

**IN THE UNITED STATES DISTRICT COURT
FOR THE EASTERN DISTRICT OF TEXAS
MARSHALL DIVISION**

ASPEN NETWORKS, INC., a California
Corporation,

Plaintiff,

v.

AT&T INC., a Delaware Corporation, and
AT&T MOBILITY, LLC, a Delaware
Corporation,

Defendants.

Case No.: 2:23-cv-00476-RWS-RSP

JURY TRIAL DEMANDED

LEAD CASE

ASPEN NETWORKS, INC., a California
Corporation,

Plaintiff,

v.

CELLCO PARTNERSHIP D/B/A VERIZON
WIRELESS, a Delaware Corporation,

Defendant.

Case No.: 2:23-cv-00557-RWS-RSP

ASPEN NETWORKS, INC., a California
Corporation,

Plaintiff,

v.

T-MOBILE US, INC., a Delaware
Corporation, and T-MOBILE USA, INC., a
Delaware Corporation,

Defendants.

Case No.: 2:23-cv-00476-RWS-RSP

**DEFENDANTS' P.R. 3-3 INVALIDITY CONTENTIONS AND SUBJECT MATTER
ELIGIBILITY CONTENTIONS**

TABLE OF CONTENTS

	Page
I. INTRODUCTION	6
II. ASSERTED PATENT AND CLAIMS	7
III. CLAIM CONSTRUCTION.....	8
IV. P.R. 3-3(A) – IDENTIFICATION OF PRIOR ART	9
A. Prior Art Patents.....	9
B. Prior Art Publications	14
C. Prior Art Sales/Public Uses.....	19
D. Prior Invention	20
V. P.R. 3-3(B) – ANTICIPATION AND OBVIOUSNESS.....	21
A. Primary References	22
1. U.S. Patent No. 8,019,335 (“Kallio”)	22
2. U.S. Patent No. 7,890,099 (“Mohammed”)	23
3. IP Mobility System (“IPM System”)	23
4. U.S. Patent No. 7,079,499 (“Akhtar”)	24
5. U.S. Patent Publication No. 2006/0276192 (“Dutta”)	25
6. U.S. Patent No. 7,957,348 (“Gallagher”).....	25
7. U.S. Patent Publication No. 2006/0193295 (“White”)	26
8. U.S. Patent Publication No. 2004/0233866 (“Bossoli”).....	27
9. U.S. Patent Publication No. 2009/0023450 (“George”)	27
10. U.S. Patent No. 7,184,418 (“Baba”)	28
11. U.S. Patent Publication No. 2005/0070288 (“Belkin”)	28
12. U.S. Patent Publication No. 2006/0187943 (“Oh”)	29
13. U.S. Patent Publication No. 2007/0183440 (“Bennet”).....	30

14.	U.S. Patent Publication No. 2005/0025163 (“Christie”)	30
15.	U.S. Patent Publication No. 2007/0015536 (“LaBauve”).....	31
16.	U.S. Patent No. 8,089,934 (“Long”).....	31
17.	U.S. Patent No. 8,467,377 (“Kottilingal”).....	32
18.	U.S. Patent Publication No. 2004/0028009 (“Dorenbosch”).....	33
19.	U.S. Patent Publication No. 2004/0199662 (“Karol”).....	33
20.	Korean Patent No. 10-0741379 (“Kwon”).....	34
21.	U.S. Patent No. 6,985,479 (“Leung”).....	34
22.	U.S. Patent Publication No. 2008/0069065 (“Wu”)	35
23.	U.S. Patent Publication No. 2008/0148379 (“Xu”)	36
24.	Japan Patent Publication No. JP2007/142786A (“Yokoyama”).....	36
25.	Unlicensed Mobile Access (UMA) protocols – Stage 1-3 Specifications (“UMA Protocols”).....	37
26.	U.S. Patent Publication No. 2005/0286466 (“Tagg”).....	38
27.	EP Patent Publication No. 1811741A1 (“Oberle”).....	38
B.	State of the Art and Obviousness.....	39
1.	Motivations for Making Combinations of Prior Art.....	41
2.	Specific Motivations	44
a.	Finite number of predictable solutions	45
b.	Inherent motivation to combine	47
c.	Overlapping authorship, inventorship, or assignees	49
d.	References from the same field.....	49
e.	Motivations based on admitted prior art	49
3.	State of the Art and Motivations to Combine	53
a.	“switching the transmission from the first network path to one of a plurality of alternative network paths”	54

b.	“Real Time Protocol (RTP)” and “Session Initiation Protocol (SIP)” and “SIP signaling sequence”	57
4.	Exemplary Combinations.....	63
a.	Combinations Based on Kallio	63
b.	Combinations Based on Mohammed.....	80
c.	Combinations Based on IPM System	96
d.	Combinations Based on Akhtar	113
e.	Combinations Based on Dutta	128
f.	Combinations Based on Gallagher.....	144
g.	Combinations Based on White.....	160
h.	Combinations Based on Bossoli	177
i.	Combinations Based on George.....	193
j.	Combinations Based on Baba	209
k.	Combinations Based on Belkin.....	226
l.	Combinations Based on Oh	242
m.	Combinations Based on Bennet.....	259
n.	Combinations Based on Christie.....	275
o.	Combinations Based on LaBauve.....	291
p.	Combinations Based on Long.....	308
q.	Combinations Based on Kottilingal	324
r.	Combinations Based on Dorenbosch	341
s.	Combinations Based on Karol	358
t.	Combinations Based on Kwon.....	375
u.	Combinations Based on Leung	392
v.	Combinations Based on Wu.....	409
w.	Combinations Based on Xu	426

x.	Combinations Based on Yokohama.....	443
y.	Combinations Based on the UMA Protocols	460
z.	Combinations Based on Tagg	477
aa.	Combinations Based on Oberle	493
5.	Secondary Considerations.....	509
VI.	P.R. 3-3(C) – CLAIM CHARTS	510
VII.	P.R. 3-3(D) – INVALIDITY UNDER 35 U.S.C. § 112.....	511
A.	Invalidity Based on Indefiniteness, Enablement, and Written Description	511
VIII.	PATENT SUBJECT MATTER INELIGIBILITY CONTENTIONS	518
IX.	DOCUMENT PRODUCTION ACCOMPANYING INVALIDITY CONTENTIONS.....	521

Pursuant to P.R. 3-3 and the Court's Docket Control Order (Dkt. 34, Case No. 2:23-CV-00476-RWS-RSP), Defendants AT&T Inc., AT&T Mobility, LLC, Cellco Partnership d/b/a Verizon Wireless, T-Mobile US, Inc. and T-Mobile USA, Inc. ("Defendants") hereby provide these P.R. 3-3 Invalidity Contentions ("Invalidity Contentions") to Plaintiff Aspen Networks, Inc. ("Plaintiff" or "Aspen").

I. INTRODUCTION

These Invalidity Contentions are based on Defendants' current knowledge and understanding of U.S. Patent No. 8,009,554 ("the '554 Patent" or "the Asserted Patent") and the prior art, along with Defendants' understanding of Plaintiff's infringement allegations and apparent claim interpretations based on those allegations, as set forth in Plaintiff's Disclosure of Asserted Claims & Infringement Contentions ("Infringement Contentions"). These Invalidity Contentions should not be construed as a waiver by Defendants of any objections to Plaintiff's Infringement Contentions or as an admission that the Infringement Contentions were adequate or that Plaintiff's apparent claim interpretations therein are correct, have merit, or should be adopted. Additionally, and as further explained below, where these Invalidity Contentions disclose that a prior art reference meets a claim limitation, such position may be based on Plaintiff's apparent claim interpretations in its Infringement Contentions; Defendants do not necessarily agree that such interpretations are correct, have merit, or should be adopted.

Defendants' investigation of the Asserted Patent, its file history, and the prior art is ongoing, and Defendants expressly reserve the right to supplement these Invalidity Contentions based on further discovery and in a manner consistent with the Federal Rules of Civil Procedure, the Local Patent Rules for the Eastern District of Texas, and any Docket Control Order, or otherwise in accordance with any order or direction of this Court or agreement of the parties. Defendants' ongoing investigation includes, but is not limited to, serving subpoenas regarding

prior art and inventors of prior art, seeking additional information related to the references and prior art systems disclosed in these Invalidity Contentions, and seeking additional information regarding Plaintiff's Infringement Contentions and the products and services accused of infringing therein. Additionally, Defendants have not yet had the benefit of taking the deposition of the named inventors or those that worked with the named inventors, or of the named inventors, authors, and entities listed on any references or systems identified in these Invalidity Contentions.

II. ASSERTED PATENT AND CLAIMS

Plaintiff asserts the following claims from the '554 Patent (with independent claims identified in **bold**): **1**, **2**, **6**, **7**, and **15** – **20** ("the Asserted Claims"). Plaintiff asserts a priority date for the '554 Patent of March 13, 2008—the date of filing for U.S. Provisional Patent Application No. 61/036,377 for which the '554 Patent allegedly claims priority to. Reference to an "asserted priority date" should not be construed to mean that Defendants agree that the Asserted Patent is in fact entitled to such priority date, or that Plaintiff has provided proper notice as to its contentions for a priority date.

