

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

**PETITION FOR *INTER PARTES* REVIEW
UNDER 35 U.S.C. § 312 AND 37 C.F.R. § 42.104**

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Ex.1005	U.S. Patent No. 7,643,414 (“Minhazuddin”).
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Ex.1007	U.S. Patent No. 7,023,839 (“Shaffer”).
Ex.1008	U.S. Patent Pub. No. 2004/0032860 (“Mundra”).
Ex.1009	U.S. Patent No. 7,830,813 (“Lo”).
Ex.1010	Metro Ethernet Networks - A Technical Overview.
Ex.1011	Introduction to Cisco Network Design.
Ex.1012	Network Systems in CIM Environment.
Ex.1013	Avaya IP Office-Job Aid; Connecting an IP600 and IP Office via H323 IP Trunk.
Ex.1014	Avaya IP Office - Small Office Edition 2003.
Ex.1015	Merrill Lynch Selects Avaya IP Telephony Systems.
Ex.1016	Avaya IP Softphone Release 4 Ver. 1.
Ex.1017	Avaya Phone Manager 2.1.
Ex.1018	Avaya Definity® 4624 IP Telephone.
Ex.1019	Timothy V. Kelly, VOIP FOR DUMMIES (Wiley Publishing, Inc. 2005).
Ex.1020	U.S. Patent No. 7,657,021.

Ex.1021	Real-Time Transport Control Protocol (RTCP), Definition from TechTarget.
Ex.1022	Combining Voice over IP with Policy-Based Quality of Service, Extreme Networks TechBrief.
Ex.1023	Access Networks and Physical Media,
Ex.1024	Jari Turunen, et al., “Assessment of objective voice quality over best-effort networks,” 28 Computer Communications 582-588 (Elsevier 2005).
Ex.1025	U.S. Patent No. 7,764,679 (“MeLampy”)
Ex.1026	M. Arango, et al., “Media Gateway Control Protocol (MGCP) Version 1.0,” IETF Request for Comments 2705 (October 1999).
Ex.1027	F. Andreasen, et al., “Media Gateway Control Protocol (MGCP) Version 1.0,” IETF Request for Comments 3435 (January 2003).
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Ex.1036	U.S. Patent Pub. No. US2006/0291643 (“Pfaff”)
Ex.1037	U.S. Patent Pub. No. US2006/0221912 (“Olivier”)
Ex.1038	U.S. Patent No. 7,881,712 (“Luo”)
Ex.1039	U.S. Patent No. 7,715,359 (“Lin”)
Ex.1040	U.S. Patent Pub. No. US2005/0053098 (“Rabenko”)

Ex.1041	U.S. Patent Pub. No. US2004/0083128 (“Buckingham”)
Ex.1042	U.S. Patent Pub. No. US2003/0093563 (“Young”)
Ex.1043	U.S. Patent Pub. No. US2006/0037072 (“Rao”)
Ex.1044	U.S. Patent Pub. No. US2004/0114581 (“Hans”)
Ex.1045	U.S. Patent No. 6,480,122 (“Oddy”)
Ex.1046	U.S. Patent No. 6,898,188 (“Hamami”)
Ex.1047	U.S. Patent No. 6,904,038 (“Moon”)
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Ex.1049	U.S. Patent Pub. No. US2006/0233158 (“Croak”)
Ex.1050	U.S. Patent Pub. No. US2005/0053055 (“Horvath”)
Ex.1051	U.S. Patent Pub. No. US2005/0195949 (“Frattura”)
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Ex.1054	U.S. Patent Pub. No. US2004/0240432 (“Ying”)
Ex.1055	U.S. Patent Pub. No. US2004/0233898 (“Otsuka”)
Ex.1056	U.S. Patent Pub. No. US2002/0191557 (“Chow”)
Ex.1057	U.S. Patent Pub. No. US2003/0128670 (“Pinier”)
Ex.1058	U.S. Patent Pub. No. US2005/0015460 (“Gole”)
Ex.1059	U.S. Patent No. 6,757,363 (“Platt”)
Ex.1060	U.S. Patent Pub. No. US2002/0064178 (“Christie”)
Ex.1061	U.S. Patent No. 7,024,475 (“Abaye”)
Ex.1062	U.S. Patent Pub. No. US2006/0056630 (“Zimmer”)
Ex.1063	U.S. Patent Pub. No. US2005/0015430 (“Rothman”)
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Ex.1066	U.S. Patent Pub. No. US2004/0164619 (“Parker”)
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Ex.1071	U.S. Patent Pub. No. US2005/0257262 (“Matityahu”)
Ex.1072	U.S. Patent Pub. No. US2006/0198360 (“Biage”)
Ex.1073	U.S. Patent Pub. No. US2003/0086425 (“Bearden”)
Ex.1074	U.S. Patent Pub. No. US2003/0093513 (“Hicks”)
Ex.1075	U.S. Patent Pub. No. US2002/0094081 (“Medvinsky”)
Ex.1076	U.S. Patent No. 6,885,658 (“Ress”)
Ex.1077	U.S. Patent Pub. No. US2002/0101881 (“Sundaresan”)
Ex.1078	U.S. Patent Pub. No. US2004/0001493 (“Cloonan”)
Ex.1079	U.S. Patent Pub. No. US2001/0053208 (“Pinard”)
Ex.1080	U.S. Patent Pub. No. US2004/0071129 (“Doerr”)
Ex.1081	U.S. Patent Pub. No. US2003/0161453 (“Veschi”)
Ex.1082	U.S. Patent Pub. No. US2007/0133515 (“Kumar”)
Ex.1083	U.S. Patent No. 6,873,604 (“Surazski”)
Ex.1084	Robert M. Gray, “The 1974 Origins of VoIP”, IEEE Signal Processing Magazine (July 2005)
Ex.1085	H. Schulzrinne, “Voice Communication Across the Internet: A Network Voice Terminal”, CS Technical Report 92-50 (July 1992)
Ex.1086	University of Southern California 1982 Annual Technical Report, Information Sciences Institute (July 1981 – June 1982)
Ex.1087	Clifford J. Weinstein and James W. Forgie, “Experience with Speech Communication in Packet Networks”, IEEE Journal on Selected Areas in Communications, Vol. SAC-1, No. 6 (December 1983)
Ex.1088	U.S. Patent Pub. No. US2001/0055331 (“Agazzi”)
Ex.1089	U.S. Patent Pub. No. US2005/0084269 (“Dallesasse”)
Ex.1090	Unopposed Motion for Leave to Amend Infringement Contentions - <i>Croga Innovations Ltd. v. Cisco Systems, Inc.</i> , 2:24-cv-00065, Dkt. 52 (E.D. Tex. July 16, 2024).
Ex.1091	U.S. Patent No. 6,934,258 (“Smith”)

Ex.1092	U.S. Patent Pub. No. US2004/0215812 (“Lu”)
Ex.1093	U.S. Patent Pub. No. US2004/0061505 (“Fitzpatrick”)
Ex.1094	U.S. Patent Pub. No. US2003/0179761 (“Dobbins”)
Ex.1095	U.S. Patent No. US6,914,900 (“Komatsu”)
Ex.1096	U.S. Patent Pub. No. US2004/0252701 (“Anandakumar”)
Ex.1097	U.S. Patent Pub. No. US2003/0063569 (“Kalliokulju”)
Ex.1098	S. Bradner, “The Internet Standards Process - - Revision 3,” IETF Request for Comments 2026 (October 1996)
Ex.1099	F. Andreasen, et al., “Media Gateway Control Protocol (MGCP) Version 1.0,” IETF Request for Comments 3435 (January 2003) (as captured by the Wayback Machine on March 10, 2003, http://web.archive.org/web/20030310204806if_/http://www.ietf.org/rfc/rfc3435.txt).
Ex.1100	B. Foster, et al., “Basic Media Gateway Control Protocol (MGCP) Packages,” IETF Request for Comments 3660 (December 2003) (as captured by the Wayback Machine on March 2, 2004, http://web.archive.org/web/20040302145853if_/http://www.ietf.org/rfc/rfc3660.txt).
Ex.1101	“RFC Database”, RFC Editor, IEFT (March 7, 2002) (as captured by the Wayback Machine on June 2, 2004, http://web.archive.org/web/20040602210051if_/http://www.rfc-editor.org/rfc.html).
Ex.1102	“RFC Index”, IETF (Created March 31, 2004) (as captured by the Wayback Machine on April 2, 2004, http://web.archive.org/web/20040402101532if_/http://www.ietf.org/iesg/1rfc_index.txt)
Ex.1103	“VoIP for Dummies” on dummies.com (as captured by the Wayback Machine on December 14, 2005, https://web.archive.org/web/20051214151026/http://www.dummies.com/WileyCDA/DummiesTitle/productCd-0764588435.html).
Ex.1104	“Everyday Computing” on dummies.com (as captured by the Wayback Machine on September 10, 2005, https://web.archive.org/web/20050910152724/http://www.dummies.com/WileyCDA/Section/id-100004.html).

Ex.1105	“Everyday Computing” on dummies.com (as captured by the Wayback Machine on November 4, 2005, https://web.archive.org/web/20051104050929/http://www.dummies.com/WileyCDA/Section/id-100004.html).
Ex.1106	“Using the Wayback Machine,” https://help.archive.org/help/using-the-wayback-machine/ .

I. INTRODUCTION

Pursuant to 35 U.S.C. §§ 311, 314(a), and 37 C.F.R. § 42.100, Cisco Systems, Inc. (“Petitioner”) respectfully requests that the Board review and cancel as unpatentable under (pre-AIA) 35 U.S.C. §103(a) claims 1 and 2 (the “Challenged Claims”) of U.S. Patent No. 7,738,368 (“’368 patent,” Ex.1001).

The ’368 patent relates to voice over internet protocol (“VoIP”) telephony. The patent describes determining call quality and, when the call quality passes a threshold, instructing VoIP phones on a telephone call to use a different audio encoder-decoder (or “codec”). Ex.1001, 3:25-35, 3:58-65.

These ideas were not new. When the ’368 patent application was filed, VoIP technology had been developing for decades, including techniques for monitoring VoIP call quality and responsively changing codecs. Ex.1003, ¶¶35-37. For example, it was known in the art to compare a “Quality of Service” metric to a threshold and, if appropriate, “change an existing live voice communication from the first codec to the second codec.” Ex.1005, Abstract. Other prior art described a “technique to change a codec in realtime during transmission of a voice over packet call” by “monitoring the dynamically changing resources and network conditions.” Ex.1008, Abstract.

Accordingly, Petitioner respectfully requests that the Board review and cancel as unpatentable the Challenged Claims of the ’368 patent.

II. GROUNDS FOR STANDING

Petitioner certifies that the '368 patent is eligible for IPR and that Petitioner is not barred or estopped from requesting IPR challenging the patent claims. 37 C.F.R. § 42.104(a).

III. NOTE

Petitioner cites to exhibits' original page numbers whenever feasible. Emphasis in quoted material has been added.

IV. SUMMARY OF THE '368 PATENT

A. Overview of the '368 patent

The '368 patent “generally relates to voice over internet protocol (‘VoIP’) telephone systems” and describes a system for changing the coder-decoder (“codec”) of a VoIP telephone call to a lower bandwidth codec. Ex.1001, 1:6-8, 1:50-2:29, Abstract.

With reference to Fig. 1 shown below, the '368 patent describes a system that includes a wide area network (“WAN”) 16 connected to two local area networks (“LANs”) 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.

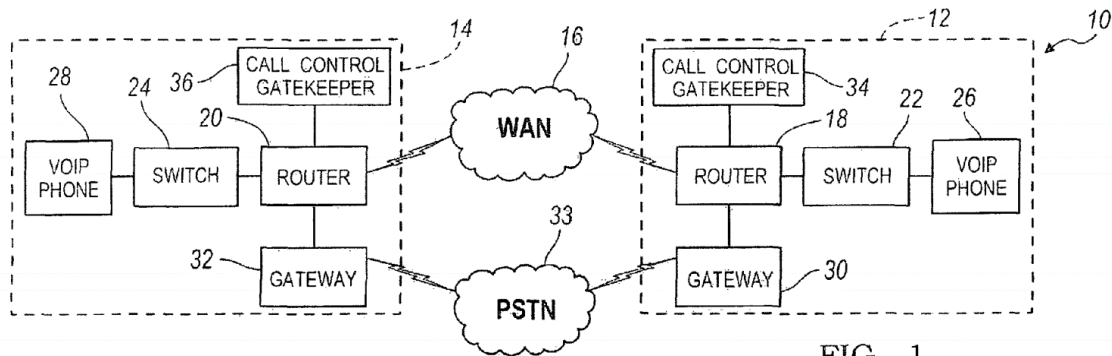


FIG. 1

Ex.1001, FIG. 1.

The '368 patent provides two embodiments for adjusting the codecs during a VoIP call. In a first embodiment, the VoIP telephones determine that call quality is low and change to a lower bandwidth codec. Ex.1001, 3:25-38, 4:42-46.

Alternatively, 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. Ex.1001, 3:58-65, 4:47-55. The Challenged Claims relate to the latter technique.

B. Prosecution History of the '368 Patent.

The '368 patent application was filed on November 10, 2005. Ex.1001, INID code (22). The Office issued a first Office Action rejecting independent claims 1, 8, and 15 and indicating that remaining claims 2-7, 9-14, and 16-18, were allowable if rewritten in independent form. Ex.1002, 52-65. In response, Applicants canceled rejected independent claims and amended the dependent

claims to be written in independent form. Ex.1002, 66-77. The Office then indicated that claims 2-7 and 9-14 were allowed and rejected 16-18 claims as anticipated. Ex.1002, 90-93. The Applicants canceled the rejected claims, and the allowed claims issued as claims 1-12. Ex.1002, 102-110, 117-120; Ex.1003, ¶¶57-58.

V. LEVEL OF ORDINARY SKILL IN THE ART

A Person of Ordinary Skill in The Art (“POSITA”) on November 10, 2005 would have been someone knowledgeable and familiar with the then-extant packet-based communication network techniques and VoIP technologies. Ex.1003, ¶¶27, 34-50. That person 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. Lack of work experience can be remedied by additional education, and vice versa. Ex.1003, ¶¶26-28.

VI. CLAIM CONSTRUCTION

Claim terms in IPR are construed according to their “ordinary and customary meaning” to those of skill in the art. 37 C.F.R. § 42.100(b). *Phillips v. AWH Corp.*, 415 F.3d 1303, 1313 (Fed. Cir. 2005) (en banc). Petitioner submits that, for the purposes of this proceeding and the grounds presented herein, no claim term requires express construction. *Nidec Motor Corp. v. Zhongshan Broad Ocean*

Motor Co., 868 F.3d 1013, 1017 (Fed. Cir. 2017). Ex.1003, ¶¶60-61.

VII. RELIEF REQUESTED AND REASONS FOR THE REQUESTED RELIEF

Petitioner asks that the Board institute a trial for *inter partes* review and cancel the Challenged Claims in view of the analysis below.

VIII. DISCRETIONARY DENIAL WOULD BE INAPPROPRIATE

A. Discretionary denial under *Fintiv* is not appropriate.

The six factors considered for § 314 discretionary denial strongly favor institution given Petitioner’s timely filing and the compelling merits of this case. *See Apple Inc. v. Fintiv, Inc.*, IPR2020-00019, Paper 11 (PTAB Mar. 20, 2020) (precedential).

1. No evidence regarding a stay

No motion to stay has been filed, so the Board should not infer the outcome of such a motion. *Sand Revolution II LLC v. Continental Intermodal Group – Trucking LLC*, IPR2019-01393, Paper 24 at 7 (PTAB June 16, 2020) (informative); *see also Dish Network L.L.C. v. Broadband iTV, Inc.*, IPR2020-01359, Paper 15 (Feb. 12, 2021) (“It would be improper to speculate, at this stage, what the Texas court might do regarding a motion to stay...”). Thus, this factor is neutral on discretionary denial.

2. Parallel proceeding trial date

The co-pending district court case was filed in the Eastern District of Texas on February 1, 2024. Ex.1029. The most recent statistics currently available for the Eastern District of Texas show a median time-to-trial of 21.6 months. Ex.1031, 35. Thus, the estimated trial date is late December 2025.¹ The Board’s final written decision is expected soon after, in early February 2026. Given the proximity of the Board’s expected Final Written Decision and the estimated trial date, this factor is at least neutral. *See e.g., Cisco Systems Inc., v. Video Solutions*, IPR2024-00194 (PTAB March 28, 2024) (granting institution and finding factor two neutral where “trial is likely to begin within approximately one month of [the Board’s] final written decision”).

