also referred to a communication devices) directing them to use a lower bit rate codec that uses less bandwidth, in view of Shaffer. A POSITA would have been motivated to combine Shaffer's signaling teachings with Minhazuddin because it would inform the

endpoints (including communication devices 116 and 132a) to renegotiate to a lower bit rate codec. *See also* Reasons to Combine Shaffer and Minhazuddin.

171. The signal sent by WAN keeper 148 to renegotiate to a lower bit codec, per the combination with Shaffer, corresponds to “*an adjustment signal*” and the lower bit codec that uses less bandwidth corresponds to the claimed “*second codec*.”

172. I note that the prior art’s teaching of sending a signal to renegotiate to a lower bit rate codec that uses less bandwidth is within the scope of the ’368 patent’s embodiment. Ex.1001, 1:67-2:3 (“When the gatekeeper receives this call quality signal, **the gatekeeper will instruct the VoIP telephones to renegotiate the telephone call and utilized a lower bandwidth codec.**”).

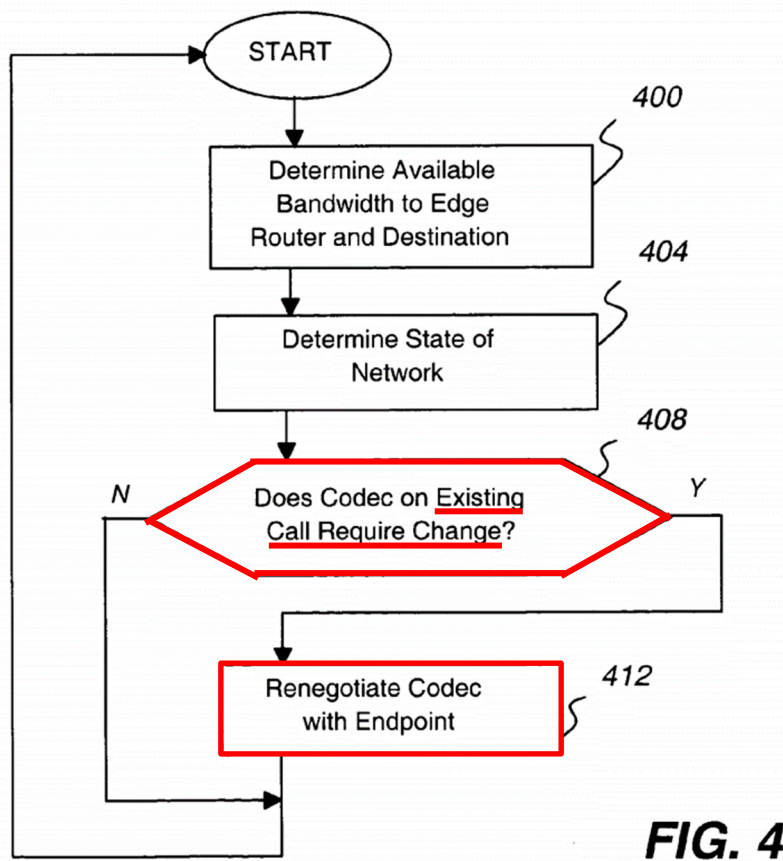
173. Thus, Minhazuddin in combination with Hepworth and Shaffer discloses that WAN keeper 148 sends a signal, based on the QoS information (e.g., jitter, packet loss, round-trip time, number of dropped packets, router bandwidth utilization, etc.), to the communication devices 116, 132a (e.g., each an Avaya VoIP telephone) directing them to renegotiate to a lower bit rate codec, which renders obvious that the call control gatekeeper configured to “*send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec*,” as claimed.

- h. **[1.6] *wherein the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones during a telephone call between the first and second VoIP telephones.***

174. Minhazuddin in combination with Hepworth and Shaffer renders obvious that the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones during a telephone call between the first and second VoIP telephones.

175. First, as discussed at limitation [1.5.2], Minhazuddin in combination with Hepworth and Shaffer discloses that the signal to renegotiate to a lower bit codec based on the QoS information signal is sent to communication devices 116, 132a (e.g., each an Avaya VoIP telephone), which corresponds to “*the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones.*”

176. Second, Minhazuddin discloses that the “**codecs used on existing VoIP calls are renegotiated** to a lower bit rate codec (that will use less bandwidth).” Ex.1005, 8:42-45. This is illustrated at Figure 4, reproduced below:



Ex.1005, FIG. 4 (annotated).

177. It would have been obvious to a POSITA, when communication devices 116, 132a (e.g., each an Avaya VoIP telephone) are participating in an existing call and the QoS metrics exceed selected thresholds (Ex.1005, 8:37-41), for WAN keeper 148 to send the adjustment signal to each endpoint to renegotiate the codec, as analyzed at limitation [1.5.2].

178. Additionally, Shaffer discloses sending a signal to endpoint telephones to adjust their codecs during an existing call. Shaffer explains that “**existing call(s) will have its (or their) codecs renegotiated to a lower level,**” for example, by

having the **“BWAS send[] a Lower Codec Speed message to all active H.323 entities...to renegotiate their codecs.”** Ex.1007, 2:9-14, 8:51-67; *see also* Ex.1007, 9:43-46 (“...if there **exist connections** whose QoS is presently more than needed or requested, the connection may be eligible to have its codec renegotiated.”), 10:17-32 (“...identifies the **existing call whose codec is to be renegotiated...**”), 11:27-31 (“...a **bandwidth allocation server configured to cause a renegotiation of which of said coding algorithms said one or more telephony devices communicates with while said one or more telephony devices are communicating** using a predetermined coding algorithm...”), 12:7-11 (“...**changing said codec speed for said existing connection...**”), FIG. 9.

179. It would have been obvious to a POSITA, when implementing Minhazuddin’s system, as modified per Shaffer (see limitation [1.5.2]), to send the adjustment signal to the communication devices 116, 132a (e.g., each an Avaya VoIP telephone) that are participating in an existing call, because it would allow for achieving Minhazuddin’s goal of having the **“codecs used on existing VoIP calls renegotiated** to a lower bit rate codec (that will use less bandwidth).” Ex.1005, 8:42-45. This implementation would beneficially allow for the existing call to continue at a lower bit rate even though available bandwidth has been reduced. *See also* Reasons to Combine Shaffer and Minhazuddin.

180. Thus, Minhazuddin in combination with Hepworth and Shaffer discloses that WAN keeper 148 sends the adjustment signal to the communication devices 116, 132a (e.g., each an Avaya VoIP telephone) during an existing call between the communication devices 116, 132a (e.g., each an Avaya VoIP telephone), which renders obvious that “*the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones during a telephone call between the first and second VoIP telephones,*” as claimed.

7. Claim 2

i. [2.0] *A system comprising:*

181. See analysis at limitation [1.0].

j. [2.1] *a VoIP service provider system having a call control gatekeeper, a first interface and a second interface;*

182. See analysis at limitation [1.1].

k. [2.2] *the first interface connected to a first LAN having an interface to a first VoIP telephone;*

183. See analysis at limitation [1.2].

l. [2.3] *the second interface connected to a second LAN having an interface to a second VoIP telephone;*

184. See analysis at limitation [1.3].

m. [2.4] *the first and second VoIP telephones being configured to communicate using a first codec;*

185. See analysis at limitation [1.4].

- n. **[2.5.1] *the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone and***

186. See analysis at limitation [1.5.1].

- o. **[2.5.2] *send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec; and***

187. See analysis at limitation [1.5.2].

- p. **[2.6] *wherein the call quality signal is based upon a number of dropped voice packets during a VoIP telephone call.***

188. The prior art renders obvious that the call quality signal is based upon a number of dropped voice packets during a VoIP telephone call.

189. As discussed at limitation [1.5.1], Minhazuddin in combination with Hepworth discloses that WAN keeper 148 receives a signal with QoS information, which corresponds to “*the call quality signal.*” Minhazuddin’s QoS information includes “**packet loss...packets that are not delivered to a destination.**” Ex.1005, 5:53-64. Furthermore, Hepworth discloses that the VoIP QoS information corresponds to “**performance information governing the communication link (e.g., key statistics about the media packets being sent and received by each end point such as jitter, packet loss, round-trip time, etc.) are collected by the end points and transmitted to a session monitor.**” Ex.1006, [0006]; *see also* Ex.1006, [0003]-[0005]. In a further example, the key statistics includes “**a number of dropped packets.**” Ex.1006, [0021]. It would have been obvious that during a

VoIP telephone call voice packets are utilized (Ex.1005, 5:61 (“...voice packets...”)), and, consequently, it would have been obvious for the number of dropped packets to correspond to voice packets.