P.R. 3-2(b) requires Plaintiff to produce, contemporaneously with its Disclosure of Asserted Claims and Infringement Contentions," "[a]ll documents evidencing the conception, reduction to practice, design, and development of each claimed invention, which were created on or before the date of application for the patent in suit or the priority date identified pursuant to P.R. 3-1(e), whichever is earlier." Based on Plaintiff's production purportedly pursuant to P.R. 3-2(b), Defendants understand that Plaintiff is not asserting an invention date, or conception, reduction to practice, design, and/or development of the claimed invention, prior to March 13, 2008. To the extent Plaintiff later alleges that any such activities occurred prior to March 13, 2008, Defendants reserve the right to supplement these Invalidity Contentions to address any such contention, including by identifying and/or charting additional prior art that may become relevant

in view of such assertion. Likewise, to the extent Plaintiff successfully establishes an invention date before any of the prior-art references relied on by Defendants, then those references serve as evidence of secondary considerations of obviousness, particularly, contemporaneous invention by others. To the extent Plaintiff alleges that any prior art relied on in these Invalidity Contentions does not actually qualify as prior art to the Asserted Patent, Defendants reserve the right to rebut those allegations (*e.g.*, by demonstrating an earlier critical date for the challenged prior art and/or a later priority date and/or date of invention for a particular Asserted Claim). Defendants also reserve the right to rely on additional documents, information, and evidence to demonstrate public availability or public availability by a certain date of a prior art reference or to rebut any efforts by Plaintiff to allege any reference was not publicly available or publicly available by a certain date or otherwise available as prior art.

Defendants reserve the right to amend or supplement these disclosures as necessary in view of any changes or amendments made, for any reason, to Plaintiff's infringement theories, infringement contentions, or Asserted Claims, including to Plaintiff's asserted priority and/or invention date(s).

III. CLAIM CONSTRUCTION

To the extent that these Invalidity Contentions rely on or otherwise embody particular constructions of terms or phrases in the Asserted Claims, Defendants are not conceding to any such constructions as proper constructions of those terms or phrases at this time. For purposes of these Invalidity Contentions, Defendants may adopt alternative claim construction positions. In particular, portions of these Invalidity Contentions, including the claim charts attached as exhibits, may be based on the underlying claim constructions and/or interpretations in Plaintiff's Infringement Contentions, to the extent such constructions and/or interpretations are understood

by Defendants. Defendants do not concede that any such constructions or interpretations by Plaintiff are proper, and expressly reserves the right to contest any such constructions.

Throughout the attached Exhibits, Defendants provide examples of where references disclose subject matter recited in preambles, without regard to whether the preambles are properly considered to be limitations of the Asserted Claims. Defendants reserve the right to argue, at the appropriate stage of this case, that the preambles are or are not limitations. Further, the fact that Defendants provide examples of any limitation does not mean that Defendants believe that the limitation is defined, or otherwise enabled, or comports with the requirements of the patent statute, including, but not limited to, 35 U.S.C. § 112.

Nothing disclosed herein is an admission or acknowledgement that any of Defendants' products or services infringes any of the Asserted Claims. Defendants reserve the right to supplement, modify, or otherwise amend these Invalidity Contentions, including based on the Court's claim construction rulings or opinions or the parties' positions or arguments taken in connection with the claim construction process. Moreover, Defendants reserve the right to rely on any disclosures in the accompanying P.R. 3-4(b) production as it believes necessary in view of positions Plaintiff may take with respect to the scope and meaning of the Asserted Claims.

IV. P.R. 3-3(A) – IDENTIFICATION OF PRIOR ART¹

A. Prior Art Patents

Patent / Publication No.	Country of Origin	Date of Issue /Publication²
U.S. 2003/0156559	United States	August 21, 2003
U.S. 2006/0193265	United States	May 26, 2009

¹ Defendants also hereby identify any systems or products that embody the technology described in any patent or publication identified in these Invalidity Contentions. Defendants reserve the right to rely on any documents or other evidence regarding any such systems.

² The dates in this column are the latest dates on which the patents or publications were publicly available. Defendants reserve the right to identify earlier dates should such dates be ascertained through discovery.

Patent / Publication No.	Country of Origin	Date of Issue /Publication²
U.S. 2005/0286466	United States	May 12, 2005
U.S. 2007/0253413	United States	November 1, 2007
U.S. 2008/0130574	United States	February 15, 2011
U.S. 2006/027612	United States	February 16, 2010
U.S. 7,957,348	United States	June 7, 2011
U.S. 7,890,099	United States	February 15, 2011
U.S. 6,922,728	United States	July 26, 2005
U.S. 8,019,335	United States	September 13, 2011
U.S. 7,079,499	United States	July 18, 2006
U.S. 2006/0193295	United States	May 26, 2009
U.S. 2004/0233866	United States	March 18, 2008
U.S. 2009/0023450	United States	May 13, 2014
U.S. 7,184,418	United States	February 27, 2007
U.S. 2005/0070288	United States	March 31, 2005
U.S. 2006/0187943	United States	August 24, 2006
U.S. 2007/0183440	United States	August 9, 2007
U.S. 2005/0025163	United States	February 3, 2005
U.S. 2007/0015536	United States	January 18, 2007
U.S. 8,467,377	United States	June 18, 2013
U.S. 2004/0028009	United States	February 12, 2004
U.S. 2004/0199662	United States	October 7, 2004
10-0741379	Korea	July 20, 2007
U.S. 6,985,479	United States	January 10, 2006
U.S. 2008/0069065	United States	March 20, 2008
U.S. 2008/0148379	United States	June 19, 2008
U.S. 2007/031667	JP	February 8, 2007
EP 0696153	EPO	November 16, 2005
EP 0700227	EPO	March 6, 1996
EP 0766490	EPO	April 2, 1997
EP 0936777	EPO	August 13, 2003
EP 1811741	EPO	July 25, 2007
GB 2,336,069	United Kingdom	October 6, 1999
JP 2000-278737	Japan	October 6, 2000
JP 2001-144815	Japan	May. 25, 2001

Patent / Publication No.	Country of Origin	Date of Issue /Publication²
JP 6-104827	Japan	April 15, 1994
JP 7-30952	Japan	January 31, 1995
JP 8-140132 A	Japan	May. 31, 1995
U.S. 2001/0014614	United States	August 16, 2001
U.S. 2002/0085516	United States	July 4, 2002
U.S. 2002/0147820	United States	October 10, 2002
U.S. 2002/0147820	United States	October 10, 2002
U.S. 4,989,230	United States	January 29, 1991
U.S. 5,353,331	United States	October 4, 1994
U.S. 5,406,615	United States	April 11, 1995
U.S. 5,659,598	United States	August 19, 1997
U.S. 5,737,703	United States	April 7, 1998
U.S. 5,758,281	United States	May 26, 1998
U.S. 5,771,353	United States	June 23, 1998
U.S. 5,796,727	United States	August 18, 1998
U.S. 5,805,689	United States	September 8, 1998
U.S. 5,815,811	United States	September 29, 1998
U.S. 5,842,122	United States	November 24, 1998
U.S. 5,845,211	United States	December 1, 1998
U.S. 5,870,673	United States	February 9, 1999
U.S. 5,873,033	United States	February 16, 1999
U.S. 5,890,064	United States	March 30, 1999
U.S. 5,911,120	United States	June 8, 1999
U.S. 5,930,712	United States	July 27, 1999
U.S. 5,949,775	United States	September 7, 1999
U.S. 5,956,331	United States	September 21, 1999
U.S. 5,970,059	United States	October 19, 1999
U.S. 6,073,015	United States	June 6, 2000
U.S. 6,075,783	United States	June 13, 2000
U.S. 6,097,942	United States	August 1, 2000
U.S. 6,097,948	United States	August 1, 2000
U.S. 6,112,088	United States	August 29, 2000
U.S. 6,141,547	United States	October 31, 2000
U.S. 6,161,008	United States	December 12, 2000

Patent / Publication No.	Country of Origin	Date of Issue /Publication²
U.S. 6,163,701	United States	December 19, 2000
U.S. 6,181,938	United States	January 30, 2001
U.S. 6,198,941	United States	March 6, 2001
U.S. 6,215,779	United States	April 10, 2001
U.S. 6,243,581	United States	June 5, 2001
U.S. 6,285,879	United States	September 4, 2001
U.S. 6,327,470	United States	December 4, 2001
U.S. 6,363,151	United States	March 26, 2002
U.S. 6,411,807	United States	June 25, 2002
U.S. 6,430,395	United States	August 6, 2002
U.S. 6,477,644	United States	November 5, 2002
U.S. 6,484,211	United States	November 19, 2002
U.S. 6,484,211	United States	November 19, 2002
U.S. 6,526,034	United States	February 25, 2003
U.S. 6,539,237	United States	March 25, 2003
U.S. 6,542,497	United States	April 1, 2003
U.S. 6,546,253	United States	April 8, 2003
U.S. 6,546,425	United States	April 8, 2003
U.S. 6,577,643	United States	June 10, 2003
U.S. 6,600,924	United States	July 29, 2003
U.S. 6,603,761	United States	August 5, 2003
U.S. 6,608,832	United States	August 19, 2003
U.S. 6,611,516	United States	August 26, 2003
U.S. 6,625,451	United States	September 23, 2003
U.S. 6,658,253	United States	December 2, 2003
U.S. 6,680,923	United States	January 20, 2004
U.S. 6,681,259	United States	January 20, 2004
U.S. 6,687,243	United States	February 3, 2004
U.S. 6,711,146	United States	March 23, 2004
U.S. 6,714,987	United States	March 30, 2004
U.S. 6,729,929	United States	May 4, 2004
U.S. 6,731,621	United States	May 4, 2004
U.S. 6,751,204	United States	June 15, 2004
U.S. 6,769,000	United States	July 27, 2004