¹ The docket control order sets the case for trial on October 20, 2025. Ex.1030, 1. However, “a court’s general ability to set a fast-paced schedule is not particularly relevant...where, like here, the forum itself has not historically resolved cases so quickly.” *In re Apple Inc.*, 979 F.3d 1332, 1344 (Fed. Cir. 2020). Notably, the court already has five other patent trials—plus four other trials—scheduled to begin October 20, 2025. *See* Ex.1032.

3. Investment in the parallel proceeding

The co-pending litigation is in its early stages, with major activities including a *Markman* hearing and the close of discovery scheduled after the Board’s expected decision on institution. *See* Ex.1030, 4 (Claim Construction hearing on April 30, 2025; close of fact discovery on May 23, 2025; close of expert discovery on July 8, 2025). Moreover, Petitioner has acted with diligence to prepare this Petition approximately six months after Petitioner was served with a complaint, and within weeks of receiving revised infringement contentions (identifying the asserted claims). Ex.1090, 2. Under *Fintiv*, Petitioner’s prompt filing “weigh[s] against exercising the authority to deny institution.” *Fintiv*, Paper 11 at 11 (“If the evidence shows that the petitioner filed the petition expeditiously, such as promptly after becoming aware of the claims being asserted, this fact has weighed against exercising the authority to deny institution under NHK”).

4. Overlapping issues with the parallel proceeding

Claim 2, challenged here, is not asserted in the district court litigation. Petitioner also offers the following stipulation: Petitioner stipulates that if trial is instituted, it will not pursue the same grounds in the district court litigation. *See Sand Revolution II, LLC v. Cont’l Intermodal Group-Trucking LLC*, IPR2019-01393, Paper 24 at 12 (PTAB June 16, 2020). Consequently, this factor favors institution.

5. Identity of parties

Petitioner is a defendant in the litigation. That is true of most petitioners in IPR proceedings. Accordingly, this factor should not be a basis for denying institution.

6. Other circumstances

The prior art presented in this Petition presents a compelling case on the merits. The claims of the '368 patent were found to recite allowable subject matter in the first Office action. Ex.1002, 65. In the same action, the prior art cited by the Examiner lacked a “call control gatekeeper configured to receive a call quality signal from [sic] 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.” Ex.1002, 54. As explained in detail below, these ideas (and the remaining claim limitations) were disclosed in other prior art not reviewed by the Examiner.

The Board’s current policy is to “not deny institution of an IPR or PGR under Fintiv (i) when a petition presents compelling evidence of unpatentability.” Director’s Memorandum dated June 21, 2022 (“Memo”), 9. “Compelling, meritorious challenges are those in which the evidence, if unrebutted in trial, would plainly lead to a conclusion that one or more claims are unpatentable by a preponderance of the evidence.” Memo, 4. Here, the Petition plainly shows that the

concepts of monitoring VOIP call quality and responsively changing to a second codec were disclosed multiple times in the prior art.

Thus, the evidence of unpatentability is compelling here, and the PTAB should not deny institution under *Fintiv*.

B. Discretionary denial under 35 U.S.C. § 325(d) is not appropriate.

Denial under § 325(d) is not warranted because the challenges presented in this petition are neither cumulative nor redundant to the prosecution of the '368 patent. The Examiner did not consider any of the references relied upon in the grounds of unpatentability presented below.² Moreover, the challenges in this petition are non-cumulative because the prior art presented here describes the claim limitations that were the reasons for allowance. *See* Ex.1002, 52-65. Accordingly, the Board should not exercise its discretion to deny this petition under § 325(d).

C. Discretionary denial under *General Plastic* is not appropriate.

The '368 patent has not been challenged in any prior IPR petition, so none of

² And of the additional references cited to confirm the background knowledge of a POSITA, only Ex.1097 (“Kalliokulju”) was discussed in prosecution. *See* Ex.1002, 54-55. Petitioner cites it here for a different reason than the Examiner, namely, to show that a packet loss rate depends on the number of lost packets. *See infra*, Ground 2 analysis of limitation [2.6].

General Plastic discretionary institution factors apply to this Petition. *See General Plastic Indus. Co., Ltd. v. Canon Kabushiki Kaisha*, IPR2016-01357, Paper 19 at 16 (PTAB Sept. 6, 2016) (Section II.B.4.i. precedential).

IX. IDENTIFICATION OF HOW THE CLAIMS ARE UNPATENTABLE

A. Challenged Claims and Statutory Grounds for Challenges³

Grounds	Claims	Basis: 35 U.S.C. § 103 (Pre-AIA) over
#1	1 and 2	Minhazuddin, Hepworth, and Shaffer
#2	1 and 2	Mundra, RFC3435, and RFC3660

U.S. Patent No. 7,643,414 (Ex.1005, “Minhazuddin”) was filed on 2/10/2004, issued on 1/5/2010, and is prior art under (pre-AIA) 35 U.S.C. 102(e).

U.S. Patent Publication No. 2004/0073690 (Ex.1006, “Hepworth”) was filed on 9/30/2002, published on 4/15/2004, and is prior art under (pre-AIA) 35 U.S.C. 102(b).

U.S. Patent No. 7,023,839 (Ex.1007, “Shaffer”) was filed on 8/19/1999, issued on 4/4/2006, and is prior art under (pre-AIA) 35 U.S.C. 102(e).

³ Petitioner relies on the teachings, and not on a physical incorporation of elements. *See In re Mouttet*, 686 F.3d 1322, 1332 (Fed. Cir. 2012); *In re Etter*, 756 F.2d 852, 859 (Fed. Cir. 1985); Ex.1003, ¶64.

U.S. Patent Publication No. 2004/0032860 (Ex.1008, “Mundra”) was filed on 8/19/2002, published on 2/19/2004, and is prior art under (pre-AIA) 35 U.S.C. 102(b).

F. Andreasen, et al., “Media Gateway Control Protocol (MGCP) Version 1.0,” Request for Comments 3435 (Ex.1027, “RFC3435”) was published by the Internet Engineering Task Force, a division of The Internet Society, in January 2003. RFC3435 is prior art under (pre-AIA) 35 U.S.C. 102(b).

B. Foster, et al., “Basic Media Gateway Control Protocol (MGCP) Packages,” Request for Comments 3660 (Ex.1028, “RFC3660”) was published by the Internet Engineering Task Force, a division of The Internet Society, in December 2003. RFC3660 is prior art under (pre-AIA) 35 U.S.C. 102(b).

IETF Request for Comments documents, such as RFC3435 and RFC3660, are printed publications. *See, e.g.*, Ex.1098, 6, (“RFCs can be obtained from a number of Internet hosts...”), 8 (IETF documents are “readily available to a wide audience”), 26 (IETF participants “shall publicly announce...every activity” relating to the standardization process); Ex.1003, ¶65. The Board has repeatedly found IETF documents, including RFCs, to be “printed publications.” *See, e.g., Apple, Inc. v. VirnetX, Inc.*, IPR2017-00337, Paper 31, 46-47 (May 30, 2018) (RFCs are “precisely the type of documents whose main purpose is for public disclosure”); *Riot Games, Inc. v. Paltalk Holdings, Inc.*, IPR2018-00130, Paper 11,

30-33 (May 15, 2018) (RFCs are printed publications). Proof of RFC3435’s public accessibility was captured by the Internet Archive’s Wayback Machine on March 10, 2003. Ex.1099; *see also* Ex.1106 (the “date code embedded in the archived url” is the date a webpage was archived, formatted as “yyyymmddhhmmss” (year, month, day, hour, minute, second)). Proof of RFC3660’s public accessibility was captured by the Internet Archive’s Wayback Machine on March 2, 2004. Ex.1100. Both documents were indexed on the IETF’s website no later than April 2, 2004. Ex.1102, 204, 219. RFC3435 and RFC3660 would have been locatable via keyword search on IETF’s RFC Editor website. Ex.1101. RFC3435 was repeatedly discussed in prior art patent applications, confirming that it was available and known to those of skill in the art. *See, e.g.*, Ex.1033, [0005]; Ex.1034, [0039]; Ex.1035, [0039]; Ex.1036, [0016]; Ex.1037, [0056]. RFC3660 was also discussed in prior art patents, confirming that it was available and known to those of skill in the art. *See* Ex.1038, 6:38-41; Ex.1039, 4:49-53. Thus, RFC3435 and RFC3660 are prior art printed publications under (pre-AIA) 35 U.S.C. 102(b).

Petitioner’s analysis cites additional prior art to support assertions regarding what a POSITA would have known or understood from the asserted prior art. *See Yeda Research v. Mylan Pharm. Inc.*, 906 F.3d 1031, 1041-1042 (Fed. Cir. 2018); 37 C.F.R. § 42.104(b); *see also K/S HIMPP v. Hear-Wear Techs., LLC*, 751 F.3d 1362, 1365-66 (Fed. Cir. 2014); *Arendi S.A.R.L. v. Apple Inc.*, 832 F.3d 1355,

(Fed. Cir. 2016).

B. Statement of Material Facts

- 1) As of the 2005 priority date, there were many models of VoIP phones commercially available. Ex.1003, ¶¶37-38; Ex.1019, 142 (“Many makes and models of VoIP telephones are available.”); Ex.1040, [0014]; Ex.1041, [0021]; Ex.1042, [0091]; *see also* Ex.1103 (showing Ex.1019 published “September 2005”); Ex.1104 (showing Ex.1019 as “Just Published” on 9/10/2005); Ex.1105 (offering Ex.1019 for sale on 11/5/2005).
- 2) The available VoIP phones included “hard” phones, which were physical devices that looked and operated much like a traditional desk telephone, and “soft” phones, which were software implementations that enabled a network-connected computer to make and receive voice calls. Ex.1003, ¶39; Ex.1019, 142-149; Ex.1005, 4:64-5:4; Ex.1043, [0140], Fig. 2A (showing “hard IP phone” 210a and “soft IP phone” 210b).
- 3) A typical VoIP hard phone included an RJ-45 connection port for coupling the VoIP hard phone to a LAN, such as an Ethernet network. Ex.1003, ¶¶40-41; 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.”), 27 (Fig. 2-3), 38

(Fig. 2-10), 74 (“VoIP runs on Ethernet LANs...”), 142 (“The RJ-45 connector on a [VoIP] hard phone is an Ethernet port used to connect the phone to your network.”); Ex.1044, [0018] (a “VoIP communicator” has “a RJ-45 interface 120 for the VoIP network”); 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); Ex.1045, 1:20-24 (“Ethernet LAN connected telephones (often referred to as IP Phones)” connect via “a standard RJ-45 jack”); Ex.1046, 2:5-6 (“Ethernet phone... plugs into an Ethernet RJ-45 jack”); Ex.1047, Fig. 1, 4:62-63 (showing “IP telephones and other IP telephony devices 52” connected to “LAN 50 [that] uses a communication protocol (such as Ethernet)”).

- 4) Ethernet networks were common and typically included one or more Ethernet switches, each of which would have multiple RJ-45 connection ports for coupling to other network devices. Ex.1003, ¶42; Ex.1010, 3 (“Ethernet has been the dominant technology in enterprise and campus LANs for many years.”); Ex.1088, [0005] (Ethernet is “the most pervasive” LAN technology); Ex.1011, 3 (“Ethernet switching provides Layer 2

switching and offers dedicated Ethernet segments for each connection. This is the base fabric of the network.”); Ex.1048, Fig. 1A, 2:63-3:8 (“ethernet switch” with “industry standard RJ-45 electrical connectors”).

- 5) The terms “IP phone,” “digital VoIP telephone,” and “IP telephony device” are substantially synonymous. Ex.1006, [0036], Fig. 1 (showing a “VoIP architecture” with “IP hard-phones” and “IP soft-phones”); Ex.1008, [0017]-[0020], Fig. 2 (referring to “digital VoIP telephone 20,” “IP telephony device 20,” and “IP phone 20”).
- 6) An IP phone is a VoIP endpoint. Ex.1003, ¶46; Ex.1006, [0036] (“end points can be...an IP hard phone”); Ex.1049, Fig.2, [0007] (depicting “VoIP endpoints” as telephones); Ex.1050, [0025] (“VoIP endpoints may include... IP phones...”); Ex.1051, [0003] (“VoIP endpoint devices (e.g., telephones...)”), [0022].
- 7) IP phones implemented one or more audio compression algorithms, also known as codecs. Ex.1003, ¶¶36, 47. Ex.1008, [0004] (“Devices that perform speech digitization are called ‘codecs’, for coder/decoder”), [0009] (“....codecs that are supported at sending and the receiving telephony devices...”); Ex.1026, 54 (“Compression Algorithm: a list of supported codecs”); Ex.1052, [0071] (“IP phone 101 settings such as:... codecs”); Ex.1053, [0020]; Ex.1054, [0025], Fig. 2 (“VoIP telephone apparatus 201

includes... encoder and decoder (codec) module 204.”); Ex.1055, [0063] (“With IP phones,... audio processing is performed by a CODEC unit”); Ex.1056, [0044] (“...IP telephone with digital processing capability to support multiple codecs...”); Ex.1057, [0003] (“When multiple coding algorithms have been embedded inside the user equipment like the IP phone....”).

8) A network communication link between two devices required each device to have an interface for sending data, receiving data, or both. Ex.1003, ¶50; 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...”); *see also* Ex.1014, 44 (disclosing standard interfaces); Ex.1020, 4:55-63 (network devices “...conventionally include[] interfaces to external communication links...”); Ex.1058, [0026] (“Each system attached to the network cloud 102 includes an appropriate conventional network interface arrangement (not shown) for communicating over the network 102.”); Ex.1059, 5:56-59 (“telephone 21 also includes a network interface circuit 142, which interfaces... to the Ethernet network”); Ex.1056, [0044].

9) Network engineers commonly used a “cloud” to depict a network comprised of various interconnected communication components, such as routers,

switches, servers, and cables. Ex.1003, ¶49. Ex.1060, [0040] (“broadband network cloud that represents any number of network elements such as switches, enhanced platforms, and servers”); Ex.1061, 4:5-11 (“...‘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.1062, [0054] (“...network infrastructure is depicted in the conventional manner using network clouds.”); Ex.1063, [0096] (“Ethernet cabling (depicted as a network cloud 1104)”).

10) “It was common to connect multiple communications devices, including IP telephones, to a LAN, especially in the context of a corporate network.” Ex.1003, ¶¶43-44; Ex.1019, 125 (Fig. 8-1 showing multiple IP phones connected to an Ethernet LAN); Ex.1064, Fig.2; Ex.1083, Fig.1, 1:39-42 (“corporate Ethernet LAN, to which a plurality of IP telephones 106 may be operatively coupled... using hardware and software standard in the art”); Ex.1059, Fig. 1, 1:40-66 (Ethernet LAN 12 including telephones 21-26).

11) It was common for IP phones to receive both power and data from an Ethernet switch using “Power-over-Ethernet” technology—including the IEEE802.3af standard—thereby providing benefits including (1) eliminating the need for a power outlet at each IP phone’s location, (2) allowing a single

backup power source located at the Ethernet switch to protect the telephone network against power outages, and (3) lowering costs. Ex.1003, ¶45.

Ex.1065, Fig. 8, [0012] (“power over Ethernet (to power the phone)”), [0055] (“In an electrical outage... voice services are maintained for emergency situations”); Ex.1066, [0019] (“a switch, which is configured to at least provide power to...Internet Protocol (IP) phones”); Ex.1067, [0004], (“the IEEE 802.3af standard identifies ways to deliver... AC power...to a variety of Ethernet devices or terminals such as IP phones,...”); Ex.1068, [0004]-[0006] (“IEEE Std 802.3af... provides an integrated power and data network connection... removes the need for independent power... [and] ultimately lowers the cost of attaching a device to the LAN.”); Ex.1069, [0041] (“This eliminates the need for client devices 12-16 to have separate power connections 18 to electrical outlets...”); Ex.1070, [0002] (“It is desirable in today's marketplace, given the building infrastructure, to provide power over ethernet cable....”); Ex.1071, [0026] (“The most common forms of connections are either the 10-100 megabyte Ethernet..., or follow the power-over-Ethernet, 802.11AF standards.”).