190. It would have been obvious to a POSITA for the QoS information transmitted to WAN keeper 148 (see limitation [1.5.1]) to include packet loss statistics, such as a number of dropped voice packets during the VoIP call, because this information represents well-known QoS information that impacts voice protocols, which are sensitive to dropped packets, and also because this information is useful for call control. Ex.1011, 10 (“...some protocols are sensitive to dropped packets...”); Ex.1025, 7:2-5 (“These statistics may include, but are not limited to, latency, jitter, a number of octets per packet, and/or the number of dropped packets.”). A POSITA would have recognized that voice protocols are particularly sensitive to dropped packets because dropped packets typically cannot be resent in time to be played (unlike with a data file where there are no hard deadlines for receiving data), making real-time voice conversation appear interrupted or broken up. *See* Ex.1074, [0090] (treating as lost “a packet that was successfully received over the network but arrived too early or too late to be played out correctly”); Ex.1087, 967 (discussing the problem of “what to play out when it has finished playing out a packet and the next packet is not available”); Ex.1096, [0211] (“VOP in some forms can only play or decode packets that have arrived in time to be

meaningful to the user.”); *see also* Reasons to Combine Hepworth and Minhazuddin.

B. Ground 2: Claims 1 and 2 are obvious over Mundra in combination with RFC3435 and RFC3660.

191. The combination of Mundra, RFC3435, and RFC3660 renders obvious claim 1 as discussed below.

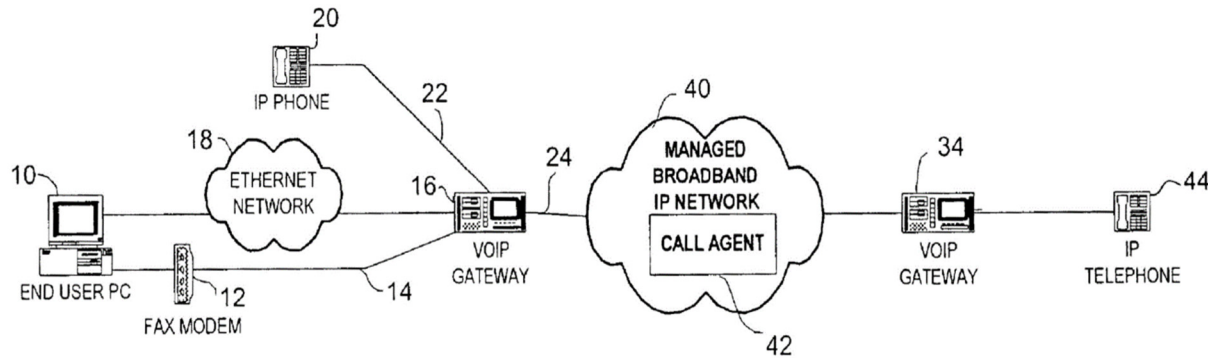
1. Summary of Mundra

192. Like the '368 patent, U.S. Patent Publication No. 7,643,414 to Mundra (“Mundra,” Ex.1008) generally relates to “Speech transmissions over packet networks, such voice over Internet Protocol (VoIP).” Ex.1008, [0002]. More specifically, Mundra is directed to facilitating a change of codecs during an active VoIP call:

A technique to **change a codec in realtime during transmission of a voice over packet call** placed over a packet switched network, such as the Internet, is described. The invention provides users real-time control over the cost and quality of a voice over packet call by monitoring the dynamically changing resources and network conditions on a packet network and allowing users to change codecs of the call transmission before and during the call according to user-defined specifications.

Ex.1008, Abstract.

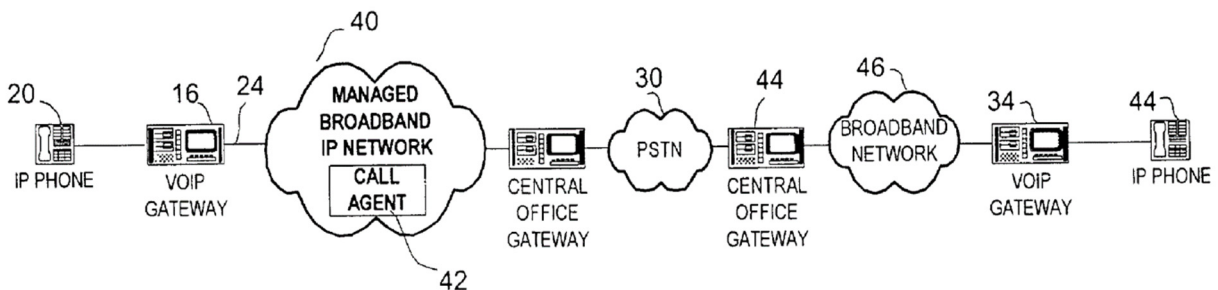
193. Mundra’s exemplary system is illustrated at Figure 2 and 3, reproduced below:



Ex.1008, FIG. 2.

FIG. 2 illustrates a diagram of a broadband IP network connecting two VoIP telephony devices is illustrated. IP network is a managed broadband network. As part of the managed network, an network provider could utilize bandwidth on the public Internet 30 as a seamless connection between two ISP-managed networks 42, 46, as illustrated in FIG. 3.

Ex.1008, [0018].



Ex.1008, FIG. 3.

Referring again to FIG. 2, the system includes a managed network 40 between a first IP telephony device 20 on one end and a second IP telephony device 44, such an IP phone, on the second end. Gateways 16, 34 support voice over packet calls and monitors network conditions

within managed network 40 Gateway 16 sends and receives messages from a call agent module 42 that is managed by the network managing entity Gateway 16 must be capable of detecting changing resource or network conditions. The ability to detect and monitor changing resource and network conditions can result in Significant cost reductions and/or improved quality.

Ex.1008, [0019].

194. Mundra notes that some existing call signaling protocols, such as “media gateway control protocol (MGCP),” provided some support for changing codecs:

However, after placing a call and speech is being transmitted, a user at one end may desire a higher quality call through either a better network managed network connection or a better codec, which may not correspond to the lowest cost transmission. **Support for such a change in codec** is generally absent from signaling protocols, such as **media gateway control protocol (MGCP), except for the case when a switch over to basic PCM is desired** when the call is being established and the call is formatted for modem or facsimile transmissions.

Ex.1008, [0010].

195. Mundra describes facilitating a codec change in other situations and for other reasons. For example, Call Agent 42 can respond to commands from a user’s IP phone to change the codec during a call:

A user may also desire a higher quality, or lower quality, codec during transmission for reasons unrelated to call quality or cost. The

desired codec may or may not be the codec of lowest cost[.] Dynamic codec changes are supported within gateways and call control protocols[.] Gateway 16 receives such user commands and takes appropriate action by signaling agent 42 in the managed network 40 to effect a change in the codec. **Call agent module 42 is managed by the network provider to receive commands from IP phone 20 or gateway 16 and the change the codecs in realtime during a call transmission.**

Ex.1008, [0020]; *see also* Ex.1008, claims 1, 5.

2. RFC3435

196. Like the '368 patent, the Internet Engineering Task Force's Request for Comments (RFC) 3435 ("RFC3435") generally relates to "Voice over IP" communications. Ex.1027, 1.

197. RFC3435 describes the "Media Gateway Control Protocol (MGCP)":

This document describes an application programming interface and a corresponding protocol (MGCP) which is used between elements of a decomposed multimedia gateway. The decomposed multimedia gateway consists of a Call Agent, which contains the call control "intelligence", and a media gateway which contains the media functions, e.g., conversion from TDM voice to Voice over IP.

Ex.1027, 1.

198. RFC3435 is an update to an earlier edition of MGCP, which published as RFC2705. Ex.1027, 1 ("Obsoletes: 2705"); *see* Ex.1026. Among the changes

between RFC2705 and RFC3435 was that RFC3435 “Removed packages from [the] protocol specification.” Ex.1027, 163. RFC3435 did not do away with packages, however, but simply anticipated that the packages would “be provided in separate documents instead.” Ex.1027, 163. One of those separate documents is RFC3660, discussed below.

3. RFC3660

199. The Internet Engineering Task Force's Request for Comments (RFC) 3660 (“RFC3660”) is titled “Basic Media Gateway Control Protocol (MGCP) Packages.” Ex.1028, 1. RFC3660 “provides a basic set of Media Gateway Control Protocol (MGCP) packages,” which are “updates of packages from RFC 2705.” Ex.1028, 1. “The basic set of packages specified in [RFC3660] is for use with MGCP 1.0 as defined in [RFC3435].” Ex.1028, 3, 60. Because RFC3660 was intended to be combined and used together with RFC3435, RFC3660 also generally relates to “Voice over IP” communications. Ex.1027, 1.