Patent / Publication No.	Country of Origin	Date of Issue /Publication²
U.S. 6,826,414	United States	November 30, 2004
U.S. 6,853,851	United States	February 8, 2005
U.S. 6,885,869	United States	April 26, 2005
U.S. 6,909,705	United States	June 21, 2005
U.S. 6,965,948	United States	November 15, 2005
U.S. 6,970,474	United States	November 29, 2005
U.S. 6,970,719	United States	November 29, 2005
U.S. 6,993,359	United States	January 31, 2006
U.S. 7,009,952	United States	March 7, 2006
U.S. 7,010,305	United States	March 7, 2006
U.S. 7,020,472	United States	March 28, 2006
U.S. 7,039,027	United States	May 2, 2006
U.S. 7,058,059	United States	June 6, 2006
U.S. 7,079,499	United States	July 18, 2006
U.S. 7,120,129	United States	October 10, 2006
U.S. 7,146,186	United States	December 5, 2006
U.S. 7,177,952	United States	February 13, 2007
U.S. 7,184,782	United States	February 27, 2007
U.S. 7,200,401	United States	April 3, 2007
U.S. 7,289,478	United States	October 30, 2007
U.S. 7,293,107	United States	November 6, 2007
U.S. 7,372,868	United States	May 13, 2008
U.S. 7,379,436	United States	May 27, 2008
U.S. 7,486,966	United States	February 3, 2009
U.S. 7,502,626	United States	March 10, 2009
U.S. 7,561,547	United States	July 14, 2009
U.S. 7,782,848	United States	August 24, 2010
U.S. 7,882,247	United States	February 1, 2011
U.S. 7,890,099	United States	February 15, 2011
U.S. 8,019,335	United States	September 13, 2011
U.S. 8,144,728	United States	March 27, 2012
U.S. 8,670,405	United States	March 11, 2014
U.S. 8,862,751	United States	October 14, 2014
U.S. 9,049,042	United States	June 2, 2015

Patent / Publication No.	Country of Origin	Date of Issue /Publication ²
U.S. Inv. Reg. H1641	United States	April 1, 1997
WO 95/23491	Japan	August 31, 1995
WO 98/02008	United Kingdom	January 15, 1998
WO 98/43446	Sweden	October 1, 1998
WO 99/48311	Finland	September 23, 1999
WO 99/48312	United States	September 23, 1999
WO 99/48315	United Kingdom	September 23, 1999
WO 99/48320	Finland	September 23, 1999
WO 00/79813	United States	December 28, 2000
WO 00/79814	United States	December 28, 2000

B. Prior Art Publications

Title, Author(s), and Publisher	Date of Publication ³
Unlicensed Mobile Access (UMA) protocols – Stage 1-3 Specifications	September, 2004
3GPP TR 123 923 v3.0.0	May 2001
3GPP TR 23.922 v.1.0.0	October 1999
3GPP TS 123 002 v3.3.0	March 2000
3GPP TS 123 002 v3.4.0	December 2000
3GPP TS 123 008 v3.5.0	September 2000
3GPP TS 123 018 v3.6.0	September 2000
3GPP TS 123 088 v3.2.0	October 2000
3GPP TS 23.288 v.2.0.0	March 2001
Akyildiz et al., <i>Mobility Management in Current and Future Communications Networks</i> , IEEE, pp. 39-49	August 1998
Albrecht et al., <i>IP Services over Bluetooth: Leading the Way to a New Mobility</i> , IEEE	1999
ANSI/IEEE Standard 802.11 (IEEE)	1999
ANSI/IEEE Standard 802.11a (IEEE)	1999
ANSI/IEEE Standard 802.11b (IEEE)	1999
Baker et al., <i>Supporting Mobility in MosquitoNet</i> (Proceedings of the 1996 USENIX Technical Conference)	January 1996

³ The dates in this column are the latest dates on which the publications were publicly available. Defendants reserve the right to identify earlier dates should such dates be ascertained through discovery.

Title, Author(s), and Publisher	Date of Publication ³
Baker, <i>Changing Communication Environments in MosquitoNet</i> , 1994 First Workshop on Mobile Computing Systems and Applications, pp. 64-68 (IEEE)	1995
Balakrishnan, <i>Challenges to Reliable Data Transport over Heterogeneous Wireless Networks</i> , UC Berkeley Doctoral Dissertation (R. Katz, Chair) (1998)	1998
Bay Area Research Wireless Access Network (BARWAN) website, available at http://bnrg.eecs.berkeley.edu/~randy/Daedalus/BARWAN/BARWAN_index.html	1996
Becker, Carey B., Patil, Basavaraj, Qaddoura, Emad, IP Mobility Architecture Framework (draft-ietf-mobileip-ipm-arch-00.txt), Wireless Technology Labs, Nortel Networks	1999
Bennington et al., <i>Wireless Andrew: Experience Building a High Speed, Campus-Wide Wireless Data Network</i> , MOBICOM'97 Budapest	1997
Brewer et al., <i>A Network Architecture for Heterogeneous Mobile Computing</i> , IEEE Personal Communications 5(5), pp. 8-24	1998
Brewer et al., <i>A Network Architecture for Heterogeneous Mobile Computing</i> , UC Berkeley - DRAFT, pp. 1-24	1998
Brown et al., <i>A mobile multimedia test-bed with application and network independent roaming capability</i> (Acts Mobile Communications Summit '98 – - Conference Proceedings)	January 14, 1999
Buchanan et al., <i>IMT-2000: Service Provider's Perspective</i> , IEEE Personal Communications, pp. 8-13	August 1997
Caceres et al., <i>Fast and Scalable Handoffs for Wireless Internetworks</i> , ACM, pp. 56-66	1996
Castro, <i>The UMTS Network and Radio Access Technology – Air Interface Techniques For Future Mobile Systems</i> (John Wiley & Sons Ltd.)	2001
Cheshire et al., <i>A Wireless Network in MosquitoNet</i> , IEEE Micro 16(1), pp. 44-52 (IEEE)	1996
ETSI EN 300 175-1 v1.4.2	June 1999
Festag et al., <i>Current developments and trends in handover design for APP-IP wireless networks</i> , Telecommunication Networks Group, Version 1.3	August 2000
Francis et al., <i>Towards Global Multimedia Mobility</i> , pp. 371-376	1997
Granberg, <i>GSM on the Net</i> , Ericsson Review No. 4, pp. 184-191	1998
Grube et al., <i>In-Building Wireless Coverage Using A Second Mode</i> , Motorola Inc., pp. 66-68	May 1996

Title, Author(s), and Publisher	Date of Publication ³
Guardini et al., <i>The Role of Internet Technology in the Future Mobile Data Systems</i> , Centro Studi E Laboratori Telecomunicazioni, pp. 1-7	2000
Haartsen, <i>Bluetooth – The Universal Radio Interface For Ad Hoc, Wireless Connectivity</i> , Ericsson Review No. 3, pp. 110-117	1998
Helal et al., <i>An Architecture for Wireless LAN/WAN Integration</i> , Computer and Information Science and Engineering Department	2000
Hills et al., <i>A Wireless Infrastructure to Support Research Activities</i> , Wireless Personal Communications, pp. 19-26 (Springer)	1996
Hills et al., <i>A Wireless Infrastructure to Support Research Activities</i> , Department of Engineering and Public Policy, Carnegie Mellon University, pp. 5-12	1996
Hills et al., <i>Wireless Data Network Infrastructure at Carnegie Mellon University</i> , IEEE Personal Communications 3(1), pp. 56-63	February 1996
Hills et al., <i>Wireless Data Network Infrastructure at Carnegie Mellon University</i> , Computer Science Department, Carnegie Mellon University, pp. 1-14	1996
Hills, <i>Large-Scale Wireless LAN Design</i> , IEEE Communications Magazine – Wireless Local Area and Home Networks (IEEE)	November 2001
Hills, <i>Wireless Andrew: Bringing mobile computing to a University community of 10,000</i> , IEEE 1999	1999
HomeRF Specification, HomeRF, Revision 2.0 (HomeRF Working Group)	April 4, 2001
Husain et al., <i>Intelligent Network: A Key Platform For PCS Interworking and Interoperability</i> , IEEE, pp. 98-105	September 1996
IETF RFC 2002 - IP Mobility Support	October 1996
IETF RFC 2543 – SIP: Session Initiation Protocol	March 1999
IETF RFC 2977 - Mobile IP Authentication, Authorization, and Accounting Requirements	October 2000
IETF RFC 3261 – SIP: Session Initiation Protocol	June 2002
IETF RFC 3550 – RTP: A Transport Protocol for Real-Time Applications	July 2003
Johnson et al., <i>DSR: The Dynamic Source Routing Protocol for Multihop Wireless Ad Hoc Networks</i> , Ad Hoc Networking, pp. 139-172	2001
Johnson et al., <i>DSR: The Dynamic Source Routing Protocol for Multihop Wireless Ad Hoc Networks</i> , Computer Science Department, Carnegie Mellon University, pp. 1-25	2001
Johnson et al., <i>Protocols for Adaptive Wireless and Mobile Networking</i> , IEEE Personal Communications	February 1996