- 12) “The quality of a VoIP call was commonly assessed based on measurements of packet delay, number of lost packets, and jitter.” Ex.1003, ¶48; Ex.1006, [0006] (“performance information governing the

communication link (e.g., key statistics about the media packets being sent and received by each end point such as...packet loss” and “a number of dropped packets.”); Ex.1072, Abstract (“The quality of a VoIP call is assesse[d] by... conventional IP packet quality measures..., for example, jitter, packet loss and packet delay.”); Ex.1073, [0041] (“suitable packet delay characteristics (as well as jitter and packet loss) prove to be of special importance in successful implementation of VoIP applications”); Ex.1074, [0009] (“take into account network attributes, such as codec, jitter buffer, delay and packet loss and model how these affect call quality scores”); Ex.1024, Abstract.

C. Ground 1: Claims 1 and 2 are Obvious over Minhazuddin, Hepworth, and Shaffer.

1. Minhazuddin

Minhazuddin describes a VoIP system, shown in Fig. 1 below. Ex.1005, 1:5-8, 3:16-65, Abstract.

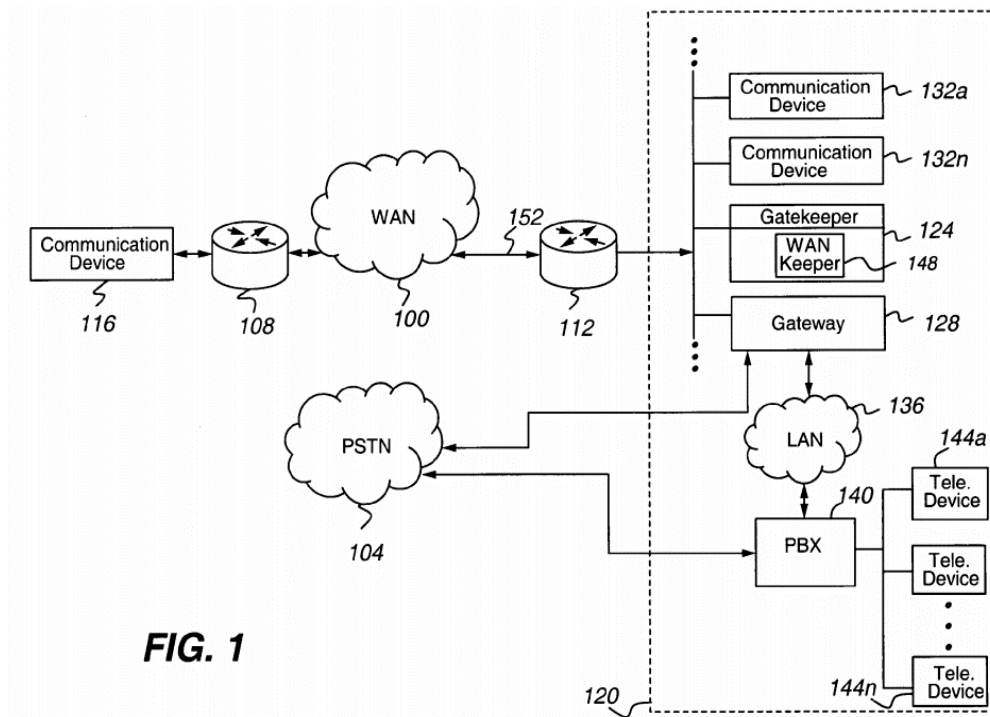
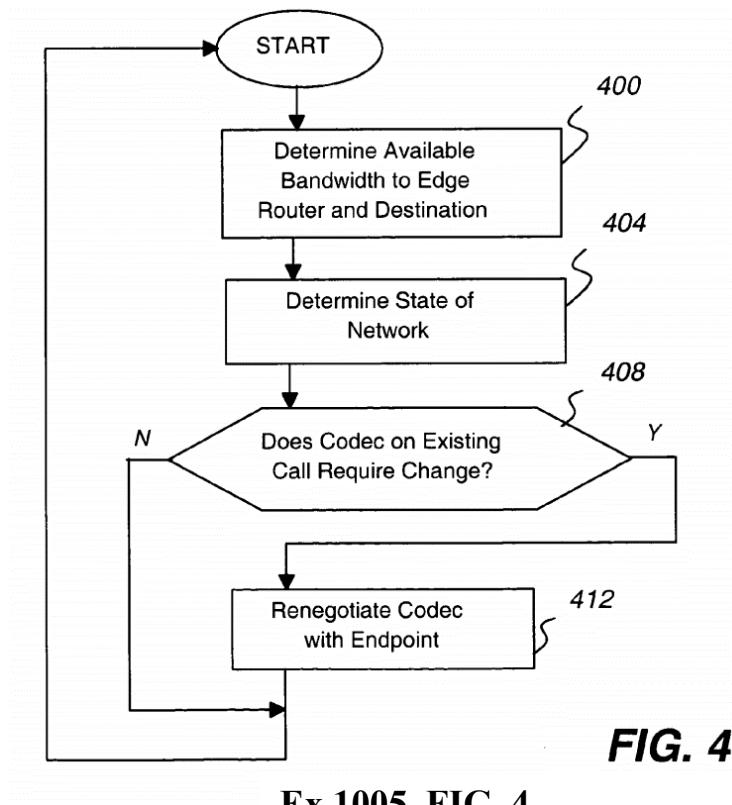


FIG. 1

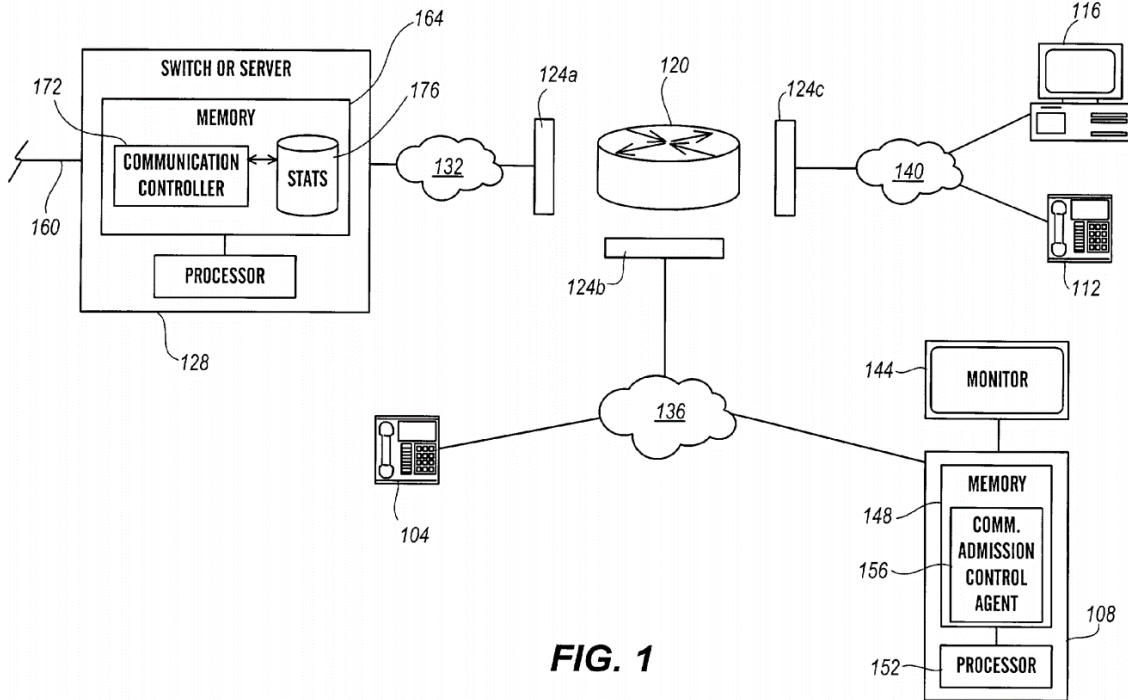
Ex.1005, FIG. 1.

Minhazuddin's communication devices 116, 132a-n initially communicate using a first selected codec. Ex.1005, 7:52-63, FIG. 2. During operation, gatekeeper 124's WAN keeper 148 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. Ex.1005, 8:23-52; 3:55-57; Ex.1003, ¶¶68-70.



2. Hepworth

Hepworth also relates to VoIP systems. Ex.1006, [0002]-[0003]. In the context of Fig. 1, Hepworth describes using router interfaces 124a-c for connecting to a network (e.g., LAN) and VoIP endpoints. Ex.1006, [0036], [0045], FIG. 1.



Ex.1006, FIG. 1.

Hepworth further discloses that each VoIP telephone endpoint measures and collects QoS information (e.g., key statistics such as jitter, packet loss, round-trip time, number of dropped packets, bandwidth utilization, etc.) of a VoIP telephone call. Ex.1006, [0006], [0021], [0044]. Based on the collected QoS information, the VoIP telephone endpoints provide decentralized call admission control (e.g., including codec renegotiation). Ex.1006, [0010], [0041]. Hepworth also contemplates a centralized session monitor (e.g., Avaya VoIP Monitoring Manager™) 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. Ex.1006, [0006], [0053]; Ex.1003, ¶¶71-74.

3. Shaffer

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. In the context of Figure 2, Shaffer discloses using a network interface that connects a telephony device to a LAN to facilitate telephony-over-LAN communications. Ex.1007, 3:14-27, FIGS. 1, 2.

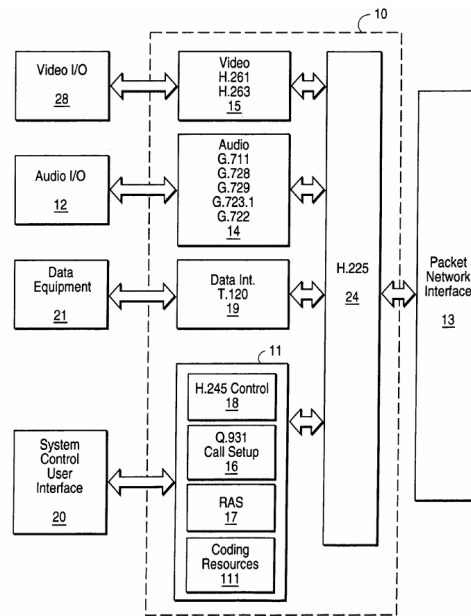


FIG. 2

Ex.1007, FIG. 2.

Shaffer further teaches that a gatekeeper includes a bandwidth allocation server (“BWAS”) for monitoring the network and if a threshold is met, the BWAS sends a signal to active telephony devices to renegotiate their codecs for existing connections. Ex.1007, 1:51-2:5, 3:4-12, 8:49-67, 9:43-46, 10:17-32, 11:20-42. An

exemplary process for renegotiating codecs is illustrated at Figure 9; Ex.1003,

¶¶75-78.

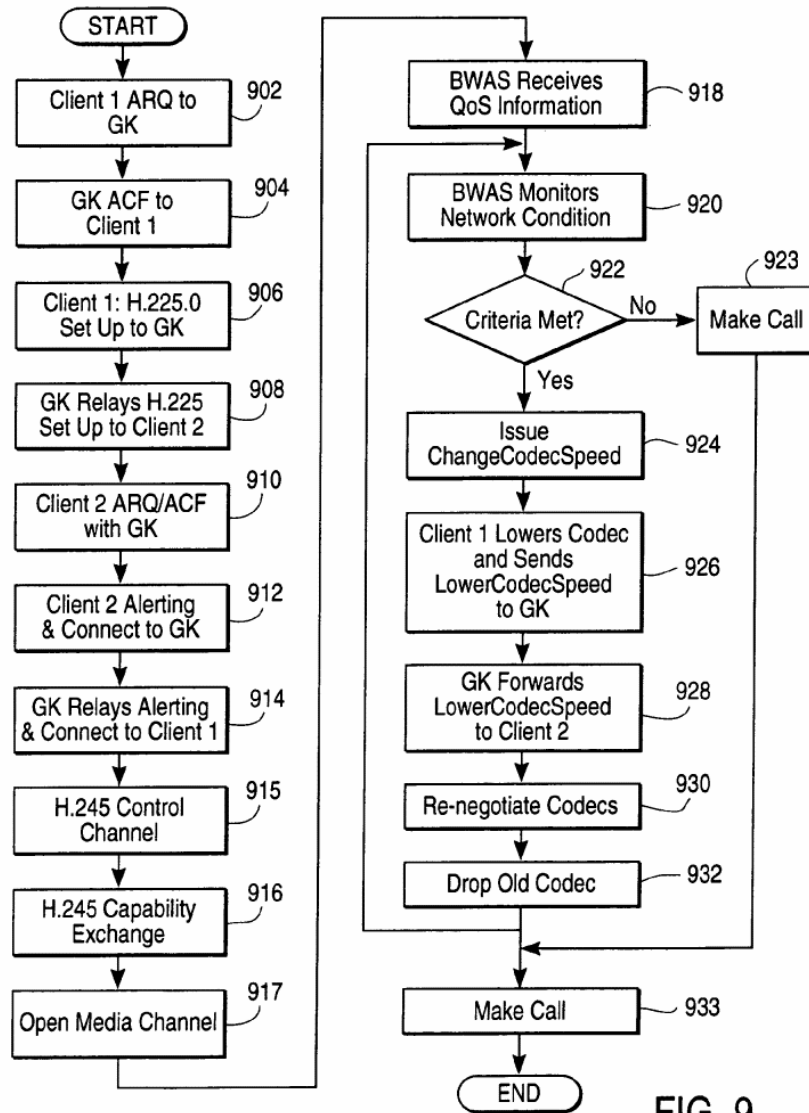


FIG. 9

Ex.1007, FIG. 9.

4. Reasons to Combine Hepworth with Minhazuddin.

a. Hepworth is Analogous Art

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. Ex.1003, ¶79.

b. Motivation to combine with Hepworth

“A POSITA would have been motivated to combine the teachings of Hepworth and Minhazuddin to produce numerous predictable and beneficial results.” Ex.1003, ¶80.

i. An interface

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 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. As Dr. Houh testifies:

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.

Ex.1003, ¶81.

Thus, the proposed combination is merely “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).”

Ex.1003, ¶82.

ii. Receive a call quality signal

Minhazuddin's 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.” Ex.1005, 5:53-64.

Hepworth teaches a well-known and suitable technique for monitoring QoS (e.g., jitter, packet loss, number of dropped packets, round-trip time, etc.) using a session monitor that collects this information from VoIP endpoints. Ex.1006, [0006], [0021], [0053]. As Dr. Houh testifies:

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.

Ex.1003, ¶84.

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.1003, ¶85 (citing Ex.1005, 5:40-64). Still other obvious benefits of distributing QoS measurement and collection activities would have included that:

- a communication device failure would not inhibit network QoS monitoring; and
- network QoS information, after cross-correlation and cross-referencing, would be useful to identify problem areas on the network.

Ex. 1003, ¶85.

"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).” Ex.1003, ¶86.

“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).” Ex.1003, ¶87.

c. Reasonable expectation of success

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. This conclusion is further bolstered by the fact that “interfaces for LAN were standardized components well-known in the art.” Ex.1003, ¶88; Material Facts #3-4, 8.

“The reasonable expectation of success is also supported by the fact that Minhazuddin already 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, as analyzed above (Ex.1006, [0044]).”

Ex.1003, ¶89. Making the combination would have been within the skillset of a POSITA. Ex.1003, ¶89.

5. Reasons to Combine Shaffer with Minhazuddin.

a. Shaffer is Analogous Art

“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.” Ex.1003, ¶90.

b. Motivation to combine with Shaffer

“A POSITA would have been motivated to combine the teachings of Shaffer and Minhazuddin to produce numerous predictable and beneficial results.” Ex.1003, ¶91.

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

Minhazuddin describes providing live voice communications between packet-switched communication devices (e.g., 116, 123a-n), such as Avaya IP telephones. Ex.1005, 3:62-65, 4:36-5:3. It would have been obvious to a POSITA

for each IP telephone or other packet-switched communication device to have a network interface suitable for LAN connections. Ex.1003, ¶92; *see* Material Facts #3-4, 8.

Shaffer similarly describes packet-switched “real-time voice” communications involving a network terminal/device 10, such as a telephony device, over a network. Ex.1007, 3:14-27. Shaffer provides additional implementation details of how the telephony device connects to a 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. “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.1003, ¶94; 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.”).