4. Reasons to Combine Mundra with RFC3435 and RFC3660.

a. Mundra, RFC3435 and RFC3660 are Analogous Art

200. RFC3435 and Mundra both disclose VoIP communications and are therefore analogous art to the '368 patent which pertains to VoIP communications. Ex.1001, 1:6-8, Abstract; Ex.1008, [0002]; Ex.1027, 1. RFC3660 was intended for use with RFC3435, so RFC3660 also pertains to VoIP communications and is analogous art. Ex.1027, 1. Thus, given the similar subject matter, a POSITA

considering Mundra would have also considered RFC3435 and RFC3660's additional teachings.

b. **Motivation to combine with RFC3435 and RFC3660**

201. A POSITA would have been motivated to combine the teachings of Mundra, RFC3435, and RFC3660 to produce numerous predictable and beneficial results.

202. Mundra describes establishing a VoIP call, including the initial selection of a codec. Mundra explains, however, that it may be desirable to change the codec during the call. Mundra notes that some VoIP signaling protocols have some support for this feature:

[0009] During a voice over packet call, the choice of codec used for initially establishing the call depends upon the codecs that are supported at sending and the receiving telephony devices. **Both the sending and receiving devices must use the same codecs during speech transmissions** to take advantage of the lower transmission rates and higher quality transmissions offered by specialized codecs for speech over packet networks. Users of an IP telephony system may subscribe to a service where lowering the cost of the voice call is the default policy, and therefore the lowest cost codec is present for the users.

[0010] However, after placing a call and speech is being transmitted, a user at one end may desire a higher quality call through either a better network managed network connection or a better codec, which may not

correspond to the lowest cost transmission. Support for such a change in codec is generally absent from signaling protocols, such as media gateway control protocol (MGCP), except for the case when a switch over to basic PCM is desired when the call is being established and the call is formatted for modem or facsimile transmissions.

Ex.1008, [0009]-[0010].

203. Thus, Mundra expressly refers to media gateway control protocol (MGCP) as an example of a VoIP signaling protocol that was known to support changing codec “when a switch over to basic PCM is desired when the call is being established and the call is formatted for modem or facsimile transmissions.”

Ex.1008, [0010]. While Mundra goes on to further describe changing a voice call codec in other situations (e.g., Ex.1008, [0016], [0025]), Mundra’s only other reference to a VoIP signaling protocol is in its background discussion, where Mundra notes that “ITU-T H.323 was drafted to standardize terminals, equipment, and services for multimedia transmissions over LANs and IP network which do not have guaranteed QoS [quality of service].” Ex.1008, [0005]. MGCP is designed to interface with “H.323 conformant systems.” Ex.1027, 7. Thus, it would have been obvious to a POSITA to consider Mundra’s disclosure in the general context of MGCP, either as an adaptation or extension of the codec-change capabilities that Mundra acknowledges were already present in MGCP.

204. When the Mundra application was filed on August 19, 2002, the then-current version of the MGCP standard was RFC2705. *See* Ex.1026. But by the time that Mundra published in February 2004, RFC3435 and RFC3660 had been published, obsoleting RFC2705. *See* Ex.1027, 1; Ex.1028, 1. This was known to those of skill in the art:

The MGCP is defined by Request for Comment (RFC) 3435, RFC 3660 and RFC 3661, and these three RFCs are updated versions for RFC 2705, where RFC 3435 defines the MGCP, RFC 3660 defines the packages of the MGCP, and RFC 3661 defines the return codes of the MGCP.

Ex.1039, 4:49-53.

205. I understand that obviousness is evaluated at the time of alleged invention, which as I referenced earlier, I am taking to be the '368 patent's earliest claimed priority date of November 10, 2005. Thus, at the time of Mundra's publication in 2004—and continuing on through November 10, 2005—a POSITA desiring more information about the MGCP protocol discussed in Mundra would have found it obvious to refer to RFC3435 and RFC3660, since those were the then-current specifications defining MGCP. It is my opinion that Mundra's express reference to using MGCP would have led a POSITA to consider RFC3435 and RFC3660. I also believe that Mundra's disclosure was intended to be understood in the general context of MGCP, as specified by RFC3435 and RFC3660, because

Mundra uses a variety of terminology that appears to be borrowed from the MGCP specification, including “call agent,” “VoIP Gateway,” “event,” and “signal.”

c. Reasonable expectation of success

206. The results would have been predictable and there would have been a reasonable expectation of success in the combination since Mundra, RFC3435, and RFC3660 address the same technology, as analyzed above in the analogous art section. Also, the results would have been predictable and there would have been a reasonable expectation of success in the combination since RFC3435 and RFC3660 described established standards that were well-known in the art. For example, others had already described processing a “CODEC change” using “MGCP-based call signaling.” Ex.1075, [0013]²; *see also* Ex.1076, 7:54-62 (“MGCP allows media channels to be modified in a variety of ways with having to be closed.... The encoding format can be changed by changing the codec.”).

207. The reasonable expectation of success is also supported by the fact that Mundra already monitors network QoS (Ex.1008, [0022]-[0023]) and RFC3435

² I note that US2002/0094081 refers to a potential problem in maintaining synchronization of cryptographic keys when changing codecs. Ex.1075, [0013] (“a high probability exists that after the CODEC change the two MTAs [multimedia terminal adapters] would loose [*sic*] synchronization on their [stream cipher] RC4 key streams and all RTP [real time protocol] packets would not be decrypted”). Since neither Mundra nor the ’368 patent discuss encrypting voice calls, that potential problem is not relevant and would not inhibit or dissuade a POSITA from combining Mundra with RFC3435 and RFC3660 in the context of ordinary (non-encrypted) voice calls.

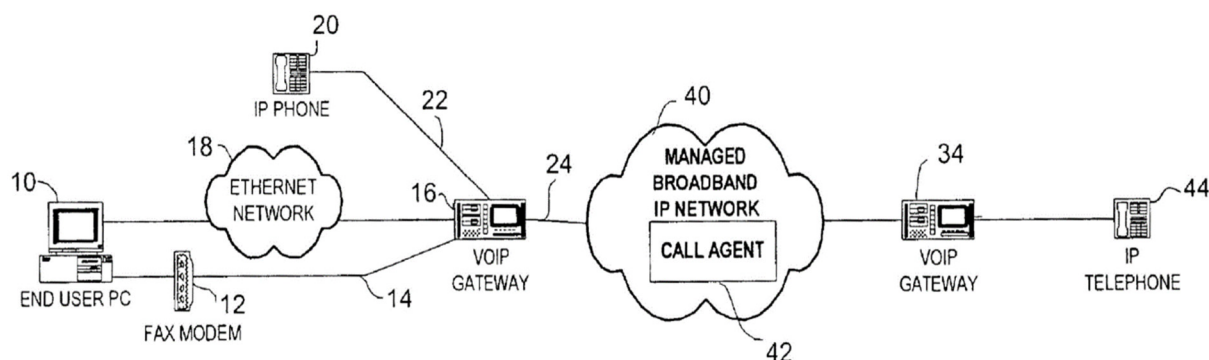
describes how the VoIP endpoints measure, collect, and send the QoS information (such as a “quality alert,” Ex.1027, 44) so that appropriate action can be taken. A POSITA would have possessed the skills required to make the proposed combination, including being able to implement the combined teachings of Mundra, RFC3435, and RFC3660 in corresponding hardware or software.

5. Claim 1

a. [1.0] *A system comprising:*

208. To the extent limiting, Mundra renders obvious a system as recited in the preamble.

209. Specifically, Mundra, in the context of Figure 2 discloses a “**system** [that] includes a managed network 40 between a first IP telephony device 20 on one end and a second IP telephony device 44, such an IP phone, on the second end [and] Gateways 16, 34 [that] support voice over packet calls and monitors network conditions within managed network 40.” Ex.1008, [0019], FIG. 2.