Title, Author(s), and Publisher	Date of Publication ³
Johnson et al., <i>Protocols for Adaptive Wireless and Mobile Networking</i> , Computer Science Department, Carnegie Mellon University, pp. 1-18	1996
Kai et al., <i>Experiences with a Mobile Testbed</i> , 1998 Worldwide Computing and Its Applications, pp. 222-237	1998
Katz et al., <i>Chapter 23: The Case for Wireless Overlay Networks</i> (Proceedings of SPIE), pp. 1-12 (Not Published)	1996
Katz et al., <i>The Bay Area Research Wireless Access Network (BARWAN)</i> (Electrical Engineering and Computer Science Department – University of California, Berkeley)	January 1996
Katz et al., <i>The Bay Area Research Wireless Access Network (BARWAN)</i> (Electrical Engineering and Computer Science Department – University of California, Berkeley)	1996
Katz et al., <i>The Case for Wireless Overlay Networks</i> (Proceedings of SPIE)	January 1996
Katz, <i>Towards a Bay Area Research Wireless Access Network (BARWAN)</i> , available at http://bnrg.eecs.berkeley.edu/~randy/Daedalus/BARWAN/BARWAN.pdf	1996
Khalil, Pillai, <i>Architecture for IP Mobility</i> , IP Mobility Group, Nortel Networks, 2221 Lakeside Boulevard, Richardson, TX	2001
Lai et al., <i>Experiences with a Mobile Testbed</i> , Stanford University	1998
Langer et al., <i>CDMA2000 – A World View</i> , Ericsson Review No. 3, pp. 150-158	2001
Lucent Technologies, Bell Labs Innovations, <i>WaveLAN/ISA Card User's Guide</i> , pp. 1-87	September 1996
Lucent Technologies, Bell Labs Innovations, <i>WaveLAN/PCMCIA Card User's Guide</i>	April 1998
Meggers et al., <i>Roaming Between GSM and Wireless LAN</i> , Mobile Communications, pp. 828-834	November 1996
Mohr, <i>Broadband Radio Access For IP-Based Networks in the IST BRAIN Project</i> , ICT, pp. 22-25	2000
Nikolaou et al., <i>Wireless Technologies Convergence: Results and Experience</i> (IEEE)	2000
Pahlavan et al., <i>Handoff in Hybrid Mobile Data Networks</i> (IEEE Personal Communications)	April 2000
Pichna et al., <i>Wireless Internet – IMT-2000/Wireless LAN Interworking</i> , Journal of Communications and Networks, Vol. 2, No. 1, pp. 46-57	March 2000

Title, Author(s), and Publisher	Date of Publication ³
Qaddoura, <i>IP Mobility “Always on, Everywhere”</i> Nortel Networks, Sprint Research Symposium (“Sprint Presentation”)	March 8-9, 2000
Redi et al., <i>Mobile IP: A Solution for Transparent, Seamless Mobile Computer Communications</i> (Fuji-Keizai’s Report on Upcoming Trends in Mobile Computing and Communications)	1998
Salberg, <i>WLAN – GPRS Interworking</i> , Graduate Thesis, Siv.ing. Degree Information and Communication Technology, pp. 1-50	May 2001
Skolt et al., <i>Intelligent Networks as a platform for provision of service in GSM and DECT</i> , pp. 82-87	1995
Specification of The Bluetooth System - Core, Volume 1.0B	December 1, 1999
Specification of The Bluetooth System - Profiles, Volume 1.0B	December 1, 1999
Stemm et al., <i>Vertical Handoffs in Wireless Overlay Networks</i> (Mobile Networks and Applications)	1998
Stemm, <i>Vertical Handoffs in Wireless Overlay Networks</i> (Computer Science Division, Department of Electrical Engineering and Computer Science) pp. 1-29	1998
Tan et al., <i>A Fast Handoff Scheme For Wireless Networks</i> , ACM, pp. 83-90	1999
Taylor et al., <i>The Challenges of Seamless Handover in Future Mobile Multimedia Networks</i> (IEEE)	April 1999
The UC Berkeley BARWAN Research Project CDROM (Daedalus Index), available at http://daedalus.cs.berkeley.edu/index.html	1995-1998
Valkó, <i>Cellular IP: A New Approach to Internet Host Mobility</i> (Ericsson)	1999
Wang et al., <i>Policy-Enabled Handoffs Across Heterogeneous Wireless Networks</i> (WMCSA’99 Proceedings)	1999
Wietfeld et al., <i>Seamless IP-based Service Integration across Fixed/Mobile and Corporate/Public Networks</i> , IEEE, pp. 1930-1934	1999
Wireless Network Reference Model, TSB100-A (Revision of TSB100) (TIA/EIA)	March 2001
Yang, <i>CDMA RF System Engineering</i> (Artech House)	1998
Ylianttila et al., <i>Handoff Procedure For Heterogeneous Wireless Networks</i> (Global Telecommunications Conference - Globecom ’99)	1999
Yoshida, et al., <i>PHS Packet Data Communications Systems</i> , NTT Review, Vol. 9, No. 3, pp. 106-116	May 1997

C. Prior Art Sales/Public Uses

Item Offered for Sale and/or Publicly Used or Known	Date of Offer / Public Use / Knowledge ⁴	Person or Entity Who Made and Received Offer, Made Public Use, or Made Information Known or to Whom Made Known
802.11 System(s)	1999	IEEE and all working groups and individuals involved in 802.11 standardization
Bluetooth System(s)	1999	Bluetooth and all working groups and individuals involved in Bluetooth standardization
DECT System(s)	1999	ETSI and all working groups and individuals involved in DECT standardization
GSM/WaveLAN Testbed ⁵	1996	Lucent Technologies
GSM/WLAN Testbed ⁶	2000	National Technical University of Athens
HomeRF System(s)	1998	HomeRF
IP Mobility System (“IPM System”)	1999 (no later than early 2001)	Nortel Networks, Richardson, TX.
MosquitoNet ⁷	1996	Stanford University
NetMotion / NetMotion Mobility Solution	February 2001	NetMotion Wireless
The Bay Area Research Wireless Access Network (BARWAN) ⁸	1996	Electrical Engineering and Computer Science Department – University of California, Berkeley

⁴ The dates in this column are the latest dates on which the systems were publicly used or known. Defendants reserve the right to identify earlier dates should such dates be ascertained through discovery.

⁵ See, e.g., Brown et al., *A mobile multimedia test-bed with application and network independent roaming capability* (Acts Monjble Communications Summit '98 – Conference Proceedings 1999).

⁶ See, e.g., Nikolaou et al., *Wireless Technologies Convergence: Results and Experience* (IEEE 2000).

⁷ See, e.g., Baker et al., *Supporting Mobility in MosquitoNet* (Proceedings of the 1996 USENIX Technical Conference 1996); Baker, *Changing Communication Environments in MosquitoNet*, 1994 First Workshop on Mobile Computing Systems and Applications, pp. 64-68 (IEEE 1995); Cheshire et al., *A Wireless Network in MosquitoNet*, IEEE Micro 16(1), pp. 44-52 (IEEE 1996); Kai et al., *Experiences with a Mobile Testbed*, 1998 Worldwide Computing and Its Applications, pp. 222-237 (1998).

⁸ See, e.g., Katz et al., *The Bay Area Research Wireless Access Network (BARWAN)*, COMPCON'96 – Technologies for the Information Superhighway Digest of Papers, pp. 15-20 (IEEE 1996); Brewer et al., *A Network Architecture for Heterogeneous Mobile Computing*, IEEE Personal Communications 5(5), pp. 8-24 (1998); Katz et al., *The Case for Wireless Overlay Networks* in Mobile Computing, pp. 621-650 (Springer 1996); Stemm et al., *Vertical handoffs in wireless overlay networks*, Mobile Networks and Applications 3, pp. 335-350 (1998); R. Katz,

Item Offered for Sale and/or Publicly Used or Known	Date of Offer / Public Use / Knowledge⁴	Person or Entity Who Made and Received Offer, Made Public Use, or Made Information Known or to Whom Made Known
HotSpot @Home	October, 2006	T-Mobile USA, Inc.

D. Prior Invention

Prior Invention	Date of Prior Invention⁹	Person(s) Involved in and Circumstances Surrounding Prior Invention
802.11 System(s)	1999	IEEE and all working groups and individuals involved in 802.11 standardization
Bluetooth System(s)	1999	Bluetooth and all working groups and individuals involved in Bluetooth standardization
DECT System(s)	1999	ETSI and all working groups and individuals involved in DECT standardization
GSM/WaveLAN Testbed	1996	Lucent Technologies
GSM/WLAN Testbed	2000	National Technical University of Athens
HomeRF System(s)	1998	HomeRF
IP Mobility System	1999 (no later than early 2001)	Nortel Networks, Richardson, TX including inventors on Akhtar reference, Mohamed Khalil and Emad Pillai, inventors on Wurch patent,
Any systems described in the Lee '405 or Lee '705 prior art patents, including from Cellco Partnership, the original assignee of those patents	2000	Cellco Partnership, Jau Young Lee, Ce Xu, William Chien-Yeh Lee, Jau Young Lee
NetMotion / NetMotion Mobility Solution	February 2001	NetMotion Wireless

Towards a Bay Area Research Wireless Access Network (BARWAN), available at <http://bnrg.eecs.berkeley.edu/~randy/Daedalus/BARWAN/BARWAN.pdf>; Bay Area Research Wireless Access Network (BARWAN) website, available at http://bnrg.eecs.berkeley.edu/~randy/Daedalus/BARWAN/BARWAN_index.html; Balakrishnan, *Challenges to Reliable Data Transport over Heterogeneous Wireless Networks*, UC Berkeley Doctoral Dissertation (R. Katz, Chair) (1998).