Thus, the Minhazuddin-Shaffer combination “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).” Ex.1003, ¶95.

ii. Sending a signal to change codecs

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; Ex.1003, ¶96.

As Dr. Houh testifies, sending a codec-change message in response to QoS measurements would have been obvious:

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.

Ex.1003, ¶97.

Thus, the Minhazuddin-Shaffer combination "merely represents the application of a known technique (e.g., sending a signal per Shaffer to 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)." Ex.1003, ¶98.

The Minhazuddin-Shaffer combination also “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).” Ex.1003, ¶99.

c. Reasonable expectation of success

The results of the Minhazuddin-Shaffer combination “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.1003, ¶100; Material Facts #3-4, 8.

A POSITA also would have reasonably expected success in sending a signal to change codecs. Ex.1003, ¶101. 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 about codec renegotiation, 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. Ex.1003, ¶101.

6. Claim 1

[1.0] *A system comprising:*

Minhazuddin, in the context of Fig. 1, discloses “methods, processes, **systems** and/or apparatus.” Ex.1005, 10:9-12; *see also* Ex.1005, 4:36-46, FIG. 1. Minhazuddin’s system of Fig. 1 renders obvious the recited “*system*.” Ex.1003, ¶¶102-105.

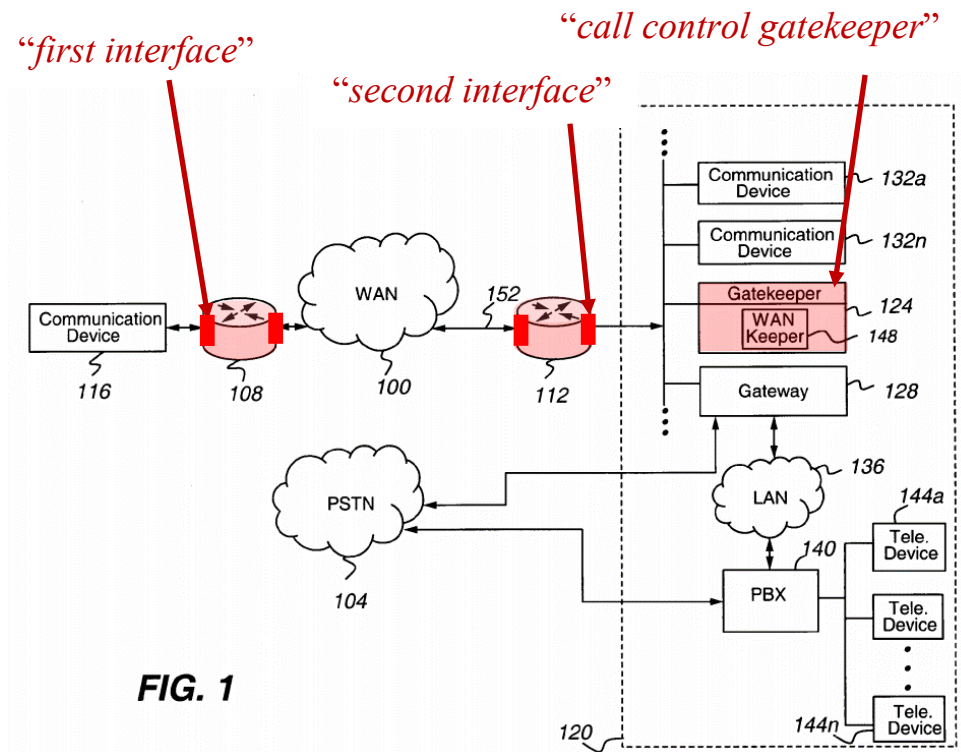
[1.1] *a VoIP service provider system having a call control gatekeeper, a first interface and a second interface;*

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**.” Ex.1005, 4:36-46, FIG. 1. Minhazuddin’s “gatekeeper 124 provides a variety of functions, including call set-up, call control” and also “includes a WAN keeper 148 that performs dynamic call admission control.” Ex.1005, 5:4-11; *see also* 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*.” Ex.1003, ¶¶106-108.

Minhazuddin’s edge router includes an “**interface**.” Ex.1005, 8:6-9. It would have been obvious to a POSITA for local edge routers 108 and 112 to also include an interface to thereby provide a physical connection to different networks,

as it was common and well-known in the art. Ex.1003, ¶109. “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.” Ex.1003, ¶109.

Minhazuddin’s system that provides VoIP services (“*a VoIP service provider system*”), illustrated at Fig. 1 reproduced below, includes 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.1003, ¶110.

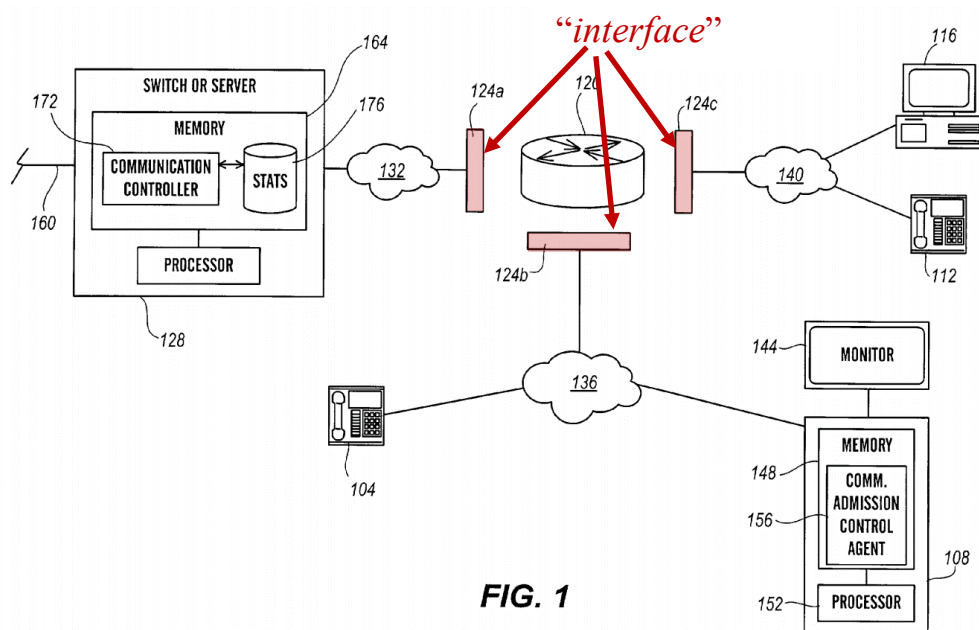


Ex.1003, ¶110; Ex.1005, FIG. 1 (annotated).

Minhazuddin’s disclosed system that corresponds to a collection of different systems or subs-systems is within the scope of the ’368 patent’s “system” which “shall...be taken to include any collection of systems or sub-systems.” Ex.1001, 6:13-17. The location of Minhazuddin’s gatekeeper 124’s within enterprise network 120 comprising a LAN is consistent with the ’368 patent’s embodiment of Fig. 1 (shown on the face of the patent) where the call control gatekeeper 34 is located within 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; Ex.1003, ¶111.

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 this limitation. Ex.1003, ¶112.

Second, to remove any doubt that an “*interface*” is disclosed, Hepworth teaches a VoIP system that includes a router with associated “**interfaces**” for connecting to a LAN or a WAN. Ex.1006, [0036], [0045]. Hepworth's router interfaces are illustrated in Fig. 1:

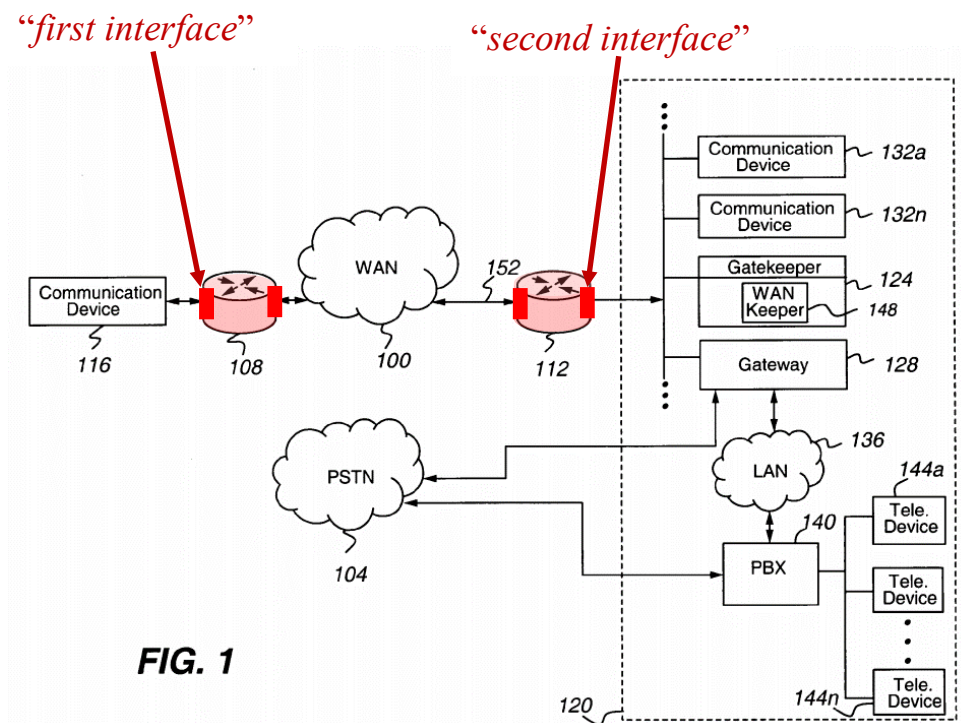


Ex.1003, ¶113; Ex.1006, FIG. 1 (annotated).

It would have been obvious to a POSITA for Minhazuddin's local edge routers 108 and router 112 to utilize interfaces, as Hepworth teaches, to facilitate

physical connections to respective networks. Ex.1003, ¶114; *see also* Reasons to Combine Hepworth and Minhazuddin.

Applying Hepworth's teachings to Minhazuddin would have provided local edge routers 108 and 112 with interfaces to connect to different networks:

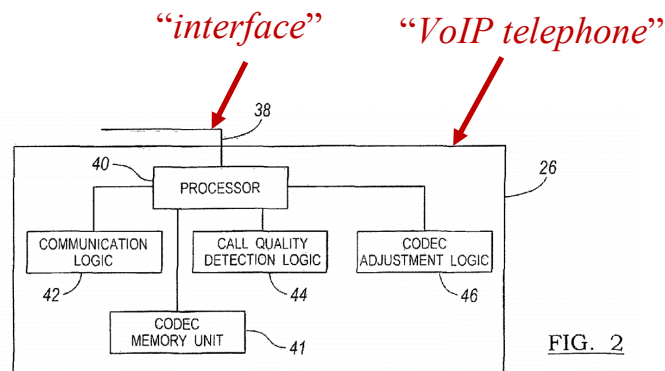


Ex.1003, ¶115; Ex.1005, FIG. 1 (annotated).

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 this limitation. Ex.1003, ¶116.

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

As an initial matter, the '368 patent's exemplary LAN interface corresponds to a network link for connecting the VoIP telephone to a switch. See Ex.1001, 3:66-4:1, FIG. 2 ("interface 38"); Ex.1003, ¶120.



Ex.1003, ¶120; Ex.1001, FIG. 2 (annotated).

First, as discussed at [1.1], the local edge router 108's interface corresponds to the "first interface." Ex.1003, ¶118.

Second, the prior art renders obvious "a first LAN having an interface to a first VoIP telephone" as discussed below. Ex.1003, ¶119.

i. Minhazuddin alone renders obvious this limitation

Minhazuddin states that destination "**communication device 116**" "can be any packet-switched communication device for live voice communications" including an "IP Hardphone" or an "IP Softphone," both of which were commercially available. Ex.1005, 4:64-5:3. Minhazuddin's communication device

116 (e.g., an IP telephone) corresponds to the “*first VoIP telephone.*” Ex.1003, ¶121.

It would have been 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.” Ex.1003, ¶122; *see* Material Facts #3-4, 10.

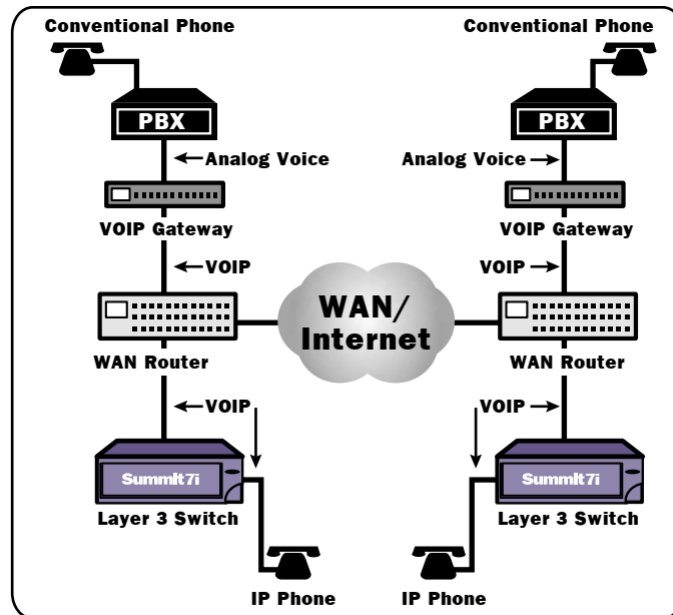
Accordingly, it would have been obvious to a POSITA for Minhazuddin’s communication device 116 (e.g., an Avaya VoIP telephone) (“*first VoIP telephone*”), illustrated at Fig. 1, to be connected to a network link (“*interface*”) of a LAN (“*first LAN*”) that is connected to the local edge router 108 interface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

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.” Ex.1003, ¶¶125-126.

The combination was recognizably 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.1003, ¶127; *see* Material Facts #10-11; 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.”).

The combination is consistent with how companies were implementing VoIP communication systems, including deploying multi-site IP phone systems in the prior art period. Ex.1003, ¶128. For example, Merrill Lynch was reportedly using Avaya IP telephony systems to provide VoIP communications to a first enterprise network in New Jersey and a second enterprise network in Japan. Ex.1015, 2; *see also* Ex.1010, 1, Fig. 1; Ex.1011, 1. Extreme Networks was similarly marketing

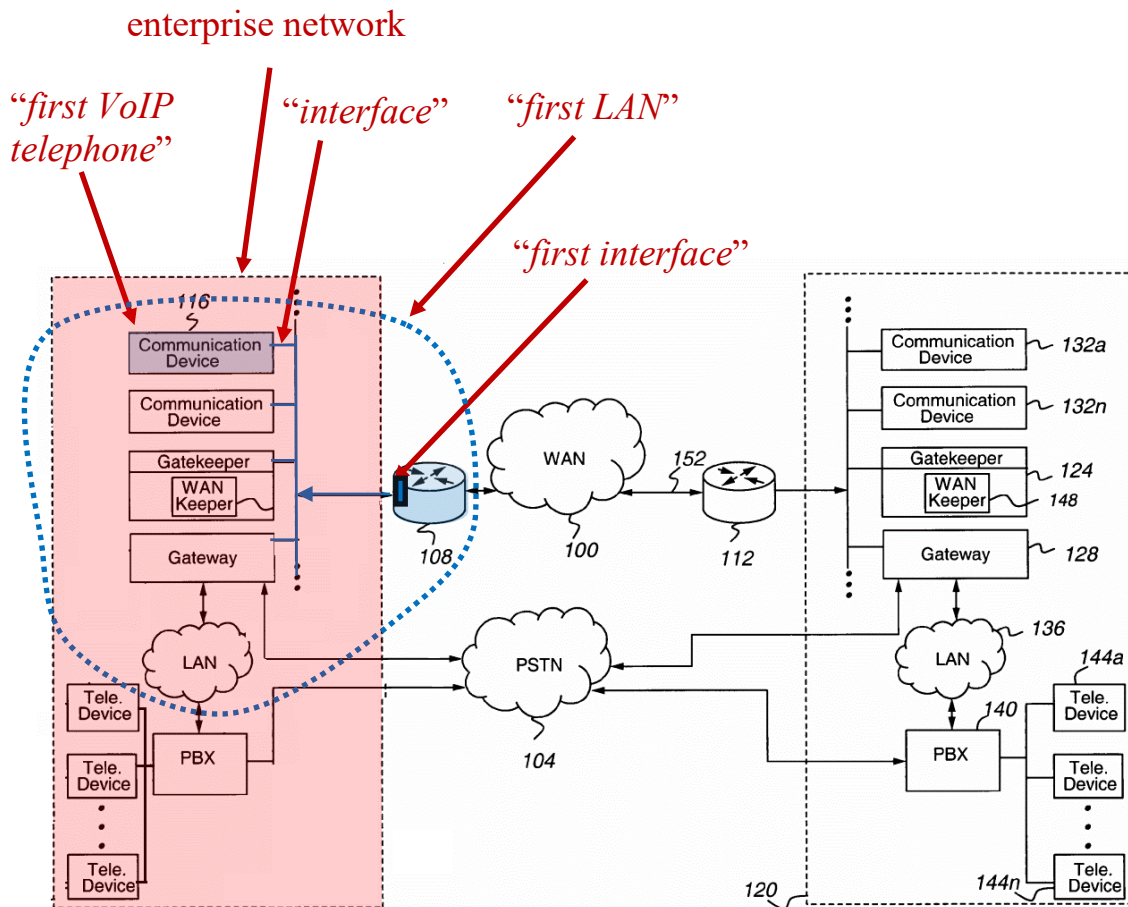
infrastructure solutions for businesses with two enterprise networks, illustrated below:



Voice over IP on the WAN

Ex.1022, p. 5

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 Fig. 1:



Ex.1003, ¶129; Ex.1005, FIG. 1 (modified and annotated to show an additional enterprise network).