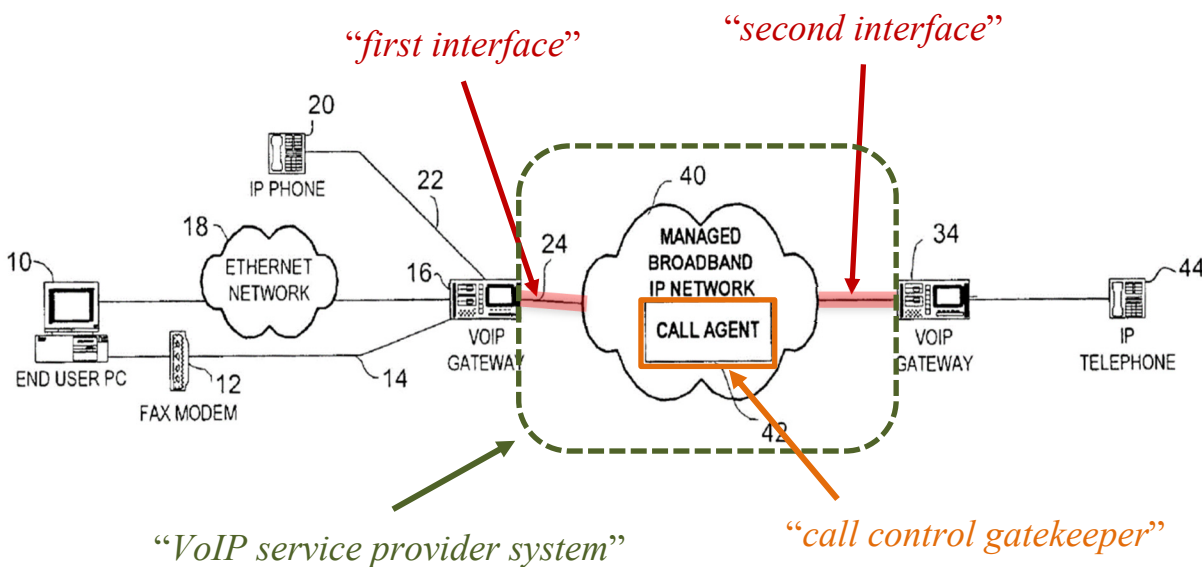
Ex.1008, FIG. 2.

210. Thus, Mundra's system renders obvious "[a] system," as recited in the preamble.

- b. [1.1] a VoIP service provider system having a call control gatekeeper, a first interface and a second interface;

211. Mundra in combination with RFC3435 renders obvious a VoIP service provider system having a call control gatekeeper, a first interface and a second interface.

212. Mundra illustrates the recited components in FIG. 2, which I have annotated below to identify each of the recited components.



Ex.1008, FIG. 2 (annotated).

213. First, Mundra's system includes a "managed network 40," which Mundra also refers to as a "managed broadband [IP] network." Ex.1008, [0018]-[0019], FIG. 2. The managed network 40 is controlled by a "network provider."

Ex.1008, [0018]. Since the system in FIG. 2 is used to provide “Internet telephony service” to a user, it would have been obvious to a POSITA for Mundra’s “network provider” to be a service provider. This understanding is bolstered by Mundra’s reference to “ISP-managed networks 42 [*sic*, 40], 46.” An “ISP” was commonly understood to be an internet service provider. *See* Ex.1009, 13:12-13 (“an Internet service provider (ISP)”; Ex.1020, 7:24 (“Internet Service Providers or ISP’s”).

214. FIG. 2 also shows two “Gateways 16, 34” that “support voice over packet calls.” Ex.1008, [0019]. The figure labels each these as a “VOIP GATEWAY.”

215. In view of these disclosures, a POSITA would have understood that managed network 40 and its connections to VoIP gateways 16, 34 correspond to a “*a VoIP service provider system*,” as claimed. I note that Mundra’s disclosed VoIP provider system is within the scope of the ’368 patent’s system, which “include[s] any collection of systems or sub-systems.” Ex.1001, 6:13-17.

216. Second, within the managed network 40 is a “call agent module 42 that is managed by the network managing entity.” Ex.1008, [0019]. A POSITA would have understood that the network managing entity is the entity managing the managed network 40, which Mundra also refers to as the “network provider.” Ex.1008, [0018]. A POSITA would have understood Mundra’s other names for the “call agent module 42,” which Mundra also refers to as the “call agent,” “signaling

agent 42,” and “call controller agent 42.” Ex.1008, [0020], FIG. 2. Since Mundra refers to the MGCP signaling protocol, it would have been obvious for call agent module 42 to act as an MGCP call agent.

217. RFC3435 further describes the operation of a call agent under the MGCP signaling protocol, including that the call agent “contains the call control ‘intelligence.’” Ex.1027, 1. For example, “the Call Agent handles the call signaling and call processing functions.” Ex.1027, 7. The call agent also controls the operation of gateways: “the gateways are expected to execute commands sent by the Call Agents.” Ex.1027, 5; *see also id.*, 47 (“LocalConnectionOptions is an optional structure used by **the Call Agent to direct the handling of the connection by the gateway.**”), 20 (“The resources allocated to a connection, and more generally the handling of the connection, are chosen by the gateway **under instructions from the Call Agent.**”). Finally, “the Call Agent implements the ‘signaling’ layers of the H.323 standard, and **presents itself as an ‘H.323 Gatekeeper.**” Ex.1027, 7. In view of these disclosures, a POSITA would have understood Call Agent 42 to correspond to the claimed “*call control gatekeeper.*”

218. Third, VoIP “gateway 16 connects to the broadband network [40] with a high speed Internet connection 24 such as a digital subscriber line (DSL), cable modem 24, or T1/T5 line.” Ex.1027, [0017]. A POSITA would have understood that Mundra’s high-speed Internet connection 24 corresponds to a “*first interface.*”

219. To the extent that Mundra's high-speed Internet connection 24 is argued not to be a "*first interface*," it would have been obvious to a POSITA for the managed network 40 to have a network interface for connecting to the Internet connection 24. For example, Mundra explains that Internet connection 24 may be a "digital subscriber line (DSL)." Ex.1027, [0017]. DSL was a common and well-known communication technology for providing high speed Internet service to a customer, typically over the copper twisted pair wires that had historically been used to provide analog telephone service (sometimes called "POTS" service, a reference to the "plain old telephone system"). To provide DSL service to a customer, an internet service provider's network would have a digital subscriber line access multiplexer (DSLAM) with a port connected to the copper wires going to the customer's location. *See, e.g.*, Ex.1077, [0014] ("A port on a digital subscribe[r] loop access multiplexor (DSLAM) is an example of a resource used exclusively for the user location."). The DSLAM port would correspond to the claimed "*first interface*."

220. As another example, Mundra explains that Internet connection 24 may be a "cable modem" connection. Ex.1027, [0017]. Cable modems were also a common and well-known communication technology for providing high speed Internet service to a customer, in this instance over the coaxial cable that had historically been used to provide cable television service. To provide cable modem

service to a customer, an internet service provider's network would have a cable modem termination system (CMTS) with a port connected (either directly, or more commonly, indirectly via an intervening fiber optic network) to the coaxial cable going to the customer's location. *See, e.g.*, Ex.1078, [0006] ("a cable modem termination system or 'CMTS,' a function of which is to provide complementary functionality to the subscriber's cable modems so as to enable data connectivity to a wide-area network like the Internet"), [0007] ("data from a subscriber received at one port of the CMTS"). The CMTS port would correspond to the claimed "*first interface*."

221. Fourth, Mundra shows in FIG. 2 that VoIP gateway 34 connects to managed network 40, but that connection is not labeled. The unlabeled connection is a "*second interface*."

222. To the extent that the unlabeled connection between VoIP gateway 34 and managed network 40 is argued not to be a "*second interface*," it would have been obvious to a POSITA for the managed network 40 to have a network interface for connecting to VoIP gateway 34. I note that Mundra indicates that VoIP gateway 34 is similar to VoIP gateway 16, since Mundra gives them the same name, illustrates them the same in FIG. 2, and refers to them together, e.g., "Gateways 16, 34." Ex.1027, [0019]. Thus, a POSITA would have understood that the connection between VoIP gateway 34 and managed network 40 to be one of the same types of

high-speed Internet connection described for VoIP gateway 16. Thus, the same analysis discussed above regarding the network interface that would be part of managed network 40, such as a DSLAM port or a CMTS port, applies equally to the unlabeled connection to VoIP gateway 34. The network interface, such as a DSLAM port or a CMTS port, for connecting managed network 40 to the communication link to VoIP gateway 34 would be a “*second interface*.”

223. Thus, Mundra’s managed network 40 with communication links to VoIP gateways 16, 34 and including a call controller agent 42, which operates as a gatekeeper as taught by RFC3435, renders obvious “*a VoIP service provider system having a call control gatekeeper, a first interface and a second interface*,” as claimed.

c. [1.2] *the first interface connected to a first LAN having an interface to a first VoIP telephone;*

224. Mundra renders obvious that the first interface is connected to a first LAN having an interface to a first VoIP telephone.

225. First, as discussed at limitation [1.1], Mundra renders obvious that the communication link from managed network 40 to VoIP gateway 16 corresponds to the “*first interface*.”