⁹ The dates in this column are the latest dates on which prior invention occurred. Defendants reserve the right to identify earlier dates should such dates be ascertained through discovery.

Prior Invention	Date of Prior Invention⁹	Person(s) Involved in and Circumstances Surrounding Prior Invention
MosquitoNet	1996	Stanford University
The Bay Area Research Wireless Access Network (BARWAN)	1996	Electrical Engineering and Computer Science Department – University of California, Berkeley
UC Berkeley Handoff Testbed	1999	Electrical Engineering and Computer Science Department – University of California, Berkeley
Wireless Andrew / Monarch	1996	Carnegie Mellon University

V. P.R. 3-3(B) – ANTICIPATION AND OBVIOUSNESS

The primary prior art references that Defendants rely on for the Asserted Patent are identified below. The exhibits to these Invalidity Contentions contain claim charts for the primary prior art references. Obviousness combinations of the primary prior art references are separately identified throughout these Invalidity Contentions (including below and in Appendix A).

Defendants’ reliance on each prior art reference identified throughout these Invalidity Contentions (whether primary references or obviousness references) includes the reference itself, anything incorporated by the reference, any system embodying the reference, and any testimony by those with knowledge of the reference, such as named authors and inventors.¹⁰ All such documents and information shall be considered one prior art reference to the extent they describe a single prior art system or solution, or the closely related work of a single prior art author. In addition to the specific references below, Defendants reserve the right to rely on all admitted prior art in the Asserted Patent and its file history, as well as anything admitted as prior art by Plaintiff or its experts or representatives. Defendants reserve the right to supplement these Invalidity

¹⁰ For example, for each patent identified as a prior art reference in these Invalidity Contentions, Defendants reserve the right to rely on the patent itself and/or any system that practices or embodies the patent to demonstrate invalidity.

Contentions as discovery reveals further information about prior use and/or prior public knowledge of products, methods, or systems that Defendants have reason to believe anticipate or render obvious one or more asserted claims. Defendants further reserve the right to identify and rely on additional prior art in Plaintiff's possession, custody, or control that has not yet been produced or identified to Defendants, including prior art identified and/or produced in prior litigations or in post-grant proceedings at the U.S. Patent and Trademark Office.

A. Primary References

The following prior art references ("primary references"), alone or in combination with any other reference identified in this subsection, render invalid one or more of the Asserted Claims.

1. U.S. Patent No. 8,019,335 ("Kallio")

U.S. Patent No. 8,019,335 ("Kallio"), titled "Identifying Neighboring Cells in Telecommunications Network," was filed on January 29, 2001. Kallio qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (e), and (g). Kallio anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-1 provides representative, non-limiting examples of where Kallio discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Kallio does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Kallio alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Kallio with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

2. U.S. Patent No. 7,890,099 (“Mohammed”)

U.S. Patent No. 7,890,099 (“Mohammed”), titled “Method for Automatic and Seamless Call Transfers Between a Licensed Wireless System and an Unlicensed Wireless System,” was filed on April 2, 2002. Mohammed claims priority to provisional application nos. 60/271,766, 60/271,767, 60/271,768, and 60/271,769, each filed on February 26, 2001, which provide proper support for the disclosure relied on in Mohammed as well as for at least one claim of the issued Mohammed patent. Mohammed qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (e), and (g). Mohammed anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-2 provides representative, non-limiting examples of where Mohammed discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Mohammed does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Mohammed alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Mohammed with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

3. IP Mobility System (“IPM System”)

IPM System was developed by Nortel at facilities in Texas and was publicly used and/or demonstrated no later than 1999. IPM System qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (b), and (g). Features and aspects of IPM System are described in at least the following documents and sources:

- Additional IPM System materials charted in Exhibit A-3
- Akhtar charted at Exhibit A-4

Additional documents relating to IPM System have been produced concurrently herewith. Defendants intend to take further discovery on this system and supplement as appropriate.

IPM System anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim charts attached as Exhibits A-3 and A-4 provide representative, non-limiting examples of where IPM System discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that IPM System does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of IPM System alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify IPM System with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

4. U.S. Patent No. 7,079,499 (“Akhtar”)

U.S. Patent No. 7,079,499 (“Akhtar”), titled “Internet Protocol Mobility Architecture Framework,” was filed on September 7, 2000 claiming priority to a set of provisional applications disclosed on its face. Akhtar qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (e) and (g). Akhtar anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-4 provides representative, non-limiting examples of where Akhtar discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Akhtar does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Akhtar alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Akhtar with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section

V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

Moreover, as noted in the chart in Exhibit A-4, Akhtar expressly incorporates multiple different additional patent applications, patents, and documents such as RFC 2002 and RFC 2486. The chart in Exhibit A-4 explains in detail the reasons that the disclosure of those references would be applicable to Akhtar.

5. U.S. Patent Publication No. 2006/0276192 (“Dutta”)

U.S. Patent Publication No. 2006/0276192 (“Dutta”), titled “Seamless handoff across heterogeneous access networks using a handoff controller in a service control point,” was filed on May 18, 2006. Dutta qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(e) and (g). Dutta anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-5 provides representative, non-limiting examples of where Dutta discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Dutta does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Dutta alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Dutta with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

6. U.S. Patent No. 7,957,348 (“Gallagher”)

U.S. Patent No. 7,957,348 (“Gallagher”), titled “Method and system for signaling traffic and media types within a communications network switching system,” was filed on April 20, 2005. Gallagher qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(e) and (g).

Gallagher anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-6 provides representative, non-limiting examples of where Gallagher discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Gallagher does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Gallagher alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Gallagher with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

7. U.S. Patent Publication No. 2006/0193295 (“White”)

U.S. Patent Publication No. 2006/0193295 (“White”), titled “Multi-access terminal with capability for simultaneous connectivity to multiple communication channels,” was filed on May 11, 2005. White qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(e) and (g). White anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-7 provides representative, non-limiting examples of where White discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that White does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of White alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify White with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

8. U.S. Patent Publication No. 2004/0233866 (“Bossoli”)

U.S. Patent Publication No. 2004/0233866 (“Bossoli”), titled “Method of software download with session mobility support in mobile communication systems,” was filed on May 17, 2004 and published on November 25, 2004. Bossoli qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(b) and (g). Bossoli anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-8 provides representative, non-limiting examples of where Bossoli discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Bossoli does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Bossoli alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Bossoli with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

9. U.S. Patent Publication No. 2009/0023450 (“George”)

U.S. Patent Publication No. 2009/0023450 (“George”), titled “System and method for handoff of session from VoIP interface to cellular interface of dual-mode device,” was filed on February 28, 2008 and published on January 22, 2009. George qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(e) and (g). George anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-9 provides representative, non-limiting examples of where George discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that George does not anticipate any Asserted Claim, such Asserted

Claim would be obvious in view of George alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify George with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

10. U.S. Patent No. 7,184,418 (“Baba”)

U.S. Patent No. 7,184,418 (“Baba”), titled “Method and system for signaling traffic and media types within a communications network switching system,” was filed on October 20, 2000 and issued February 27, 2007. Baba qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(b) and (g). Baba anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-10 provides representative, non-limiting examples of where Baba discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Baba does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Baba alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Baba with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

11. U.S. Patent Publication No. 2005/0070288 (“Belkin”)

U.S. Patent Publication No. 2005/0070288 (“Belkin”), titled “Handover method and apparatus,” was filed on September 29, 2003 and published on May 31, 2005. Belkin qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(b) and (g). Belkin anticipates and/or

renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-11 provides representative, non-limiting examples of where Belkin discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Belkin does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Belkin alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Belkin with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

12. U.S. Patent Publication No. 2006/0187943 (“Oh”)

U.S. Patent Publication No. 2005/0070288 (“Oh”), titled “Handoff system and method between different kinds of devices, SIP server and operational method of SIP server,” was filed on February 16, 2006 and published on August 24, 2006. Oh qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(b) and (g). Oh anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-12 provides representative, non-limiting examples of where Oh discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Oh does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Oh alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Oh with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

13. U.S. Patent Publication No. 2007/0183440 (“Bennet”)

U.S. Patent Publication No. 2007/0183440 (“Bennet”), titled “Communication device supporting both Internet and public switched telephone network telephony,” was filed on February 7, 2006 and published on August 9, 2007. Bennet qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a) and (g). Bennet anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-13 provides representative, non-limiting examples of where Bennet discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Bennet does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Bennet alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Bennet with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

14. U.S. Patent Publication No. 2005/0025163 (“Christie”)

U.S. Patent Publication No. 2007/0183440 (“Christie”), titled “Mobility in a multi-access communication network,” was filed on July 28, 2003 and published on February 3, 2005. Christie qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(b) and (g). Christie anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-14 provides representative, non-limiting examples of where Christie discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Christie does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Christie alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious

to combine or modify Christie with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