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 Avaya’s own 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. Thus, in this second way, Minhazuddin alone renders obvious this limitation. Ex.1003, ¶¶130-131.

ii. Minhazuddin and Hepworth render obvious this limitation

Minhazuddin, in combination with Hepworth and Shaffer, also renders this limitation obvious. Ex.1003, ¶132.

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

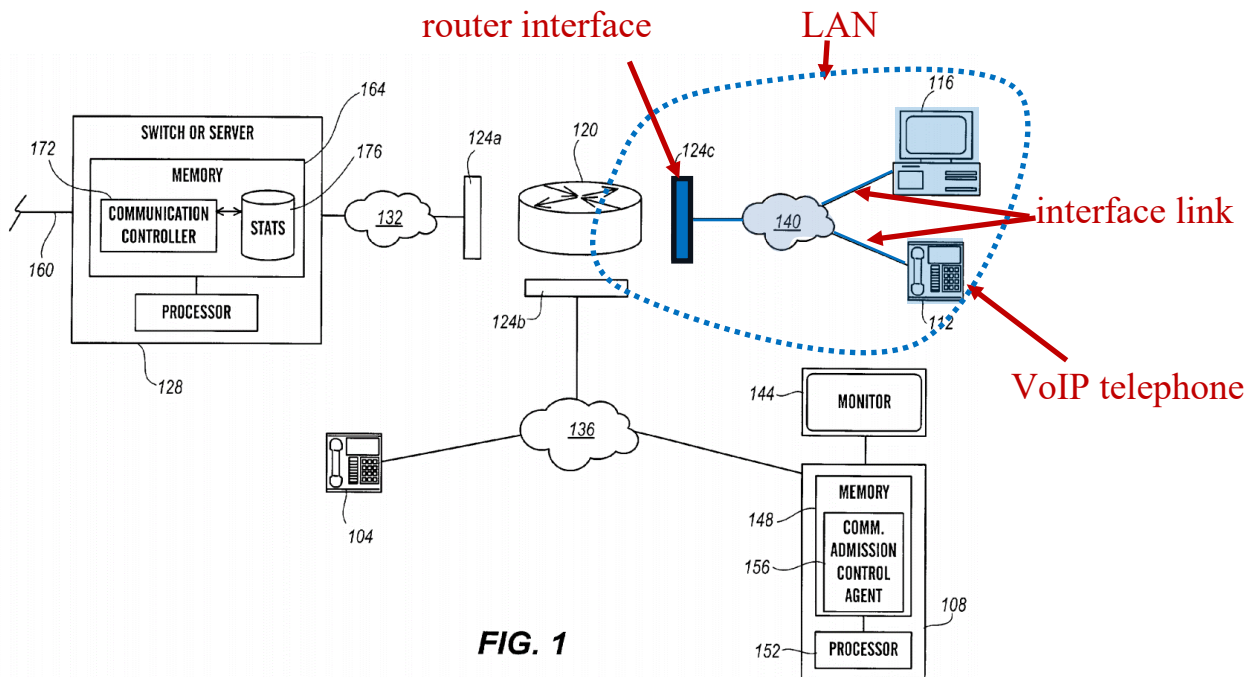


FIG. 1
Ex.1003, ¶133; Ex.1006, FIG. 1 (annotated).

“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 links for connecting a plurality of communication devices to the LAN.”
Ex.1003, ¶134; *see also* Reasons to Combine Hepworth and Minhazuddin.

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 this limitation. Ex.1003, ¶135.

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

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, 3:14-27, 11:21-26, FIGS.1, 2; Ex.1003, ¶136.

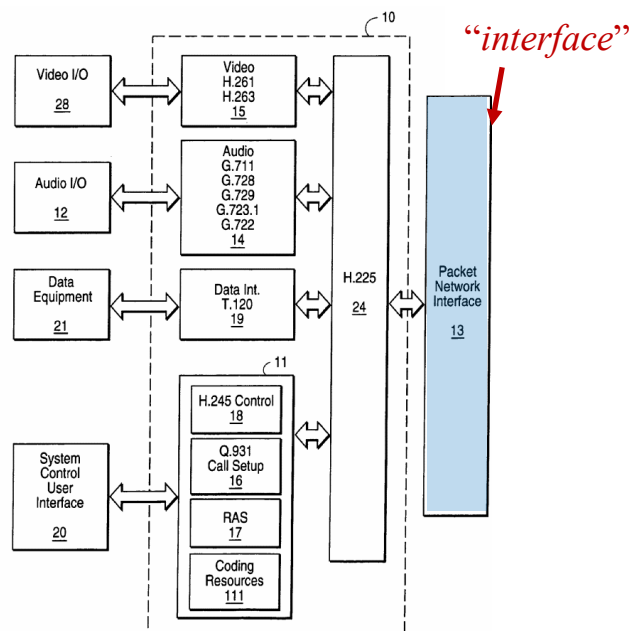


FIG. 2

Ex.1003, ¶136; Ex.1007, FIG. 2 (annotated).

“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.” Ex.1003, ¶137; *see also* Reasons to Combine Shaffer and Minhazuddin.

Thus, 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 this limitation. Ex.1003, ¶138.

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

First, as discussed at [1.1], the local edge router 112’s interface corresponds to the “*second interface*.” Ex.1003, ¶140.

Second, Minhazuddin’s local edge router 112 connects to “**enterprise network 120**” comprising “a **Local Area Network or LAN 136**.” Ex.1005, 4:36-46, FIG. 1. “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.” Ex.1003, ¶¶141-142. Routers were generally understood to represent a LAN/WAN junction point. Ex.1011, 3 (“Routers within enterprise networks represent the LAN/WAN junction points of a network.”);

Ex.1012, 153 (“The enterprise network system is usually built using interconnected LANs.”); *see also* Ex.1010, 3-4; Ex.1011, 1; Ex.1019, 121, 152 (Figure10-5); Ex.1023, 4. Thus, Minhazuddin’s description of local edge router 112 connecting to enterprise network 120 renders obvious that the “*second interface*” (see limitation [1.1]) is “*connected to a second LAN.*” Ex.1003, ¶142-143.

Enterprise network 120 also 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; *see also* Material Facts #1-2. Minhazuddin’s communication device 132a (e.g., an Avaya VoIP telephone) corresponds to a “*second VoIP telephone.*” Ex.1003, ¶144.

A POSITA would have understood that Minhazuddin’s communication device 132a connects to a LAN for communication. Ex.1003, ¶145; *see* Material Facts #3-4; Ex.1013, 2, Fig. 1; Ex.1016, 44; Ex.1018, 65.

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

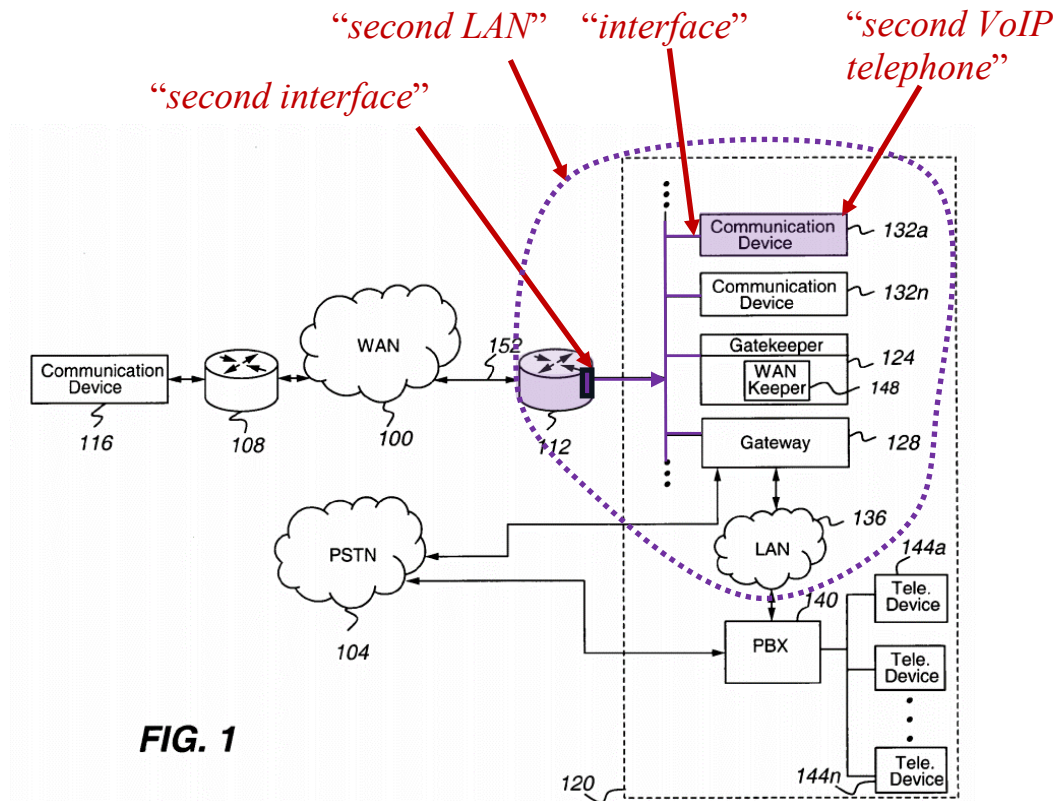


FIG. 1

Ex.1003, ¶146; Ex.1005, FIG. 1 (annotated).

Additionally, to the extent argued that the claimed “interface” is not a network link, it would have been obvious to a POSITA to utilize a network interface, as taught by Shaffer, to facilitate coupling Minhazuddin’s communication devices, including communication device 132a, to a LAN so that packets may be communicated over the LAN. Ex.1003, ¶147; *see also* Reasons to Combine Shaffer and Minhazuddin.

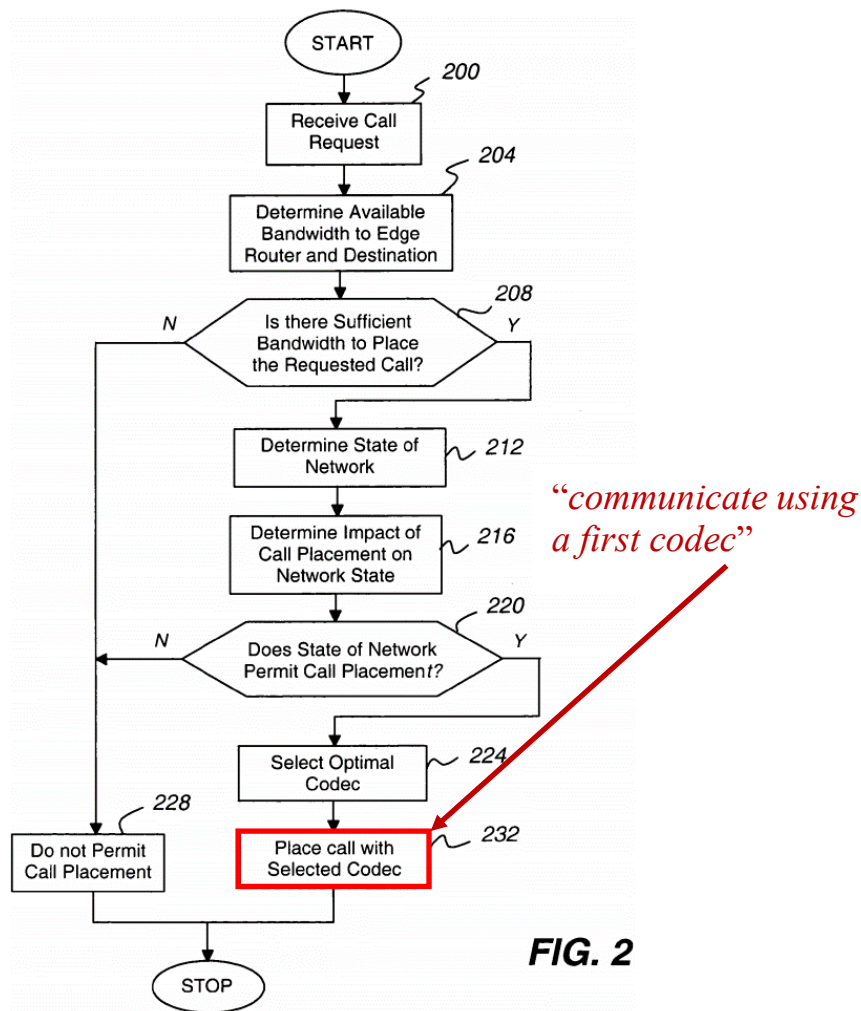
Thus, Minhazuddin in combination with Hepworth and Shaffer discloses that the local router 112 interface is connected to a LAN that has an interface to

communication device 132a (e.g., an Avaya VoIP telephone), which renders obvious this limitation. Ex.1003, ¶148.

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

First, as discussed at [1.2]-[1.3], Minhazuddin’s communication devices 116 and 132a respectively correspond to “*the first and second VoIP telephones.*” Ex.1003, ¶¶149-150.

Second, Minhazuddin teaches selecting a codec to be used for communication between the communication devices and that “**the call is permitted to proceed with the selected codec.**” Ex.1005, 7:52-8:2. If “no such codec exists...the WAN keeper 148...[will] not permit placement of the call.” Ex.1005, 8:1-5; Ex.1003, ¶151. Minhazuddin’s process is shown below at Figure 2.



Ex.1003, ¶152; Ex.1005, FIG. 2 (annotated).

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. Ex.1003, ¶153. The selected codec is a “first codec.”

Thus, Minhazuddin's disclosure that communication devices 116 and 132a are configured to communicate using the selected codec, renders obvious this limitation. Ex.1003, ¶154.

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

First, as discussed at [1.1], Minhazuddin's gatekeeper 124 (including WAN keeper 148) corresponds to "*the call control gatekeeper.*" As discussed at [1.2], Minhazuddin's communication device 116 corresponds to "*the first VoIP telephone.*" Ex.1003, ¶156.

Second, 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**" and that the "measured QoS metrics include packet delay, jitter, packet loss,..." Ex.1005, 5:53-64; Ex.1003, ¶157.

Hepworth teaches a suitable monitoring technique: a monitor (e.g., Avaya VoIP Monitoring ManagerTM) receives a signal with QoS information (e.g., jitter, packet loss, number of dropped packets, round-trip time, etc.) from VoIP endpoints. In one example, "**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.” Ex.1006, [0006]; *see also* Ex.1006, [0021], [0044], [0053]. Transmitting QoS information to a session monitor was broadly known in the art. Ex.1003, ¶¶158-159 (citing Ex.1021, Ex.1024).

Hepworth and Minhazuddin render obvious implementing Minhazuddin’s 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. Ex.1003, ¶160. Such QoS information would have allowed 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.” Ex.1003, ¶160; *see e.g.*, Ex.1005, 8:23-52; *see also* Reasons to Combine Hepworth and Minhazuddin.