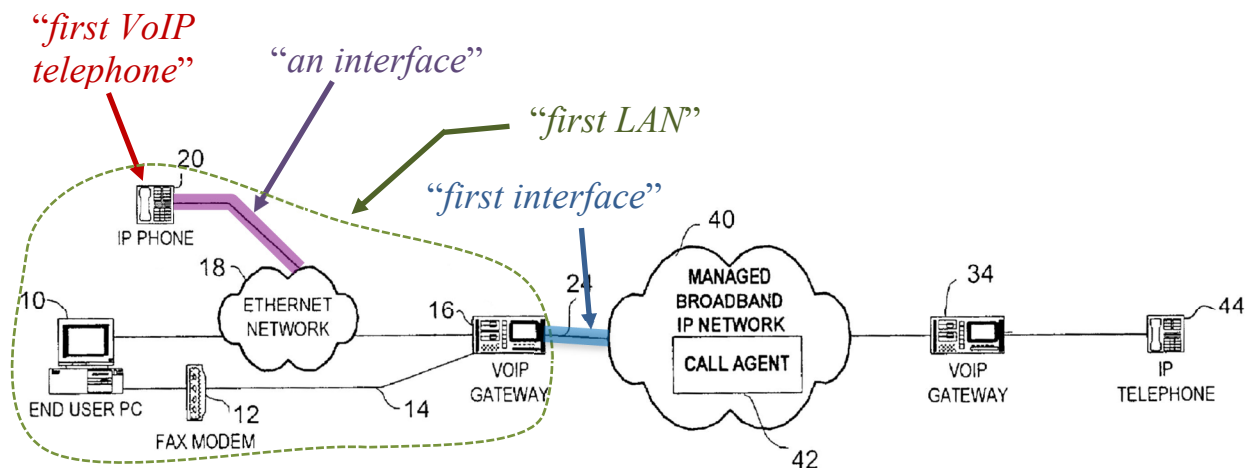
226. Mundra further teaches that a “**digital VoIP telephone 20 may also connect to gateway 16.**” Ex.1008, [0017]; *see also* Ex.1008, [0019] (“Referring again to FIG. 2, the system includes a managed network 40 between a **first IP**

telephony device 20 on one end and a second IP telephony device 44, such an IP phone, on the second end. Gateways 16, 34 support voice over packet calls.”). Mundra’s first VoIP telephone 20 corresponds to a “*first VoIP telephone*,” as claimed.

227. Second, Mundra illustrates VoIP telephone 20 connecting to VoIP Gateway 16 via “telephony port 22.” Ex.1008, [0017]. As I discussed earlier, however, it was well-known and would have been background knowledge to a POSITA that most commercially available VoIP telephones did not require a special telephony port, but instead a VoIP telephone would include a standard Ethernet RJ-45 port for connecting to an Ethernet network. Mundra illustrates an Ethernet network 18 near VoIP phone 18, so a POSITA would have found it obvious to modify Mundra’s depicted network topology to connect VoIP telephone 18 to Ethernet network 18. *See, e.g.*, Ex.1079, FIG. 1, [0027] (“IP phone sets 12 and 14 connected to the LAN.... **This is a typical arrangement which is well identified in the art.**”). With this modification, the connection from IP phone 20 to Ethernet network 18 is “*an interface*.” This change is consistent with how IP phones were usually used, especially in corporate networks where many IP phones would be connected to an Ethernet network. If the Ethernet network 18 included an Ethernet switch supporting Power-over-Ethernet (which was common), the change

would also let VoIP telephone be powered from the switch, thereby simplifying the wiring and bringing other benefits as I discussed previously.

228. I have provided an exemplary nonlimiting implementation showing this change below. The Ethernet network 18 and its attached devices (including IP telephone 20, end user PC 10, and VoIP gateway 16) form a “*first LAN*.”



Ex.1008, FIG. 2 (modified to connect IP phone 20 to Ethernet network 18 instead of VoIP Gateway 16, annotated).

229. **Third**, To the extent that it is argued that the connection from IP phone 20 to Ethernet network 18 is not “an interface,” it would have been obvious for there to be a network interface facilitating the connection of the IP phone 20 to Ethernet network 18. For example, as I discussed previously, it would have been understood by a POSITA that Mundra’s Ethernet network 18—depicted as a cloud—would include an Ethernet switch, and Ethernet switches commonly included numerous RJ-45 connection ports for connecting to Ethernet devices such

as IP telephones. The RJ-45 connection port for connecting to IP phone 20 would be “*an interface.*”

230. Thus, Mundra renders obvious “*the first interface connected to a first LAN having an interface to a first VoIP telephone.*”

d. [1.3] *the second interface connected to a second LAN having an interface to a second VoIP telephone;*

231. Mundra renders obvious that the second interface is connected to a second LAN having an interface to a second VoIP telephone.

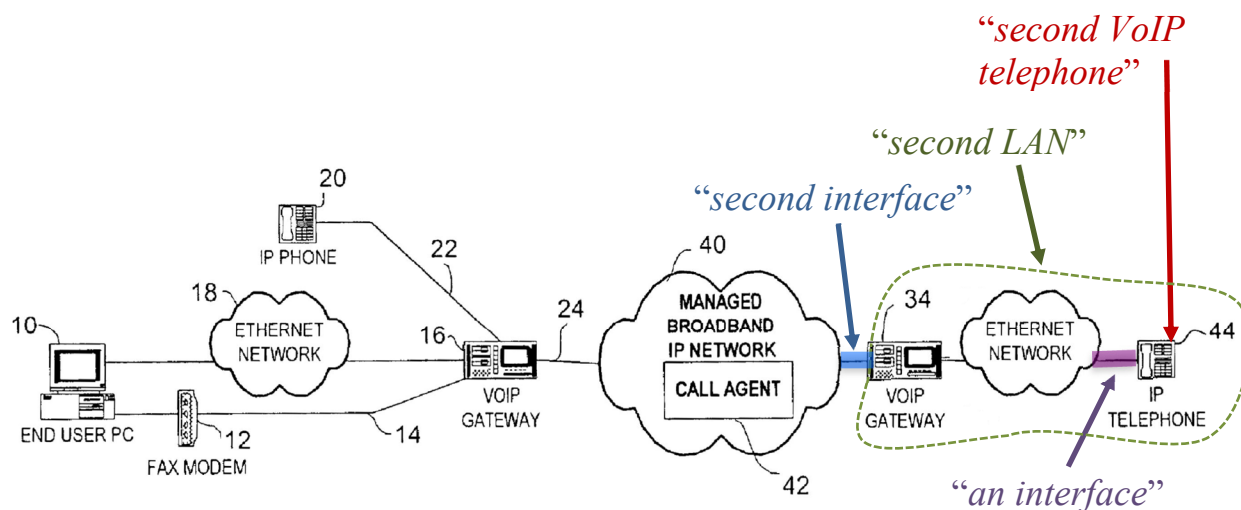
232. First, as discussed at limitation [1.1], Mundra renders obvious that VoIP gateway 34 has a communication link to managed network 40 that corresponds to the “*second interface.*”

233. Second, Mundra further teaches that VoIP gateway 34 supports voice over packet calls between the first IP telephone 20 and “**a second IP telephony device 44.**” Ex.1008, [0019]. The second IP telephone 44 is “*a second VoIP telephone.*”

234. Third, FIG. 2 shows that IP telephone 44 connects to the second VoIP gateway 34. But as discussed at limitation [1.2], it would have been obvious to a POSITA to modify Mundra’s FIG. 2 so that (first) IP phone 20 connects to (first) VoIP Gateway 16 via Ethernet network 18. For the same reasons, it would have been obvious to further modify FIG. 2 so that IP phone 44 connects to VoIP gateway 34 through an Ethernet network. *See, e.g.,* Ex.1079, FIG. 1, [0027] (“IP

phone sets 12 and 14 connected to the LAN.... **This is a typical arrangement which is well identified in the art.**"). Adding an Ethernet network to the right side of FIG. 2 would enable well-known benefits, such as the ability to connect multiple devices (whether additional IP phones or computers like end user PC 10) and the ability to use Power-over-Ethernet to provide power to IP phone 44.

235. I have provided an exemplary nonlimiting implementation showing this change below. The Ethernet network and its attached devices (including IP telephone 44 and VoIP gateway 34) form a “*second LAN*.” The connection from IP telephone 44 to the Ethernet network is “*an interface*.”



Ex.1008, FIG. 2 (modified to add an Ethernet network between IP phone 44 and VoIP Gateway 34, annotated).