15. U.S. Patent Publication No. 2007/0015536 (“LaBauve”)

U.S. Patent Publication No. 2007/0183440 (“LaBauve”), titled “Handset registration in a dual-mode environment,” was filed on July 19, 2006 and published on January 18, 2007. LaBauve qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(b) and (g). LaBauve anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-15 provides representative, non-limiting examples of where LaBauve discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that LaBauve does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of LaBauve alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify LaBauve with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

16. U.S. Patent No. 8,089,934 (“Long”)

U.S. Patent No. 8,089,934 (“Long”), titled “Method and system for controlling a call handover between telecommunication networks,” was filed on September 4, 2007 and issued January 3, 2012. Long qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(e) and (g). Long anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-16 provides

representative, non-limiting examples of where Long discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Long does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Long alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Long with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

17. U.S. Patent No. 8,467,377 (“Kottilingal”)

U.S. Patent No. 8,467,377 (“Kottilingal”), titled “Interleaving VoIP/VIP transmission in multiple sessions to increase quality of service in mobile devices having multiple interfaces,” was filed on August 24, 2005 and issued June 18, 2013. Kottilingal qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(e) and (g). Kottilingal anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-17 provides representative, non-limiting examples of where Kottilingal discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Kottilingal does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Kottilingal alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Kottilingal with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

18. U.S. Patent Publication No. 2004/0028009 (“Dorenbosch”)

U.S. Patent Publication No. 2004/0028009 (“Dorenbosch”), titled “Method and apparatus for effecting a seamless handoff between IP connections,” was filed on August 6, 2002 and published on February 12, 2004. Dorenbosch qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (b), and (g). Dorenbosch anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-18 provides representative, non-limiting examples of where Dorenbosch discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Dorenbosch does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Dorenbosch alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Dorenbosch with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

19. U.S. Patent Publication No. 2004/0199662 (“Karol”)

U.S. Patent Publication No. 2004/0199662 (“Karol”), titled “System and method to improve the resiliency and performance of enterprise networks by utilizing in-built network redundancy,” was filed on April 2, 2003 and published on October 7, 2004. Karol qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102 (a), (b), (e), and (g). Karol anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-19 provides representative, non-limiting examples of where Karol discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Karol does not anticipate any Asserted

Claim, such Asserted Claim would be obvious in view of Karol alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Karol with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

20. Korean Patent No. 10-0741379 (“Kwon”)

Korean Patent No. 10-0741379 (“Kwon”), titled “Application switch for controlling session initiation protocol communication via at least two subscriber's lines and method of operating the application switch,” was filed on September 9, 2006 and issued July 20, 2007. Kwon qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a) and (g). Kwon anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-20 provides representative, non-limiting examples of where Kwon discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Kwon does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Kwon alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Kwon with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

21. U.S. Patent No. 6,985,479 (“Leung”)

U.S. Patent No. 6,985,479 (“Leung”), titled “Method and apparatus for processing internet protocol transmissions,” was filed on March 4, 2002 and issued January 10, 2006. Leung qualifies

as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (b), and (g). Leung anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-21 provides representative, non-limiting examples of where Leung discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Leung does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Leung alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Leung with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

22. U.S. Patent Publication No. 2008/0069065 (“Wu”)

U.S. Patent Publication No. 2008/0069065 (“Wu”), titled “Method of seamlessly roaming between multiple wireless networks using a single wireless network adaptor,” was filed on September 20, 2006 and published on March 20, 2008. Wu qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (e), and (g). Wu anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-22 provides representative, non-limiting examples of where Wu discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Wu does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Wu alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Wu with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section

V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

23. U.S. Patent Publication No. 2008/0148379 (“Xu”)

U.S. Patent Publication No. 2008/0148379 (“Xu”), titled “Session initiation and maintenance while roaming,” was filed on November 1, 2006 and published on June 19, 2008. Xu qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(e) and (g). Xu anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-23 provides representative, non-limiting examples of where Xu discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Xu does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Xu alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Xu with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

24. Japan Patent Publication No. JP2007/142786A (“Yokoyama”)

Japan Patent Publication No. JP2007/142786A (“Yokohama”), titled “Handover server, and mobile communication terminal communicable thereof,” was filed on November 18, 2005 and published on June 7, 2007. Yokohama qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a) and (g). Yokohama anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-24 provides representative, non-limiting examples of where Yokohama discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent

Plaintiff asserts that Yokohama does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Yokohama alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Yokohama with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

25. Unlicensed Mobile Access (UMA) protocols – Stage 1-3 Specifications (“UMA Protocols”)

The Unlicensed Mobile Access Protocols (“the UMA Protocols”) consist of three technical specifications that were published at least as of February 26, 2005. The UMA Protocols were known or used by others in the United States at least as early as February 26, 2005 and formed the basis for certain publicly available Wi-Fi calling systems, such as T-Mobile’s HotSpot @Home system. The UMA Protocols were drafted by parties identified in the UMA Protocol technical specifications as “Participating Companies.” The Participating Companies were Alcatel, AT&T Wireless Services, Inc., BT PLC, Cingular Wireless LLC, Ericsson AB, Kineto Wireless Inc., Motorola, Inc., Nokia, Nortel Networks, Inc. O2, Rogers Wireless Inc., Siemens AG, Sony Ericsson, and T-Mobile USA. See [Stage 1] at 2. The UMA Protocols are prior art under at least 35 U.S.C. §§ 102(b) and 103(a). The UMA Protocols anticipate and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-25 provides representative, non-limiting examples of where the UMA Protocols disclose, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that the UMA Protocols do not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of the UMA Protocols alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or

modify the UMA Protocols with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

26. U.S. Patent Publication No. 2005/0286466 (“Tagg”)

U.S. Patent Publication No. 2005/0286466 (“Tagg”), titled “System for Providing Mobile VoIP,” was filed on May 12, 2005 and published on December 29, 2005. Tagg qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a), (b), and (g). Tagg anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-26 provides representative, non-limiting examples of where Tagg discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Tagg does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Tagg alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Tagg with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

27. EP Patent Publication No. 1811741A1 (“Oberle”)

EP Patent Publication No. 1811741A1 (“Oberle”), titled “Seamless session mobility for multimedia streams” was filed on January 20, 2006 and published on July 25, 2007. Oberle qualifies as prior art to the Asserted Patent under at least 35 U.S.C. § 102(a) and (g). Oberle anticipates and/or renders obvious (in combination with the knowledge of a person of ordinary skill in the art) all Asserted Claims. The claim chart attached as Exhibit A-27 provides

representative, non-limiting examples of where Oberle discloses, either expressly or inherently, each element, thereby anticipating those claims. To the extent Plaintiff asserts that Oberle does not anticipate any Asserted Claim, such Asserted Claim would be obvious in view of Oberle alone and/or in view of the knowledge of one of ordinary skill in the art, and it would have been obvious to combine or modify Oberle with concepts from other prior art, including any of the other primary references or the references identified in Appendix A or Section V.B (State of the Art and Obviousness), and/or in view of the knowledge of one of ordinary skill in the art, including as explained herein, to achieve the claimed inventions.

B. State of the Art and Obviousness

The primary references identified above, as further described in the Exhibits, each discloses, either expressly or inherently, every element of the Asserted Claims, thereby anticipating those claims. Additionally, each of the primary references alone (including in combination with the knowledge of a person of ordinary skill in the art) renders obvious the Asserted Claims. To the extent Plaintiff contends that any primary reference does not anticipate or render obvious the Asserted Claims as set forth above, it would have been obvious to combine or modify the primary references with concepts from other prior art, as explained throughout these Invalidity Contentions. In particular, for each limitation of the Asserted Claims that Plaintiff contends is not met by a particular primary reference, Defendants contend that the limitation (and claim as a whole) is obvious based on a combination of that particular primary reference with (1) any other primary reference disclosing that limitation, (2) any reference identified in Appendix A as disclosing that limitation, (3) any statements made in the intrinsic record of the Asserted Patent, and/or (4) the knowledge of one of ordinary skill in the art (as explained in additional detail below).

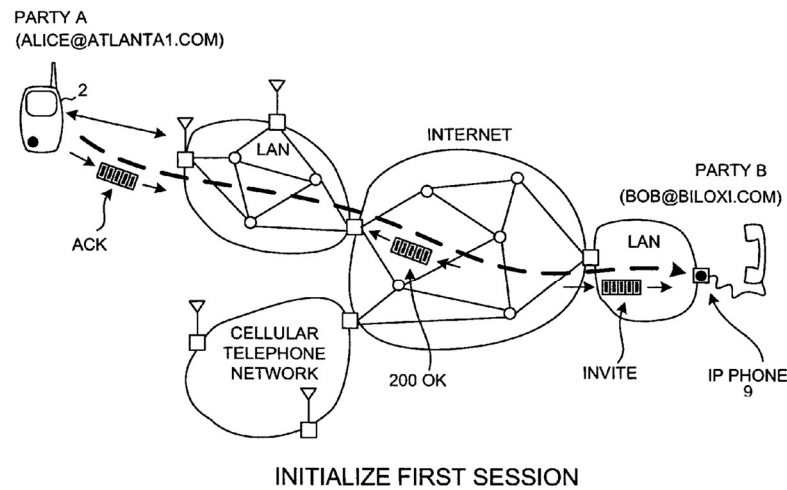
**Exhibit A-17: Invalidity of U.S. Patent No. 8,009,554 in view of
U.S. Patent No. 8,467,377 (“Kottilingal”)**

Based upon Plaintiff’s Complaint, Infringement Contentions, and apparent claim constructions and application of the claims to Defendants’ accused products, as best as they can be deciphered, the reference charted below anticipates or at least renders obvious the asserted claims. These invalidity contentions are not an admission by Defendants that the accused products are covered by or infringe the asserted claims, particularly when these claims are properly construed and applied. These invalidity contentions are not an admission that Defendants concede or acquiesce to any claim construction implied or suggested by Plaintiff’s Complaint or Infringement Contentions. Nor are Defendants asserting any claim construction positions through these charts, including whether the preamble is a limitation. The portions of the prior art reference cited below are not exhaustive but are exemplary in nature.