A POSITA would have seen the combination as consistent with Minhazuddin’s description of monitoring QoS information “by any suitable technique including receiving feedback signals.” Ex.1005, 5:53-58; Ex.1003, ¶161.

Thus, Minhazuddin in combination with Hepworth discloses configuring WAN keeper 148 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 this limitation. Ex.1003, ¶162.

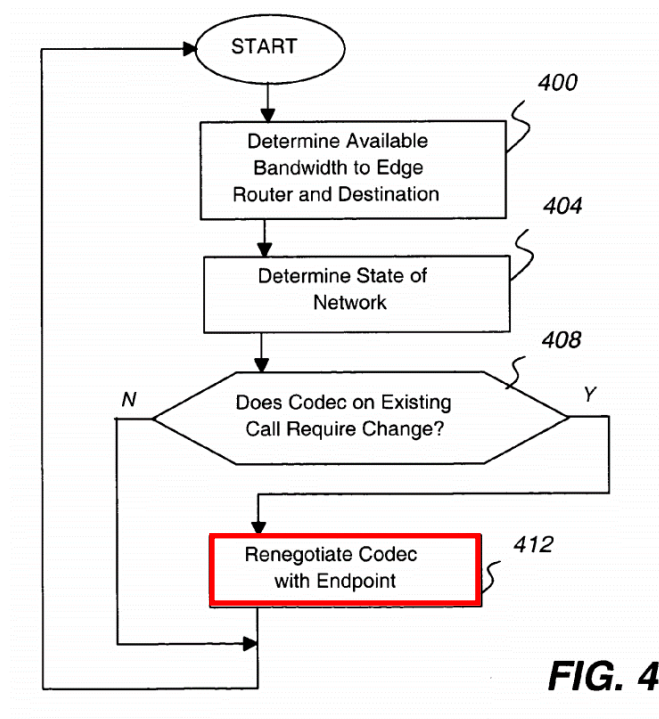
[1.5.2] send an adjustment signal based on the call quality signal to the first and second VoIP telephones instructing the first and second VoIP telephones to communicate using a second codec; and

First, as discussed at [1.2]-[1.3], Minhazuddin's communication devices 116 and 132a respectively correspond to "*the first and second VoIP telephones.*" Further, as discussed at [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.*" Ex.1003, ¶¶163-164.

Second, Minhazuddin discloses that WAN keeper 148 (implemented in gatekeeper 124) utilizes QoS information to enact "bandwidth control measures" when selected thresholds are exceeded. Ex.1005, 8:23-41. "In one measure, **the codecs used on existing VoIP calls are renegotiated** to a lower bit rate codec (that will use less bandwidth)...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.*" Ex.1003, ¶¶165-166.

Minhazuddin's Fig. 2 shows an exemplary "operation of the WAN keeper."

Ex.1005, 4:33-34. Step 232 has Minhazuddin's endpoints (e.g., communication device 116 and communication device 132a) renegotiate to a lower bit rate codec:



Ex.1003, ¶167; Ex.1005, FIG. 4 (annotated).

Since the codec is implemented in Minhazuddin's endpoints, e.g., communication device 116 and communication device 132a, it would have been obvious for WAN keeper 148 send a signal to both communication endpoints directing them to change codecs. Ex.1003, ¶¶168; *see* Material Facts #6-7.

Shaffer discloses a similar "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") incorporated in a gatekeeper that monitors

QoS information and, if a threshold is met, directs the endpoints to renegotiate to a codec that uses less bandwidth by sending a message. Ex.1007, 1:51-61, 2:9-14, 3:4-10, 10:17-38, 11:20-42, 12:21-47, FIGS. 4-9; Ex.1003, ¶169.

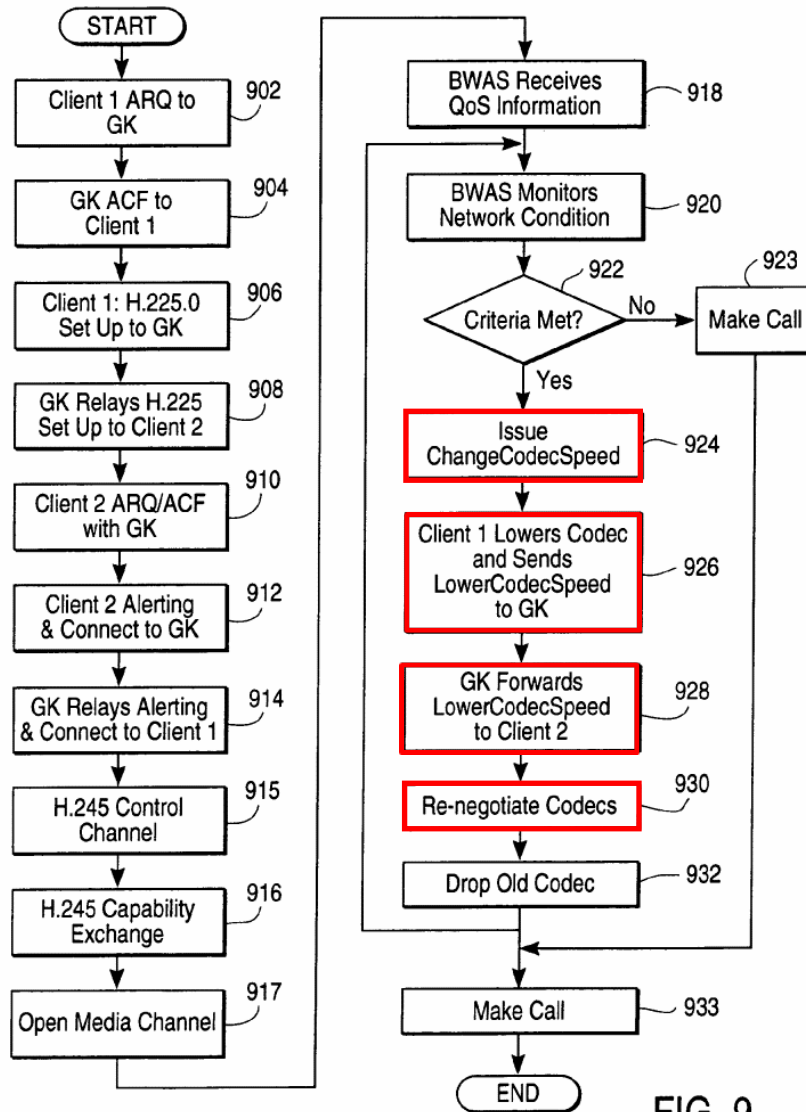


FIG. 9

Ex.1003, ¶169; Ex.1007, FIG. 9.

“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.” Ex.1003, ¶170. By making this combination, Shaffer’s signaling techniques “would inform the endpoints (including communication devices 116 and 132a) to renegotiate to a lower bit rate codec.” Ex.1003, ¶170; *see also* Reasons to Combine Shaffer and Minhazuddin.

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*.” Ex.1003, ¶171.

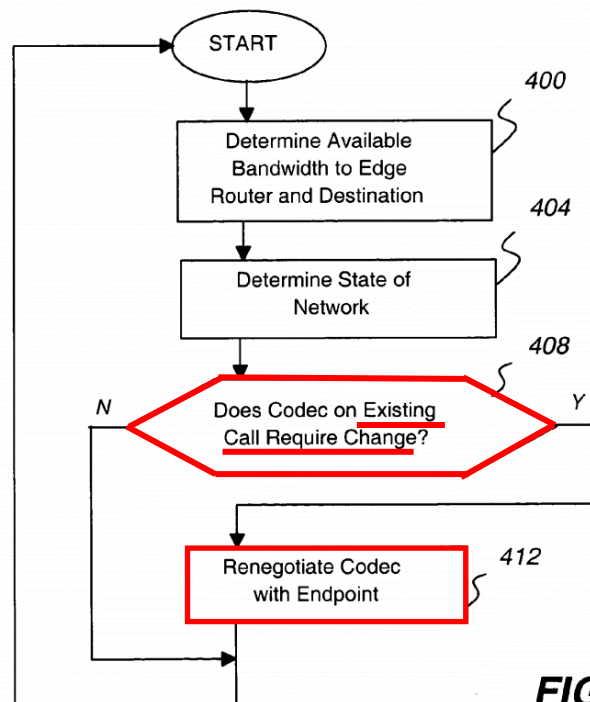
This mapping of the prior art is consistent with the ’368 patent specification. Ex.1001, 1:67-2:3 (“When the gatekeeper receives this call quality signal, the gatekeeper will instruct the VoIP telephones to renegotiate the telephone call and utilized a lower bandwidth codec.”); Ex.1003, ¶172.

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 directing them to renegotiate to a lower bit rate codec, which renders obvious this limitation. Ex.1003, ¶173.

[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.

First, as discussed at [1.5.2], the prior art discloses sending a signal to renegotiate to a lower bit codec based on QoS information, which corresponds to “the adjustment signal based on the call quality signal is sent to the first and second VoIP telephones.” Ex.1003, ¶¶174-175.

Second, Minhazuddin in the context of Fig. 4 discloses that “codecs used on existing VoIP calls are renegotiated to a lower bit rate codec (that will use less bandwidth).” Ex.1005, 8:42-45.



Ex.1003, ¶176; Ex.1005, FIG. 4 (annotated).

When communication devices 116, 132a are participating in an existing call and the QoS metrics exceed selected thresholds (Ex.1005, 8:37-41), it would have been obvious “for WAN keeper 148 to send the adjustment signal to each endpoint to renegotiate the codec,” as discussed at [1.5.2]. Ex.1003, ¶177.

Shaffer further discloses signaling endpoint telephones to change codecs during a call. Shaffer explains that “**existing call(s) will have its (or their) codecs renegotiated to a lower level,**” for example, by having the “BWA send[] a Lower Codec Speed message to all active H.323 entities...to renegotiate their codecs.” Ex.1007, 2:9-14; *see also* Ex.1007, 8:51-67 (“...**while calls are ongoing...the existing call(s) will have its (or their) codecs renegotiated** to a lower level.”); *see also* Ex.1007, 9:43-46, 10:17-32, 11:27-31, 12:7-11, FIG. 9; Ex.1003, ¶178.

“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... 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.1003, ¶179 (quoting Ex.1005, 8:42-45). In this way, the existing call would continue despite a reduction in available

bandwidth. Ex.1003, ¶179; *see also* Reasons to Combine Shaffer and Minhazuddin.

Thus, Minhazuddin in combination with Hepworth and Shaffer discloses that WAN keeper 148 sends the adjustment signal to the communication devices 116, 132a during an existing call between the communication devices 116, 132a, which renders obvious this limitation. Ex.1003, ¶180.

7. Claim 2

Claim 2 is identical to claim 1—and thus obvious as discussed above—except for its final limitation, analyzed below. Ex.1003, ¶¶181-187.

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

Minhazuddin’s QoS information includes “packet loss...[corresponding to] packets that are not delivered to a destination.” Ex.1005, 5:53-64. Additionally, Hepworth discloses that 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...packet loss” and “**a number of dropped packets.**” Ex.1006, [0006]; *see also* Ex.1006, [0003]-[0005], [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.” Ex.1003, ¶189.

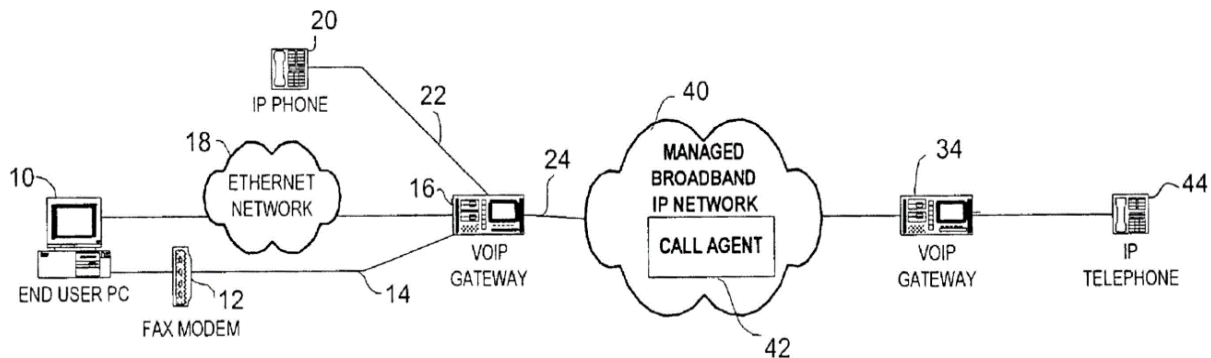
Consistent with the analysis at [1.5.1], it would have been obvious to a POSITA for the QoS information (“*the call quality signal*”) transmitted to WAN keeper 148 to include packet loss statistics, such as a number of dropped voice packets during the VoIP call between the endpoints, because this information 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.1003, ¶¶190 (citing Ex.1074, Ex.1087, Ex.1096 regarding VoIP sensitivity to dropped packets), 285 (citing Exs.1091-1094 regarding synonymity of “dropped” and “lost” packets); Ex.1011, 10 (“...some protocols are sensitive to dropped packets...”); Ex.1025, 7:2-5 (“These statistics may include... number of dropped packets.”); Material Fact #12; *see also* Reasons to Combine Hepworth and Minhazuddin.

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

1. Mundra

Mundra describes a VoIP system. Ex.1008, [0002], Abstract. Mundra illustrates a “broadband IP network connecting two VoIP telephony devices” 20 and 44 in Figure 2, below. Ex.1008, [0018]. The network also includes a Call

Agent 42.



Ex.1008, Fig. 2.

Mundra acknowledges that existing protocols, such as “media gateway control protocol (MGCP),” had some limited support for changing codecs. Ex.1008, [0010]. Mundra describes extending that capability so that Call Agent 42 would “receive commands from IP phone 20” and “change the codecs in realtime during a call transmission.” Ex.1008, [0020].

Mundra is analogous art to the ’368 patent because it pertains to the same field of VoIP communications. Ex.1003, ¶¶192-195, 200; Ex.1008, [0002]. Mundra is also analogous art because it is reasonably pertinent to the problem of changing a voice codec during an ongoing VoIP telephone call. *See* Ex.1008, Abstract; Ex.1001, Abstract.

2. RFC3435

RFC3435 describes the Media Gateway Control Protocol (MGCP) that facilitates communication between a Call Agent and media gateways, for example, in the context of “Voice over IP.” Ex.1027, 1.

RFC3435 is analogous art to the '368 patent because it pertains to the same field of VoIP communications. Ex.1003, ¶¶196-198, 200; Ex.1027, 1; Ex.1001, Abstract. RFC3435 is also analogous art because it is reasonably pertinent to the problem of changing a voice codec during an ongoing VoIP telephone call. Ex.1027, 52 (“...switching to a different coding scheme...”); Ex.1001, Abstract.

3. RFC3660

RFC3660 describes a “basic set of Media Gateway Control Protocol (MGCP) packages.” Ex.1028, 1. The packages are “for use with MGCP 1.0 as defined in [RFC3435].” Ex.1028, 3, 60.

RFC3660 is analogous art because it is intended for use with RFC3435, and thus, RFC3660 pertains to the same field of “Voice over IP” communications. Ex.1003, ¶¶199-200; Ex.1028, 3; Ex.1027, 1; Ex.1001, Abstract. RFC3660 is also analogous art because it is reasonably pertinent to the problem of changing a voice codec during an ongoing VoIP telephone call. Ex.1028, 17 (“The Call Agent may for example instruct the gateway to change the codec used on a connection...”); Ex.1001, Abstract.

4. Reasons to Combine Mundra with RFC3435 and RFC3660

A POSITA would have been motivated to consider Mundra's disclosures in combination with those of RFC3435 and RFC3660. Ex.1003, ¶201.

Mundra refers to Media Gateway Control Protocol (MGCP) as an example signaling protocol used with Voice of Packet communications. Ex.1008, [0010]. Mundra further notes that MGCP provides some support for changing the codec of an active communication session. Ex.1008, [0010]; *see* Ex.1003, ¶¶202-203.