236. Fourth, to the extent that it is argued that the connection from IP phone 44 to its Ethernet network is not “*an interface*,” it would have been obvious for there to be a network interface facilitating that connection. For example, as a I

discussed previously, it would have been understood by a POSITA that an Ethernet network—depicted as a cloud—would include an Ethernet switch, and Ethernet switches commonly included numerous RJ-45 connection ports for connecting to Ethernet devices such as IP telephones. The RJ-45 connection port for connecting to IP phone 44 would be “*an interface.*”

237. Thus, Mundra renders obvious “*the second interface connected to a second LAN having an interface to a second VoIP telephone,*” as claimed.

e. [1.4] *the first and second VoIP telephones being configured to communicate using a first codec;*

238. Mundra renders obvious that the first and second VoIP telephones are configured to communicate using a first codec.

239. First, as discussed at limitations [1.2]-[1.3], Mundra’s first VoIP telephone 20 and second VoIP telephone 44 correspond to “*the first and second VoIP telephones.*”

240. Second, Mundra teaches that the “first IP telephony device 20 on one end and [the] second IP telephony device 44...on the second end... support voice over packet calls.” Ex.1008, [0019]. Since Mundra describes “a VoIP call,” and IP phones 20 and 44 are the only telephones shown in FIG. 2, it would have been obvious for there to be a voice of packet call (e.g., a VoIP telephone call) between them. Ex.1008, [0021]; *see also* Ex.1008, claim 1 (“...establishing a voice call on a

packet network...”). Initially, the voice over packet call would be established “*using a first codec.*”

During a voice over packet call, the choice of **codec used for initially establishing the call** depends upon the codecs that are supported at sending and the receiving telephony devices. Both the sending and receiving devices must use the same codecs during speech transmissions to take advantage of the lower transmission rates and higher quality transmissions offered by specialized codecs for speech over packet networks.

Ex.1008, [0009]; *see also* Ex.1008, [0008] (“...When placing a VoIP call, there is typically an **original VOP codec**...”).

...establishing a voice call on a packet network between a first user and a second user wherein the call is transmitted **using a first codec**...

Ex.1008, claim 1.

241. Thus, Mundra’s disclosure that the first IP telephone 20 and the second IP telephone 44 are configured to perform voice packet calls using a first codec renders obvious “*the first and second VoIP telephones being configured to communicate using a first codec,*” as claimed.

f. **[1.5.1] *the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone and***

242. Mundra, alone or in combination with RFC3435 and RFC3660, renders obvious that the call control gatekeeper is configured to receive a call quality signal from the first VoIP telephone.

243. First, as discussed at limitation [1.1], Mundra in combination with RFC3435 renders obvious that Call Agent 42 corresponds to “*the call control gatekeeper*.” Further, as discussed at limitation [1.2], Mundra’s first IP telephone 20 corresponds to “*the first VoIP telephone*.”

244. Second, Mundra discloses that Call Agent 42 receives commands from IP phone 20 that prompt a change of codecs:

Dynamic codec changes are supported within gateways and call control protocols[.] Gateway 16 receives such user commands and takes appropriate action by signaling agent 42 in the managed network 40 to effect a change in the codec. **Call agent module 42 is managed by the network provider to receive commands from IP phone 20 or gateway 16 and the[n] change the codecs in realtime during a call transmission.** Monitored network events and user commands are received by gateway 16, which then sends commands and call control protocols to call controller agent 42 to cause a codec change.

Ex.1008, [0020].

245. Mundra further explains that a user can send “commands” that control which codecs will be enabled for use on voice calls, which in turn affects the voice call quality:

For example, a user may desire a higher quality and higher cost call when speaking to a client or customer, but would settle for a lower-quality and lower cost call when making personal calls. Before or during call transmission, **commands are sent from pre-assigned keys**

on a dialpad or directly from an end user PC **to the gateway and agent** which switches event detection **to enable the best, lowest, or midrange codec, according to the category of classification the user desires, such as highest quality or lowest cost.**

Ex.1008, [0025].

...a **call protocol agent that receives commands** from each gateway and **from each user**, wherein the call agent changes a first codec to a second codec in realtime during transmission of the call.

Ex.1008, claim 5.

...the **call protocol agent receives commands to change codecs** from a telephony device operated by the user.

Ex.1008, claim 7.

246. Mundra gives the example above of a command sent using a key on a “dialpad.” A POSITA would have been familiar with the term dialpad, which refers to the collection of keys found on physical telephones, such as VoIP hard phones. *See, e.g.*, Ex.1019, 146 (“hard telephony dialpad buttons”).

247. It would have been obvious that a command that controls whether to “enable the best, lowest, or midrange codec” would affect call quality. For example, enabling the “best” codec would provide a call with the “highest quality.” *See* Ex.1008, [0025]. Conversely, enabling the “lowest... codec” (e.g., the codec capable of conducting a voice call with the lowest amount of bandwidth) would allow for a call at the “lowest cost.” *See* Ex.1008, [0025]. Thus, Mundra’s

“command” sent from IP phone 20 to Call Agent 42 to control which codecs are enabled is a “*call quality signal*.”³

248. Accordingly, since Mundra explains that Call agent module 42 is managed to receive commands from IP phone 20 that control which codecs are enabled, Mundra renders obvious that “*the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone.*”

249. Third, to the extent that it is argued that the claimed “*call quality signal*” has to provide information about the currently detected call quality, the combination of Mundra, RFC3435, and RFC3660 renders obvious such a call quality signal.

250. Mundra describes monitoring indicators of call quality, such as packet delay and jitter:

[0020] ...**Monitored network events** and user commands **are received by gateway 16**, which then sends commands and call control protocols to call controller agent 42 to cause a codec change. A user-specified change in codec is made according to user-defined criteria including cost and call quality based on the network conditions as monitored by gateway during the call. An alternative to enabling the

³ I note that the '368 patent describes, for example, a “a call quality signal indicating that voice packets are being dropped.” Ex.1001, 3:61-62. Claim 1, however, does not recite any limitations relating to what, if anything, the recited “*call quality signal*” indicates.

present invention by call control protocols is to add a vendor's extension to an existing telephony system.

[0021] Broadband **gateways 16 monitor and detect events** affecting a VoIP call, **including** processing power, **delay, and jitter**.

Ex.1008, [0020]-[0021].

251. As I discussed earlier, packet delay and jitter were well-known measurements of call quality. Thus, Mundra's each of monitored events corresponds to a "*call quality signal*."

252. Similar to Mundra, RFC3660 describes various events used in MGCP, including a Quality alert event:

Quality alert (qa):

The packet loss rate or the combination of delay and jitter exceeding a quality threshold. The quality thresholds for delay, jitter, and packet loss rate are provisioned values.

Ex.1028, 46.

253. RFC3435 also discusses the quality alert, explaining that a quality alert is an event that is detected by an endpoint (e.g., Mundra's IP phone 20) with respect to a connection:

Some events and signals, such as the in-line ringback or the quality alert, are performed or detected on connections terminating in the endpoint rather than on the endpoint itself.⁴

Ex.1027, 44.

254. A POSITA would have understood that an MGCP quality alert, which provides information about packet loss rate, delay, and/or jitter, is a “*call quality signal*.” Since a quality alert event is detected “in the endpoint,” it would have been obvious for Mundra’s IP phone 20 to send a quality alert message.

255. RFC3435 further explains that such event messages are sent by an endpoint to the Call Agent per the Call Agent’s request:

The concept of events and signals is central to MGCP. **A Call Agent may ask to be notified about certain events occurring in an endpoint** (e.g., off-hook events) by including the name of the event in a RequestedEvents parameter (in a NotificationRequest command – see Section 2.3.3).

Ex.1027, 28.

256. For example, RFC3435 provides an example for specifying a request to receive “The quality alert event on all connections”:

⁴ The passage is not stating that a quality alert event could not be detected by a VoIP endpoint; rather, it is clarifying that a quality alert would be detected by a VoIP endpoint with respect to a specific connection terminating at that VoIP endpoint.

In addition, the range and wildcard notation of events can be used, instead of individual names, in the RequestedEvents and DetectEvents parameters. The event code "all" is reserved and refers to all events or signals in a package. The star sign ("*") can be used to denote "all connections", and the dollar sign ("\$") can be used to denote the "current" connection (see Section 2.1.7 for details).

The following are examples of such notations:

M/[0-9]	Digits 0 to 9 in the MF package.
hf	Hook-flash, assuming that the line package is a default package for the endpoint.
[0-9*#A-D]	All digits and letters in the DTMF packages (default for endpoint).
T/all	All events in the trunk package.
R/qa@*	The quality alert event on all connections.
G/rt@\$	Ringback on current connection.

Ex.1027, 90-91.

257. It would have been obvious to modify Mundra's system by having Call Agent 42 request to be notified about monitored events such as a quality alert, as taught by RFC3435. Since a quality alert is an event that is "detected on connections terminating in the endpoint" (Ex.1027, 44), it would have been obvious in the combination for a quality alert event to be detected and sent by Mundra's IP Phone 20 (a VoIP endpoint) to Call Agent 42. Accordingly, a POSITA would have found it obvious for Mundra's Call Agent 42 ("*call control gatekeeper*") to be "*configured to receive*" a "monitored event" such as a quality alert message ("*a call quality signal*").

258. Thus, the combination of Mundra, RFC3435, and RFC3660 renders obvious for Call Agent 42 (“*call control gatekeeper*”) to be “*configured to receive*” a monitored event such as a quality alert message (“*a call quality signal*”) from IP phone 20 (“*from the first VoIP telephone*”).