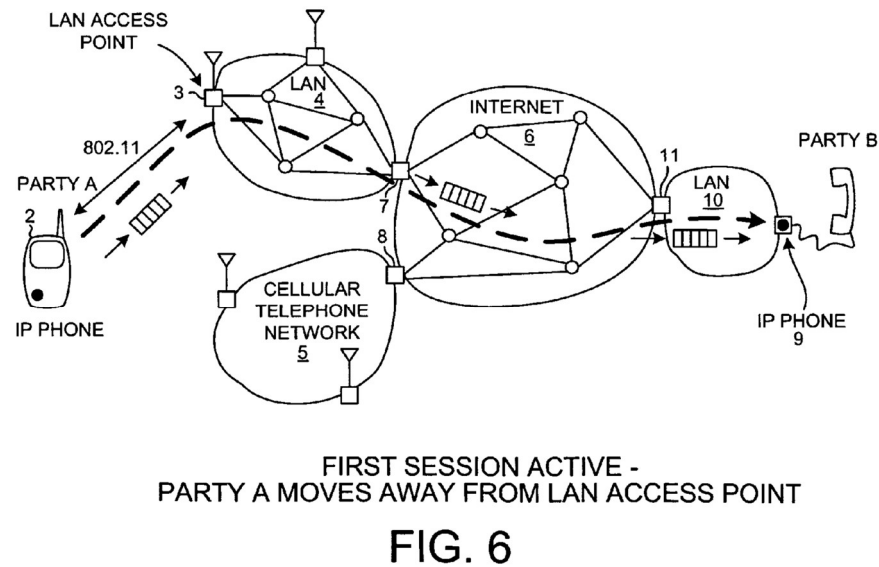
U.S. Patent No. 8,467,377 (“Kottilingal”) was filed on August 24, 2005, and issued on June 18, 2013. This patent is prior art under at least 35 U.S.C. §§ 102(e) and (g). As described in the following claim chart, the asserted claims of U.S. Patent No. 8,009,554 (the “’554 Patent”) are invalid as anticipated by Kottilingal.

To the extent that Kottilingal is found not to anticipate one or more of the asserted claims of the ’554 Patent, these claims are invalid as obvious in view of Kottilingal alone or in combination with other prior art references disclosed in Defendants’ Invalidity Contentions Cover Pleading and accompanying charts, including without limitation as set forth below.

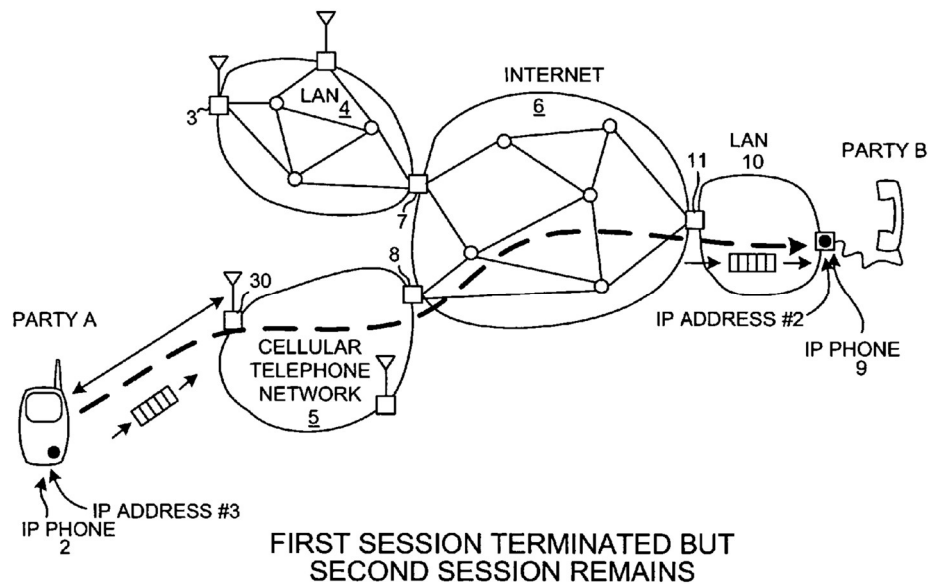
Claim No.	’554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
1[Pre]	A method of enhancing the reliability of a transmission from a source to a destination in a communications network using a network protocol that operates on top of the Internet Protocol (IP), the method comprising:	To the extent the preamble is limiting, Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i> “A first mobile device has two air interfaces. The first mobile device uses its first interface to transmit VoIP/VIP packets to a second device in a first session. If a quality of service decreases, then the first device causes a second session to be set up where the second session uses the second air interface. Both the first and second air interfaces are then used to communicate packets of a single media stream from the first device to the second device. The packets are transmitted in the first and second sessions in interleaved fashion, where the ratio of packets communicated in one session versus the other per unit time is adjusted in accordance with a quality of service afforded by one session versus a quality of service afforded by the other session. The second device orders the data payloads of the packets, thereby reconstructing the single media stream.” Kottilingal at Abstract.



Kottilingal at Figure 5.



Kottilingal at Figure 6.



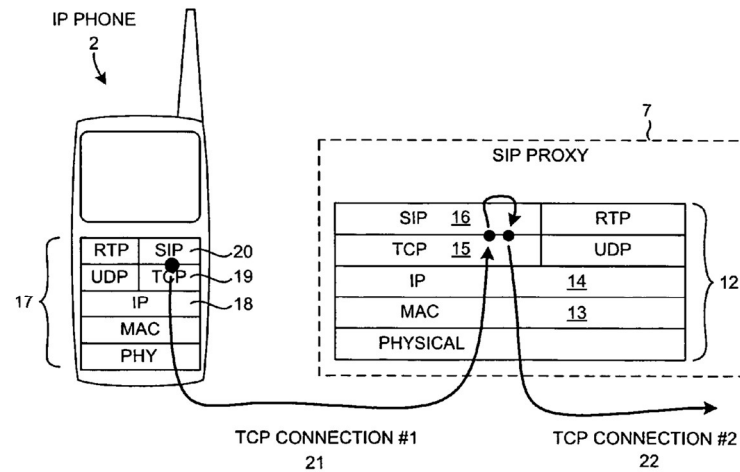
Kottilingal at Figure 17.

“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24.

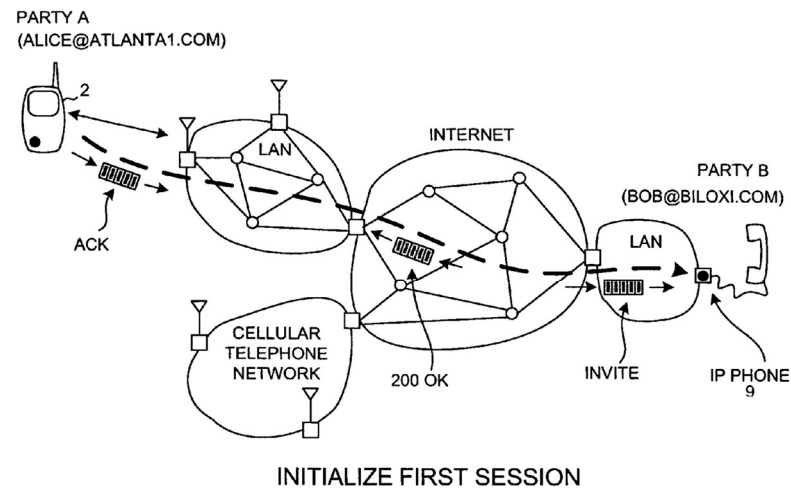
“FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
1[a]	transmitting data between a source and destination over a first network path made up of one or more network links where at least one of the network links is to a local area network that is remote from the transmission source; and	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i> “A first mobile device has two air interfaces. The first mobile device uses its first interface to transmit VoIP/VIP packets to a second device in a first session. If a quality of service decreases, then the first device causes a second session to be set up where the second session uses the second air interface. Both the first and second air interfaces are then used to communicate packets of a single media stream from the first device to the second device. The packets are transmitted in the first and second sessions in interleaved fashion, where the ratio of packets communicated in one session versus the other per unit time is adjusted in accordance with a quality of service afforded by one session versus a quality of service afforded by the other session. The second device orders the data payloads of the packets, thereby reconstructing the single media stream.” Kottilingal at Abstract.



Kottilingal at Figure 2.