At the time of Mundra's filing, MGCP was defined by Request for Comments 2705. *See* Ex.1026. When Mundra published in February 2004, RFC2705 had been obsoleted by RFC3435 and RFC3660. *See* Ex.1039, 4:49-53. More specifically, RFC3435 updated the main parts of the protocol specification, but "Removed packages from [the] protocol specification," indicating that the packages would "be provided in separate documents instead." Ex.1027, 163. RFC3660 is one of those "separate documents," as RFC3660 "provides a basic set of Media Gateway Control Protocol (MGCP) packages." Ex.1028, 1; Ex.1003, ¶204.

Thus, at the time of Mundra's publication, it would have been obvious for a POSITA seeking more information about MGCP to consult RFC3435 and RFC3660, which defined the MGCP protocol. Ex.1003, ¶¶204-205. Thus, the

combination would have been obvious because of Mundra's express reference to the MGCP protocol. Ex.1003, ¶205.

A POSITA would have expected success because others in the art were already successfully using MGCP for monitoring call quality and causing codec changes. Ex.1003, ¶¶206-207 (citing Ex.1075, [0013]; Ex.1076, 7:54-62).

Additional analysis supporting the obviousness of the combination is provided in the detailed claim analysis below.

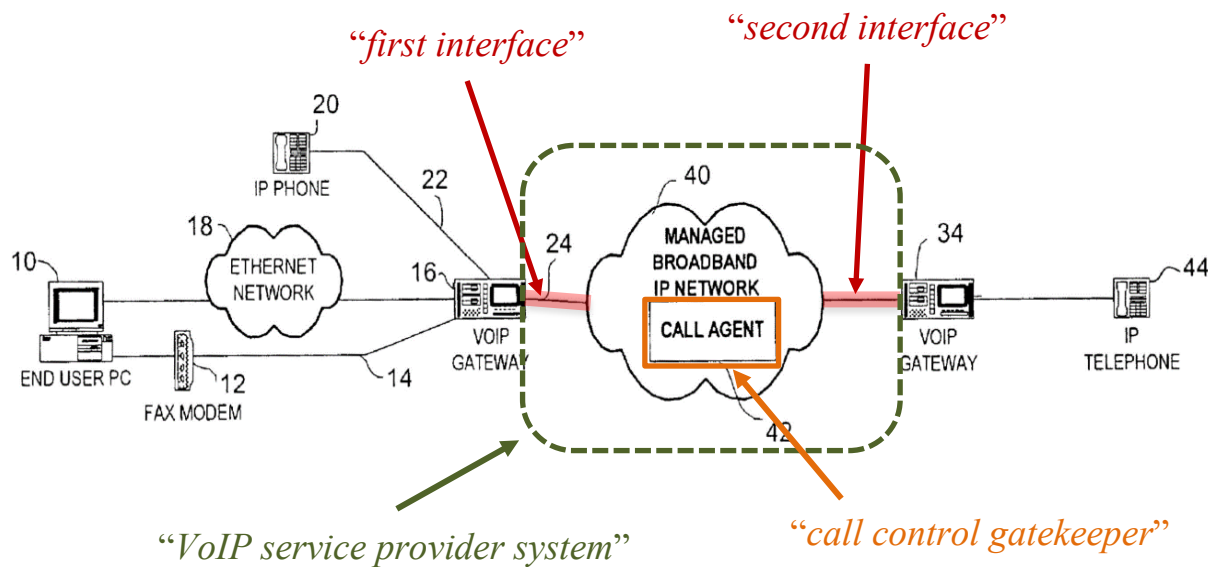
5. Claim 1

[1.0] *A system comprising:*

Mundra illustrates in Fig. 2 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.” Ex.1008, [0019]. Mundra's system of Fig. 2 (modified as discussed below) renders obvious the claimed “*system*.” Ex.1003, ¶¶208-210.

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

Mundra illustrates the features recited in [1.1] in Fig. 2, annotated below. Ex.1003, ¶¶211-223.



Ex.1003, ¶212; Ex.1008, Fig. 2 (annotated).

“VoIP service provider system”: Mundra’s Fig. 2 system includes a “managed network 40” providing connectivity between two IP telephones. Ex.1008, [0019]. The network 40 is a “managed broadband [IP] network” under the control of a “network provider.” Ex.1008, [0018]. It would have been obvious for Mundra’s “network provider” to be a *service provider* because Mundra’s system provides “Internet telephony service” to a user. Ex.1008, [0020]; Ex.1003, ¶213. Mundra also refers to “ISP-managed networks 42 [*sic*, 40], 46.” Ex.1008, [0018]. “ISP” was a commonly understood acronym for “internet service provider.” Ex.1003, ¶213 (citing Ex.1009, 13:12-13; Ex.1020, 7:24). Connected to network 40 are “Gateways 16, 34” (each labeled “VOIP GATEWAY”) that “support voice over packet calls.” Ex.1008, [0019], Fig. 2. Finally, Mundra’s

whole disclosure is in the context of “a VoIP managed network.” Ex.1008, [0008]. Managed network 40, together with the connections to VoIP Gateways 16 and 34, corresponds to the claimed “*VoIP service provider system*.” Ex.1003, ¶214-215; *see also* Ex.1001, 6:13-17.

“call control gatekeeper”: Within network 40 is a “call agent module 42 that is managed by the network managing entity.” Ex.1008, [0019]. Mundra also refers to call agent module 42 as the “call agent,” “signaling agent 42” and “call controller agent 42.” Ex.1008, [0020], Fig. 2. Mundra’s disclosure is made in the context of media gateway control protocol (MGCP). Ex.1008, [0010]. Thus, it would have been obvious for Mundra’s “call agent” to operate in general accordance with MGCP, which is explained in RFC3435. Ex.1003, ¶216-217. RFC3435 describes “a Call Agent, which contains the call control ‘intelligence.’” Ex.1027, 1. In other words, “the Call Agent handles the call signaling and call processing functions.” Ex.1027, 7. For example, the Call Agent will control the operation of the gateways, such as to “direct the handling of [a new] connection by the gateway.” Ex.1027, 47, 20, 5. The Call Agent also “presents itself as an ‘H.323 **Gatekeeper**.’” Ex.1027, 7. Since an MGCP call agent handles call signaling, includes call control intelligence, directs incoming calls, controls the operation of the gateways, and presents itself as a gatekeeper, it would have been obvious to a

POSITA for Call Agent 42 to act as a “*call control gatekeeper*” as claimed.

Ex.1003, ¶217.

“*first interface*”: Mundra illustrates in Fig.2 that VoIP gateway 16 connects to managed network 40 via “a high speed Internet connection 24 such as a digital subscriber line (DSL), cable modem 24, or T1/T5 line.” Ex.1008, [0017]; Fig. 2. The high-speed internet connection 24 is a “*first interface*.” Ex.1003, ¶218.

Alternatively, to the extent that Patent Owner argues that the claimed “*first interface*” is a network interface (e.g., a port), it would have been obvious for managed network 40 to have a network interface for connecting to connection 24. Ex.1003, ¶219; Material Fact #8. For example, for a “digital subscriber line (DSL)” link, the managed network 40 would have a digital subscriber line access multiplexer (DSLAM) with a port (“*first interface*”) connected to connection 24. Ex.1003, ¶219; 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.”). For a “cable modem” line, the managed network 40 would have a cable modem termination system (CMTS) with a port (“*first interface*”) connected to connection 24. Ex.1003, ¶220; 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”).

“second interface”: Mundra illustrates in Fig.2 that VoIP Gateway 34 connects to managed network 40 through an unlabeled connection, which is a “second interface.” Ex.1003, ¶221.

Alternatively, to the extent that Patent Owner argues that the claimed “second interface” is a network interface (e.g., a port), it would have been obvious for managed network 40 to have a network interface for connecting to the unlabeled connection. Ex.1003, ¶222; Material Fact #8. For example, Mundra describes VoIP Gateway 34 as similar to VoIP Gateway 16, so it would have been obvious for VoIP Gateway 34 to connect to managed network 40 via a “high speed Internet connection.” Ex.1008, [0017]; Ex.1003, ¶222. Thus, it would have been obvious for managed network 40 to have a network interface—such as a DSLAM port or a CMTS port—for the same reasons discussed above for the “first interface.” Ex.1003, ¶222.

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

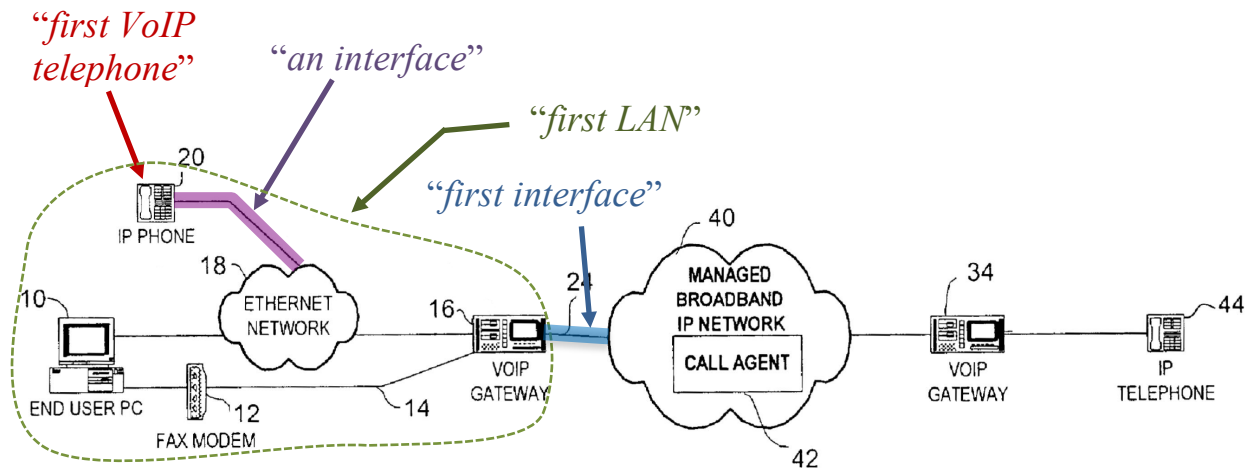
As discussed above in [1.1], Mundra renders obvious a “first interface” connecting managed network 40 to VoIP Gateway 16. As explained below, it

would have been obvious for VoIP Gateway 16 to be in a “*first LAN having an interface to a first VoIP telephone.*” Ex.1003, ¶¶224-230.

Mundra illustrates VoIP Gateway 16 as connecting to IP Phone 20 (“*a first VoIP telephone,*” illustrated as a hard phone) via “telephony port 22.” Ex.1008, [0017], Material Fact #2; Ex.1003, ¶226. A POSITA would have known that many IP phones would not require a specific “telephony port,” but instead they would have an Ethernet port for connecting to Mundra’s Ethernet network 18. Ex.1003, ¶227; *see* Material Facts #3-4. A POSITA would have been motivated to modify Mundra’s network arrangement—so that IP Phone 20 connects to Ethernet network 18—because it was common to connect IP phones to an Ethernet network, especially in corporate networks. Ex.1003, ¶227; *see* Material Fact #10. Connecting IP phones to an Ethernet network also allowed for the use of Power-over-Ethernet to power the IP phone from the switch, which simplified wiring, provided service during a power outage, and lowered costs. Ex.1003, ¶227; *see* Material Fact #11.

Thus, it would have been obvious to modify Mundra’s Fig. 2 to connect IP Phone 20 to Ethernet network 18, is illustrated below. Ex.1003, ¶228; *see* Ex.1079, Fig. 1, [0027] (connecting “IP phone sets” to a LAN “is a typical arrangement...well identified in the art”). Ethernet network 18 and the devices

attached to it form a “*first LAN*.” The connection from IP phone 20 to Ethernet network 18 is “*an interface*.”



Ex.1003, ¶228; Ex.1008, Fig. 2 (modified to connect IP phone 20 to Ethernet network 18 instead of VoIP Gateway 16, annotated).

To the extent that Patent Owner argues that the claimed “*an interface*” is a network interface (e.g., a port), it would have been obvious to a POSITA that the connection from IP phone 20 to Ethernet network 18 would include an interface, such as an RJ-45 port. Ex.1003, ¶229; see Material Facts #2-4, 8.

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

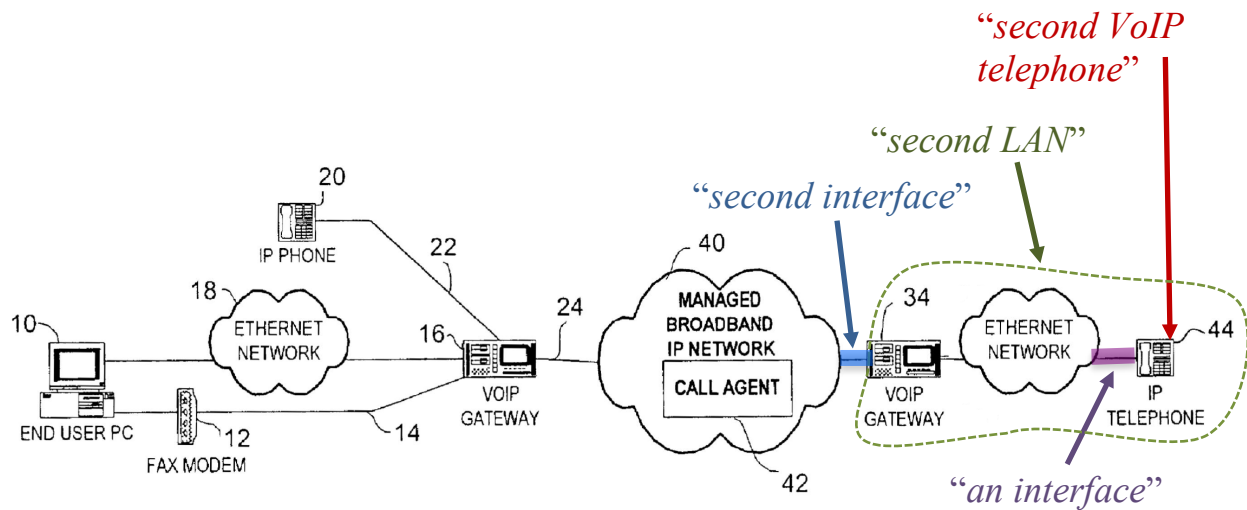
As discussed at [1.1], Mundra renders obvious a “*second interface*” connecting managed network 40 to VoIP Gateway 34. As explained below, it would have been obvious for VoIP Gateway 34 to be in a “*second LAN having an interface to a second VoIP telephone*.” Ex.1003, ¶¶231-237.

VoIP Gateway 34 connects to IP Phone 44 (“*a second VoIP telephone*”).

Ex.1008, Fig.2. For the reasons discussed at [1.2], it would have been obvious for IP phone 44 to connect to VoIP Gateway 34 through an Ethernet network.

Ex.1003, ¶234. Adding an Ethernet network to the right side of Mundra’s Fig. 2 would have beneficially allowed additional communication devices (for example, computers like personal computer 10 or additional IP phones) to access managed network 40, and for the use of power-over-Ethernet. Ex.1003, ¶234; *see* Material Facts #10-11.

Mundra’s Fig. 2, modified to show IP Phone 44 connecting through an Ethernet network to VoIP Gateway 34, is illustrated below. The added Ethernet network, together with the devices attached to it, is a “*second LAN*.” The connection from IP phone 44 to the added Ethernet network is “*an interface*.”



Ex.1003, ¶235; Ex.1008, Fig. 2 (modified to add an Ethernet network between IP phone 44 and VoIP Gateway 34, annotated).

To the extent that Patent Owner argues that the claimed “*an interface*” is a network interface (e.g., a port), it would have been obvious to a POSITA that the connection from IP phone 44 to the added Ethernet network would include an interface such as an RJ-45 port. Ex.1003, ¶236; *see also supra*, Material Facts #2-4, 8.

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

It would have been obvious to a POSITA for there to be a VoIP telephone call between IP telephones 20 and 44 (“*the first and second VoIP telephones*”). Ex.1003, ¶240. Mundra is directed to “improv[ing] the quality of voice calls over IP networks,” and Fig. 2 illustrates “a broadband IP network connecting two VoIP telephony devices” (i.e., IP telephones 20 and 44). Ex.1008, [0016], [0017], Fig.2.

Notably, IP telephones 20 and 44 are the only telephones in Fig. 2. It would have been obvious for IP telephones 20 and 44 to be “*configured to communicate*” with one another, for example, by having “a VoIP call” between them. Ex.1008, [0021]; Ex.1003, ¶240.