- g.** [1.5.2] *send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec; and*

259. Mundra, alone or in combination with RFC3435 and RFC3660, renders obvious that the call control gatekeeper is configured to send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec.

260. First, as discussed at limitations [1.2]-[1.3], Mundra’s first VoIP telephone 20 and second VoIP telephone 44 correspond to “*the first and second VoIP telephones.*” Further, as discussed at limitation [1.5.1], Mundra, alone or in combination with RFC3435 and RFC3660, renders obvious that Call Agent 42 is configured to receive a signal with call quality information (e.g., a user’s codec selection, or measurements of delay, jitter, or packet loss), which corresponds to “*the call quality signal.*”

261. Second, Mundra discloses “changing the first codec to the second codec in realtime during transmission of the voice call.” Ex.1008, claim 1. Mundra

repeatedly describes Call Agent 42 as being responsible for making a change of codecs, and doing so in response to information about a user's codec selection commands:

Dynamic codec changes are supported within gateways and call control protocols[.] Gateway 16 receives such user commands and takes appropriate action by **signaling agent 42** in the managed network 40 **to effect a change in the codec. Call agent module 42 is managed by the network provider to** receive commands from IP phone 20 or gateway 16 and the[n] **change the codecs in realtime during a call transmission.** Monitored network events and user commands are received by gateway 16, which then sends commands and call control protocols to **call controller agent 42 to cause a codec change.**

Ex.1008, [0020].

Gateway 16 detects network conditions affecting the call quality, and if a codec change is feasible to improve quality, signals an event to **the agent module to prepare both gateways for a potential change in codec.** A user could define protocols to automatically **have agent change the codec**, such as lowest cost, highest quality, or highest bandwidth.

Ex.1008, [0023].

262. In the above quotation immediately, Mundra references the Call Agent 42 acting “to prepare” for a change in codec. But Mundra describes the Call Agent 42 not merely preparing for a codec change, but causing the codec change to come about. *See, e.g.*, Ex.1008, [0020] (“controller agent 42 to cause a codec change”).

Thus, it would have been obvious to a POSITA from Mundra's overall disclosure that Call Agent 42 is responsible for effecting a codec change—an understanding further confirmed by additional disclosures in Mundra:

A user could normally subscribe to a policy from a network provider where lowering the cost could be the prime concern. In such case the **call agent would request additional events, thereby effecting the use of a low-cost codec** whenever feasible. A user may have a call wherein a higher quality transmission is desired, regardless of the cost considerations. For example, a user may desire a higher quality and higher cost call when speaking to a client or customer, but would settle for a lower-quality and lower cost call when making personal calls. Before or during call transmission, commands are sent from pre-assigned keys on a dialpad or directly from an end user PC to the gateway and **agent which switches event detection to enable the best, lowest, or midrange codec**, according to the category of classification the user desires, such as highest quality or lowest cost.

Ex.1008, [0025].

263. In the context of a VoIP telephone call between IP phones 20 and 44, it would have been understood by a POSITA that the codecs in operation would be on the IP phones 20 and 44, since that is where the audio information would be encoded and decoded. As I discussed previously, it would have been part of a POSITA's background knowledge that codecs execute on an IP phone. As a result, it would have been obvious to a POSITA for Call Agent 42 to effect a codec change

for an ongoing VoIP telephone call change by sending a message to IP phones 20 and 44, instructing them to change to the newly selected codec. Such a message is *“an adjustment signal... to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec.”* Call Agent 42 makes its codec change decisions based on information received about a user’s codec preferences (e.g., “according to the category of classification the user desires,” Ex.1008, [0025]), and hence, the message to change codecs is *“based on the call quality signal.”* Thus, Mundra renders obvious for Call Agent 42 (the “*call control gatekeeper*”) to *“send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec.”*

264. Third, to the extent that Mundra alone does not render this limitation obvious, it would have been obvious from the combination of Mundra with RFC3435 and RFC3660.

265. As discussed at limitation [1.5.1], it would have been obvious for Call Agent 42 to receive a “*call quality signal*” providing, for example, information about network conditions like delay or jitter. Consistent with Mundra’s explanation that it is the responsibility of “call controller agent 42 to cause a codec change” (Ex.1008, [0020]), it would have been obvious to a POSITA for Call Agent 42 to make the decision of when and whether to change codecs in response to the

network conditions. RFC3435 further supports that obviousness, as it states that in a VoIP telephony system, the “call control ‘intelligence’” resides with the call agent. Ex.1027, 5. RFC3435 further states that a Call Agent may send instructions for a codec change:

Once a connection has been established, the connection parameters can be modified at any time by a "modify connection" command. The **Call Agent may** for example instruct the gateway to **change the codec** used on a connection....

Ex.1008, 17.

266. The immediately above quotation discusses sending a codec change instruction to a gateway, but it would have been obvious to a POSITA for Call Agent to instead send the codec change instruction to IP phones 20 and 44. This is because, as I discussed previously, a POSITA would have understood that the codec implementation is on IP phones 20 and 44. For example, Mundra describes “codecs that are supported at sending and the receiving telephony devices.” Ex.1008, [0004]. It would have been obvious that the codec change instruction would be sent to both IP phones (not just one of them) because, as Mundra states, “the sending and receiving devices must use the same codecs during speech transmissions.” Ex.1008, [0009]. This overall understanding of a codec change process—that instructions to change codecs would be sent from a call agent to both IP phones—had been known in the field of VoIP telephony for years before the ’368 patent’s

filing date. *See, e.g.,* Ex.1075, [0013] (“With the MGCP-based call signaling, each endpoint is controlled by a Call Agent...and...two communicating endpoints will be notified (by their Call Agent) of the CODEC change...”). Thus, it would have been obvious to combine the features of Mundra, RFC3435, and RFC3660 as I have discussed, and a POSITA would have had an expectation of success in doing so, because others had already made VoIP telephony systems that performed codec changes in this way.

267. Thus, Mundra, RFC3435, and RFC3660 render obvious for Call Agent 42 (the “*call control gatekeeper*”) to “*send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec.*”

h. [1.6] *wherein the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones during a telephone call between the first and second VoIP telephones.*

268. Mundra, alone or in combination with RFC3435 and RFC3660, renders obvious that the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones during a telephone call between the first and second VoIP telephones.

269. First, as discussed at limitation [1.4], Mundra renders obvious a VoIP telephone call between IP phones 20 and 44 (“*a telephone call between the first and*

second VoIP telephones”), since those are the only phones in Mundra’s FIG. 2 and Mundra describes “a VoIP call.” Ex.1008, [0021].

270. Second, as discussed at limitation [1.5.2], Mundra, alone or in combination with RFC3435 and RFC3660, renders obvious Call Agent 42 responding to information from IP phone 20 (“*call quality signal*”) by sending a message to cause a change of codec (“*the adjustment signal*”) to IP phones 20 and 44 (“*to the first and second VoIP telephones*”).

271. Third, Mundra expressly teaches changing codecs during a VoIP telephone call:

A technique to **change a codec in realtime during transmission of a voice over packet call** placed over a packet switched network, such as the Internet, is described. The invention provides users real-time control over the cost and quality of a voice over packet call by monitoring the dynamically changing resources and network conditions on a packet network and allowing users to change codecs of the call transmission before and during the call according to user-defined specifications.

Ex.1008, Abstract.

...changing the first codec to the second codec in realtime during transmission of the voice call.

Ex.1008, claim 1.

...a call protocol agent that receives commands from each gateway and from each user, wherein **the call agent changes a first codec to a second codec in realtime during transmission of the call.**

Ex.1008, claim 5.

There is described herein a technique to lower the cost and improve the quality of voice calls over IP networks by providing users the ability to **select an appropriate codec during a voice over IP call.** The invention gives users real-time control over the cost and quality of the voice call by monitoring the dynamically changing resources and network conditions on an IP network and allowing users to select appropriate codecs before and during the call.

Ex.1008, [0016].

Call agent module 42 is managed by the network provider to receive commands from IP phone 20 or gateway 16 and the **change the codecs in realtime during a call transmission.**

Ex.1008, [0020].

272. As discussed at [1.5.2], a POSITA would have found it obvious for Mundra's Call Agent 42 to effect a codec change for an ongoing VoIP telephone call change by sending a message ("*the adjustment signal*") to IP phones 20 and 44. Thus, Mundra's further disclosure of changing codecs in realtime during a call renders obvious for "*the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones during a telephone call between the first and second VoIP telephones.*"

273. Fourth, as discussed at [1.5.2], the combination of Mundra and RFC3435 renders obvious for Call Agent 42 to cause a change of codecs (send a “*adjustment signal*”) based on information received about network conditions. RFC3435 further describes changing codecs after a telephone call has been established:

Once a connection has been established, the connection parameters can be modified at any time by a "modify connection" command. **The Call Agent may** for example instruct the gateway to **change the codec** used on a connection, or to modify the IP address and UDP port to which data should be sent, if a connection is "redirected".

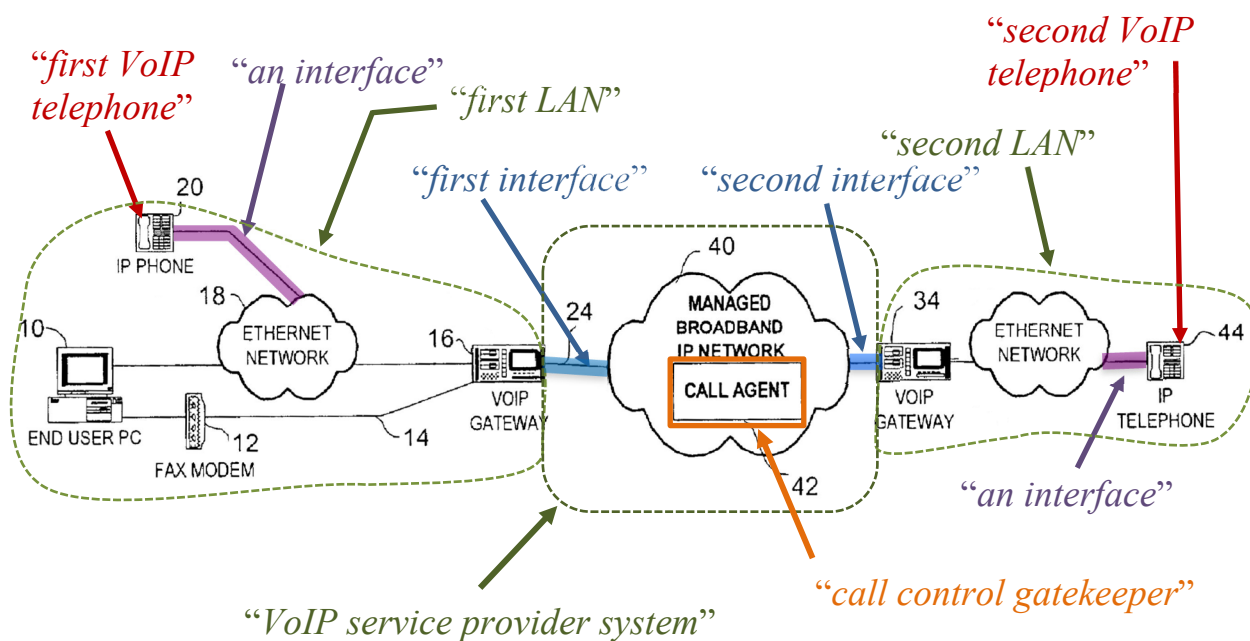
Ex.1027, 17.

274. The combination of Mundra and RFC3435 would have suggested to a POSITA for Call Agent 42 to send a “‘modify connection’ command... to change the codec used on a connection,” such as a VoIP telephone call between IP phones 20 and 44 (“*during a telephone call*”).

275. Thus, Mundra in combination with RFC3435 discloses that a command to change codecs, based on information received about network conditions, is sent to the first and second VoIP telephones 20, 44, during a realtime existing call between the first and second VoIP telephones 20, 44, which renders obvious that “*the adjustment signal based on the call quality signal is sent to the*

first and second VoIP telephones during a telephone call between the first and second VoIP telephones,” as claimed.

276. In summary, it would have been obvious to modify the system shown in Mundra’s Fig. 2 and to operate that system as recited in claim 1. For clarity, I have created below a modified version of Mundra’s Fig. 2 to show how all the recited claim components correspond to items in the figure. The modifications to the figure are those I discussed previously regarding limitations [1.2] and [1.3].



Ex.1008, FIG. 2 (modified to connect IP Phone 20 to Ethernet network 18 instead of VoIP Gateway 16 and add an Ethernet network between IP phone 44 and VoIP Gateway 34, annotated).

6. Claim 2

a. [2.0] *A system comprising:*

277. See analysis at limitation [1.0].

b. [2.1] *a VoIP service provider system having a call control gatekeeper, a first interface and a second interface;*

278. See analysis at limitation [1.1].

c. [2.2] *the first interface connected to a first LAN having an interface to a first VoIP telephone;*

279. See analysis at limitation [1.2].

d. [2.3] *the second interface connected to a second LAN having an interface to a second VoIP telephone;*

280. See analysis at limitation [1.3].

e. [2.4] *the first and second VoIP telephones being configured to communicate using a first codec;*

281. See analysis at limitation [1.4].

f. [2.5.1] *the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone and*

282. See analysis at limitation [1.5.1].

g. [2.5.2] *send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec; and*

283. See analysis at limitation [1.5.2].

h. [2.6] *wherein the call quality signal is based upon a number of dropped voice packets during a VoIP telephone call.*

284. The prior art renders obvious that the call quality signal is based upon a number of dropped voice packets during a VoIP telephone call.

285. As discussed at [1.5.1], Mundra, alone or in combination with RFC3435 and RFC3660 render obvious that Call Agent 42 receives information about network conditions, such as through a monitored event (*“the call quality signal”*). Mundra explains that information about network conditions includes information about “packet loss” in a “media stream.” Ex.1008, [0022]-[0023]. Mundra also describes “lost voice packets,” explaining how “call quality degrades... due to... lost packets.” Ex.1008, [0006], [0011]. Those of skill in the art used the terms “dropped” and “lost” as largely synonymous and interchangeable terms for packets that do not timely arrive. Ex.1091, 6:26-28 (“Packet loss is determined by measuring the number of packets that are dropped from the network over a specified time.”) Ex.1092, [0003] (in the context of “voice and multimedia data over IP,” “dropped packets are lost”); Ex.1093, [0005] (“lost information, in the form of dropped packets, is unacceptable to a business that wishes to provide its customers with Voice-over-IP (VoIP)”), Ex.1094, [0044] (“Dropped packets are determined by computing net lost packets”).

286. It would have been obvious to a POSITA for the Mundra’s monitored event to provide information about packet loss in a media stream such as a VoIP telephone call. *See* Ex.1008, [0021] (“monitor and detect events affecting a VoIP call”). Thus, Mundra renders obvious for a monitored event (“*the call quality signal*”) to be based on “*a number of dropped voice packets during a VoIP telephone call.*”

287. Alternatively, as discussed at [1.5.1], an MGCP quality alert as described by RFC3435 and RFC3660 corresponds to the claimed “*call quality signal.*” As noted in that earlier analysis, RFC3660 further explains that the quality alert is generated a packet loss rate exceeds a threshold value:

Quality alert (qa):

The packet loss rate or the combination of delay and jitter **exceeding a quality threshold.** The quality thresholds for delay, jitter, and packet loss rate are provisioned values.

Ex.1028, 46.

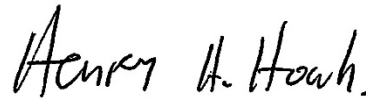
288. It would have been obvious to a POSITA for RFC3660’s “packet loss rate” to be based on a number of dropped packets. A POSITA also would have understood that the packet loss rate would usually be calculated as a fraction or ratio, with the number of lost packets in the numerator. For example, the packet loss rate would be the percentage of transmitted packets that become lost. *See, e.g.,* Ex.1095, 5:7-9 (calculating a “packet loss rate” by dividing the “**lost packet count**

N_{PL} ” by the “transmitted packet count N_{PT} ”); Ex.1096, [0145] (computing “the fraction lost (or packet-loss rate)”; Ex.1097, [0010]-[0011] (“the quality level can be defined by means of the packet loss rate (PLR). This can be computed by the sequence number information as[:] $QL=100*\text{Number of packets lost}/\text{Number of packets in transmission in a network element.}$ ”). As I’ve discussed at [1.4] and 1.6], Mundra renders obvious a “VoIP call” between IP phones 20 and 44. Ex.1008, [0021]. It would have been obvious for the “packet loss rate” that triggers generation of an MGCP quality alert to be calculated from the number of lost packets (“*a number of dropped voice packets*”) affecting a VoIP call between IP phones 20 and 44 (“*during a VoIP telephone call*”).

289. Thus, Mundra in combination with RFC3435 and RFC3660 renders obvious for a quality alert (“*the call quality signal*”) to be based on “*a number of dropped voice packets during a VoIP telephone call.*”

IX. DECLARATION

290. I declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and that these statements were made with knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under section 1001 of Title 18 of the United States Code.

A handwritten signature in black ink that reads "Henry H. Houh." The signature is written in a cursive, slightly slanted style.

Dated: August 11, 2024

Henry H. Houh, Ph.D.