Mundra explains that “When placing a VoIP call, there is typically an original VOP [voice over packet] codec limitation that is negotiated at the beginning of the call.” Ex.1008, [0008], [0003]. Mundra provides a list of exemplary codecs. Ex.1008, [0004]-[0005].

Mundra further explains that for “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.” Ex.1008, [0009]. Mundra further describes “**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. Since the IP telephones 20 and 44 “support voice over packet calls” and are the “sending and receiving devices” for users’ speech, it would have been obvious to a POSITA for IP telephones 20 and 44 to be “*configured to communicate using a first codec.*” Ex.1008, [0019], [0009]; Ex.1003, ¶¶240-241; Material Fact #7.

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

Mundra describes Call Agent 42 (“*call control gatekeeper*”) receiving “**commands**” from IP phone 20 (“*the first VoIP telephone*”) that then cause Call Agent 42 to “**change the codecs in realtime during a call transmission.**” Ex.1008, [0020]; Ex.1003, ¶242-244.

The user’s commands are sent using “keys on a dialpad” to “enable the best, lowest, or midrange codec” according to user preferences. Ex.1008, [0025], [0008] (“Changing the codec based upon the user’s intentions...”), claims 5, 7. It would have been obvious for the dialpad keys to be on IP phone 20. Ex.1003, ¶246. Thus, Mundra’s “command” sent from IP phone 20 to Call Agent 42 to control which codecs are enabled is a “*call quality signal.*” Ex.1003, ¶¶247-248.

Alternatively, if Patent Owner argues that the “*call quality signal*” provides information about the currently detected call quality, Mundra describes monitoring call quality. For example, “Monitored events... are received by gateway 16.” Ex.1008, [0020]. These monitored events include “events affecting a VoIP call, including processing power, delay, and jitter.” Ex.1008, [0021]. A POSITA would have recognized that measurements of delay and jitter are relevant to VoIP call quality. Ex.1003, ¶249-251; *see* Material Fact #12. Thus, Mundra’s monitored

event, which provides information about network conditions such as delay or jitter, is a “*call quality signal*.” Ex.1003, ¶251.

Consistent with Mundra’s discussion of monitored events, RFC3660 describes a “Quality alert” event that occurs when “packet loss rate or the combination of delay and jitter exceeding a quality threshold.” Ex.1028, 46. Thus, an MGCP quality alert is also a “*call quality signal*.” Ex.1003, ¶¶252-254. Since Mundra describes its technology in the context of MGCP, it would have been obvious to a POSITA for Mundra’s monitored event to be implemented as an MGCP quality alert. Ex.1003, ¶¶255-257.

As discussed above, Mundra describes a monitored event being received by VoIP Gateway 16. Ex.1008, [0020]-[0021]. VoIP Gateway 16 then sends “call control protocols to call controller agent 42 to cause a codec change.” Ex.1008, [0021]. Mundra is unclear whether these “call control protocols” include the monitored event.

Considering Mundra in combination with RFC3435, however, it would have been obvious for Call Agent 42 to receive a “monitored event” (such as a quality alert) from IP Phone 20 (either directly or via VoIP Gateway 16). Ex.1003, ¶257. RFC3435 explains that a call agent (e.g., Call Agent 42) can request to receive such event messages: “A Call Agent may ask to be notified about certain events occurring in an endpoint.” Ex.1027, 28. A POSITA would have expected success

because MGCP (defined by RFC3435 and RFC3660) was a well-known signaling protocol, and the combination merely reflects how RFC3435 and RFC3660 were intended to be used. Ex.1003, ¶¶204, 207; *see* Ex.1080, [0026] (“Signaling protocols include, for example, SIP, MGCP,.... The use of signaling packets and signaling protocols is well-known.”); Ex.1081, [0019]; Ex.1082, [0063]-[0064] (“A quality alert may be used to designate when the packet loss rate or the combination of delay and jitter exceed a quality threshold in accordance with an RTP package in RFC 3660.”). Thus, it would have been 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*”). Ex.1003, ¶257.

RFC3435 also explains that a quality alert message originates on a VoIP endpoint: “Some events..., such as... the quality alert, are performed or detected on connections terminating in the endpoint.” Ex.1027, 44. A POSITA would have recognized that IP phone 20 is an endpoint, for example, for a VoIP telephone call between IP Phone 20 and IP phone 44. Ex.1003, ¶257; *see* Material Fact #6. Thus, it would have been obvious for IP Phone 20 to send a “monitored event” such as a quality alert message (“*a call quality signal*”). Ex.1003, ¶¶253-254.

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*”). Ex.1003, ¶¶257-258.

[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

As discussed at [1.1], Call Agent 42 corresponds to the “*call control gatekeeper,*” and as at [1.2]-[1.3], IP phones 20 and 44 correspond to “*the first and second VoIP telephones.*” Ex.1003, ¶¶259-260.

Mundra generally describes “changing the first codec to the second codec in realtime during transmission of the voice call.” Ex.1008, claim 1. For example, “network congestion... may result in selection of a low bit rate codec,” whereas the “availability of greater bandwidth could also result in changing to a high bit rate codec.” Ex.1008, [0022].

As discussed at [1.5.1], it would have been obvious for Call Agent 42 to receive a “*call quality signal*” providing, for example, information about the user’s selection of enabled codecs, or about network conditions like delay, jitter, or packet loss. Ex.1003, ¶260. Mundra explains that receiving a notification about changing network conditions causes “the agent module [42] to prepare both gateways for a potential change in codec.” Ex.1008, [0023]. Since Mundra contemplates not merely “to prepare” for a codec change, but indeed to “the[n] change the codecs in realtime during a call transmission” (Ex.1008, [0020]), it

would have been obvious to a POSITA for Call Agent 42 to send an “*adjustment signal*” message to change codecs (“*instructing... to communicate using a second codec*”). Ex.1003, ¶¶261-262. Since the codec is implemented on IP phones 20 and 44, it would have been obvious to a POSITA for the codec-change message from Call Agent 42 to be sent to IP phones 20 and 44 (“*to the first and second VoIP telephones instructing the first and second VoIP telephones*”). Ex.1003, ¶263; Material Fact #7; *see* Ex.1008, [0004], [0009] (discussing “codecs that are supported at sending and the receiving telephony devices” and that “the sending and receiving devices must use the same codecs during speech transmissions.”).

Alternatively, as discussed at [1.5.1], it would have been obvious to modify Mundra, based on RFC3435 and RFC3600, so that Call Agent 42 would receive information about network conditions and make a codec-change decision. Ex.1003, ¶¶264-265. As discussed above at [1.5.1], it would have been obvious for Call Agent 42 (“*call control gatekeeper*”) to receive a monitored event such as a quality alert message. It further would have been obvious for Call Agent 42 to make the codec-change decision. Ex.1003, ¶265. Such a modification would have been obvious to a POSITA in view of RFC3435, which explains that a gateway operates under the “call control ‘intelligence’” of the call agent. Ex.1027, 5; Ex.1003, ¶265. RFC3435 expressly contemplates the call agent sending an instruction to change a codec:

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.

The above quotation refers to sending a modify-connection command with an instruction to change codecs to a gateway. For a VoIP call between IP phones 20 and 44, however, a POSITA would have recognized that the codec implementation would not be on VoIP Gateways 16 and 34, but instead on the VoIP endpoints, IP Phones 20 and 44. Ex.1003, ¶266; Material Facts #6-7; Ex.1008, [0004], [0009]. Thus, it would have been obvious to send a modify-connection command with an instruction to change codecs (“*an adjustment signal*”) to IP phones 20 and 44 to cause them to use a different codec. Ex.1003, ¶266. The obviousness of the combination is confirmed by the fact that this approach to changing codecs had been known in the art for years. 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...**”); Ex.1003, ¶266.

In summary, the combination of Mundra and RFC3435 renders obvious for Call Agent 42 to respond to information about changing network conditions by sending a “modify connection” command (“*an adjustment signal based on the call*

quality signal”) to IP phones 20 and 44 (*“to the first and second VoIP telephones”*) to use a different codec (*“instructing the first and second VoIP telephones to communicate using a second codec”*). Ex.1003, ¶267.

[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.

As discussed at [1.4], it would have been obvious for there to be a VoIP telephone call between IP phones 20 and 44 (*“a telephone call between the first and second VoIP telephones”*). Ex.1003, ¶¶268-269.

As discussed at [1.5.2], Mundra and RFC3435 render obvious Call Agent 42 responding to information about network conditions by sending an instruction to change codec (*“the adjustment signal based on the call quality signal”*) to IP phones 20 and 44 (*“is sent to the first and second VoIP telephones”*). Ex.1003, ¶270.

Mundra discloses “changing the first codec to the second codec in realtime **during transmission of the voice call.**” Ex.1008, claim 1; *see also* [0016] (“select an appropriate codec during a voice over IP call”), [0020] (“change the codecs in realtime during a call transmission”), Abstract (“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.”). Thus, Mundra renders

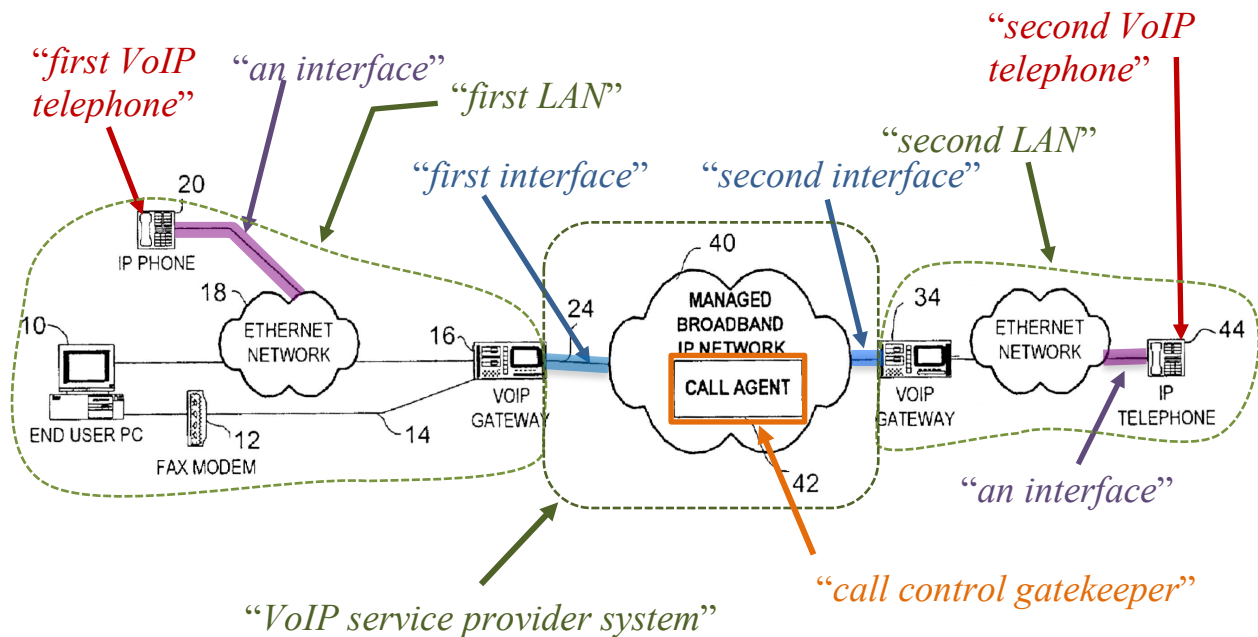
obvious that “*the adjustment signal... is sent... during a telephone call*” as claimed. Ex.1003, ¶¶271-272.

Alternatively, RFC3435 further describes how the “Call Agent” can “change the codec” “at any time” after a connection has been established. Ex.1027, 17.

Thus, RFC3435 further confirms the obviousness of sending an “*adjustment signal*” to change a codec for an already-established connection between IP phones 20 and 44 (“*during a telephone call between the first and second VoIP telephones*”). Ex.1003, ¶¶273-274. In the combination of Mundra and RFC3435, it would have been obvious for Call Agent 42 to send a “modify connection” command (“*an adjustment signal*”) on an established connection (“*during a telephone call*”). Ex.1003, ¶274-275.

6. Summary

Mundra’s Fig. 2 embodiment, modified and combined with RFC3435 and RFC3660 as discussed above, renders obvious claim 1:



Ex.1003, ¶276; 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).

7. Claim 2

Claim 2 is identical to claim 1—and thus obvious as discussed above—except for its final limitation, analyzed below. Ex.1003, ¶¶277-283.

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

As discussed at [1.5.1], Mundra’s monitored event is a “call quality signal.” Mundra further describes monitoring a VoIP call for “packet loss,” which was a well-known call quality metric. Ex.1008, [0022]-[0023], [0006] (“lost voice packets”), [0011] (“call quality degrades...due to...lost packets”); Material Fact

#12. Thus, it would have been obvious for Mundra’s monitored event to be triggered by the number of lost packets (“*based upon a number of dropped voice packets during a VoIP telephone call*”). Ex.1003, ¶¶285-286 (citing Exs.1091-1094 regarding synonymy of “dropped” and “lost” packets).

Additionally, as discussed at [1.5.1], an MGCP quality alert also corresponds to the claimed “*call quality signal*.” Ex.1003, ¶287. RFC3660 explains that a quality alert is generated based on observed “packet loss rate,” which would obviously be based on the “*number of dropped voice packets*.” Ex.1028, 46; Ex.1003, ¶¶287-288 (citing Exs.1095-1097 showing how a packet loss rate depends on the number of lost packets). Thus, the combined prior art renders obvious a quality alert (“*call quality signal*”) based on the number of lost packets (“*based upon a number of dropped voice packets during a VoIP telephone call*”). Ex.1003, ¶289.

X. CONCLUSION

Petitioner has established a reasonable likelihood that the Challenged Claims are unpatentable.

Respectfully submitted,

Dated: August 13, 2024
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XI. MANDATORY NOTICES

A. Real Party-in-Interest

Pursuant to 37 C.F.R. § 42.8(b)(1), Petitioner certifies that the real party-in-interest is Cisco Systems, Inc.

B. Related Matters

Pursuant to 37 C.F.R. § 42.8(b)(2), to the best knowledge of the Petitioner, the '368 patent is involved in the following cases:

Case Heading	Number	Court	Date
<i>Croga Innovations Ltd. v. Cisco Systems, Inc.</i>	2-24-cv-65	EDTX	February 1, 2024

C. Lead and Back-up Counsel and Service Information

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Please address all correspondence to lead and back-up counsel. Petitioner consents to service in this proceeding by email at the addresses above.

XII. CLAIMS APPENDIX

[1.0] 1. A system comprising:

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

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

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

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

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

[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

[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.

[2.0] 2. A system comprising:

- [2.1] a VoIP service provider system having a call control gatekeeper, a first interface and a second interface;
- [2.2] the first interface connected to a first LAN having an interface to a first VoIP telephone;
- [2.3] the second interface connected to a second LAN having an interface to a second VoIP telephone;
- [2.4] the first and second VoIP telephones being configured to communicate using a first codec;
- [2.5.1] the call control gatekeeper configured to receive a call quality signal from the first VoIP telephone and
- [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
- [2.6] wherein the call quality signal is based upon a number of dropped voice packets during a VoIP telephone call.

CERTIFICATE OF WORD COUNT

Pursuant to 37 C.F.R. § 42.24(d), Petitioner hereby certifies, in accordance with and in reliance on the word count provided by the word-processing system used to prepare this Petition, that the number of words in this paper is 13,985.

Pursuant to 37 C.F.R. § 42.24(d), this word count excludes the table of contents, table of authorities, mandatory notices under § 42.8, certificate of service, certificate of word count, appendix of exhibits, and any claim listing.

Dated: August 13, 2024

/Theodore M. Foster/

Theodore M. Foster
Lead Counsel for Petitioner
Registration No. 57,456

CERTIFICATE OF SERVICE

The undersigned certifies that, in accordance with 37 C.F.R. § 42.6(e) and 37 C.F.R. § 42.105, service was made on Patent Owner as detailed below.

Date of service August 13, 2024

Manner of service FEDERAL EXPRESS

Documents served Petition for *Inter Partes* Review Under 35 U.S.C. § 312 and 37 C.F.R. § 42.104 of U.S. 7,738,368; Petitioner's Exhibit List; All Exhibits; Petitioner's Power of Attorney.

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