Kottilingal at Figure 5.

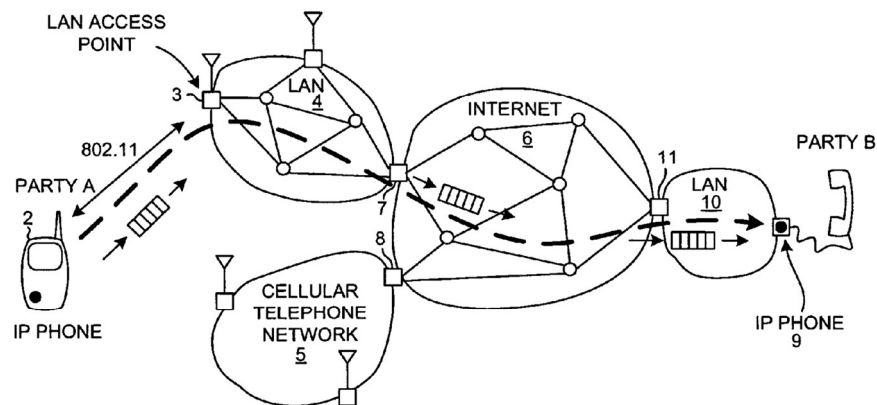


FIG. 6

Kottilingal at Figure 6.

```

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
MAX-FORWARDS: 70
TO: BOB <SIP:BOB@BILOXI.COM>
FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
CSEQ: 314160 SPAWN
CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
CONTENT-TYPE: APPLICATION/SDP
CONTENT-LENGTH: 142

```

```

MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
MASTER TO: BOB <SIP:BOB@BILOXI.COM>
MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
MASTER CSEQ: 314159 INVITE
MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

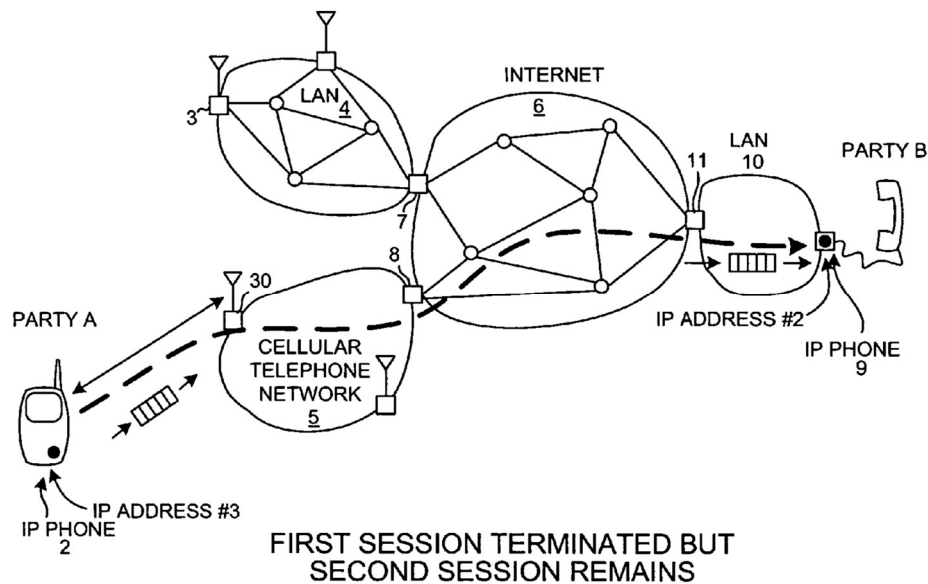
```

SO CALLEE CAN
IDENTIFY WITH
FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24.

“In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31. “FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“Although IP phone 9 in the example described above is a landline IP phone, IP phone 9 is a mobile wireless communication device (for example, a cellular telephone) in another example. The first and second sessions can be initiated by either mobile or landline IP phones.” Kottilingal at 11:33–37. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[b]	switching the transmission from the first network path to one of a plurality of alternative network paths;	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>

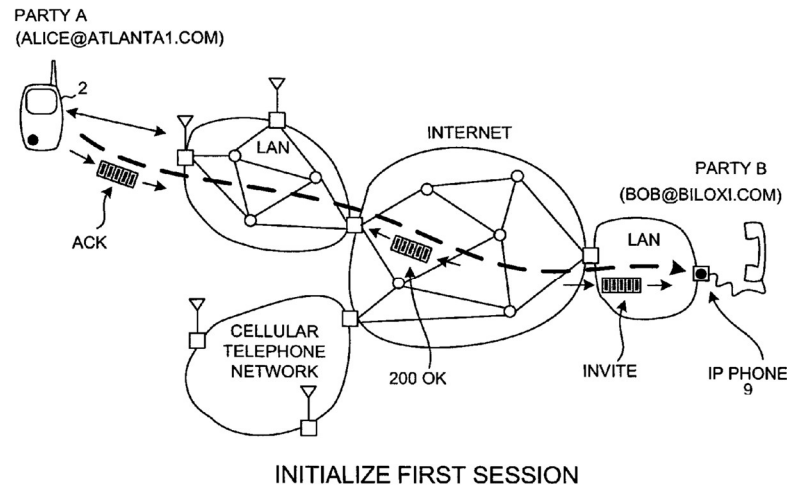


FIG. 5

Kottilingal at Figure 5.

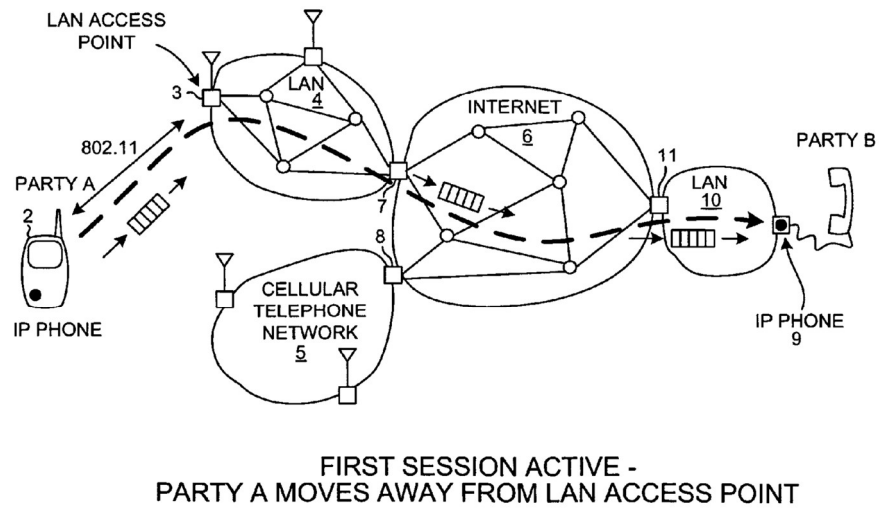


FIG. 6

Kottilingal at Figure 6.

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
 VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
 MAX-FORWARDS: 70
 TO: BOB <SIP:BOB@BILOXI.COM>
 FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
 CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
 CSEQ: 314160 SPAWN
 CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
 CONTENT-TYPE: APPLICATION/SDP
 CONTENT-LENGTH: 142

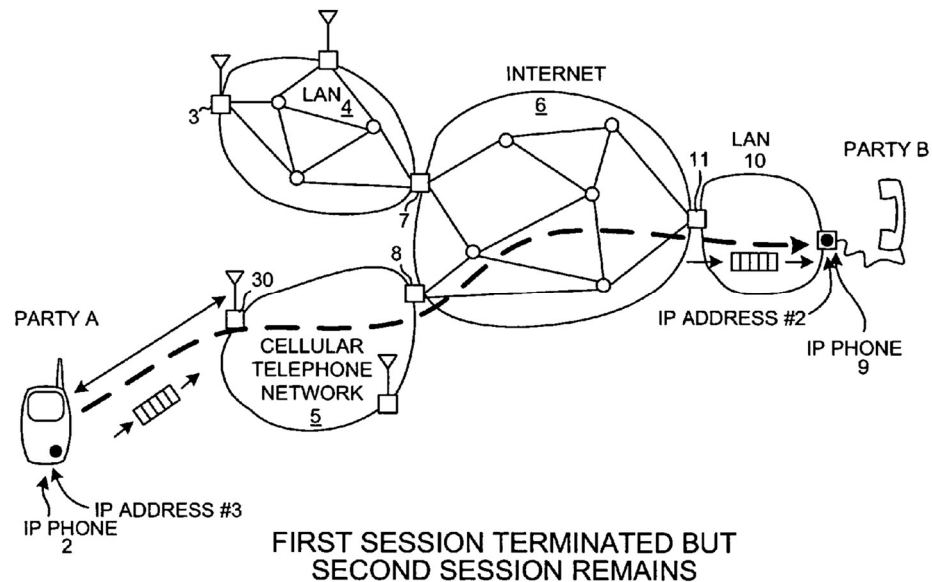
MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
 MASTER TO: BOB <SIP:BOB@BILOXI.COM>
 MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
 MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
 MASTER CSEQ: 314159 INVITE
 MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

SO CALLEE CAN
 IDENTIFY WITH
 FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24. “Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61. “In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47. “IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[c]	wherein said transmission is not disrupted as a result of said switch to the one of a plurality of alternative network paths, said transmission contains audio or video data and at least one Network Address Translation (NAT) is performed on said transmission;	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>

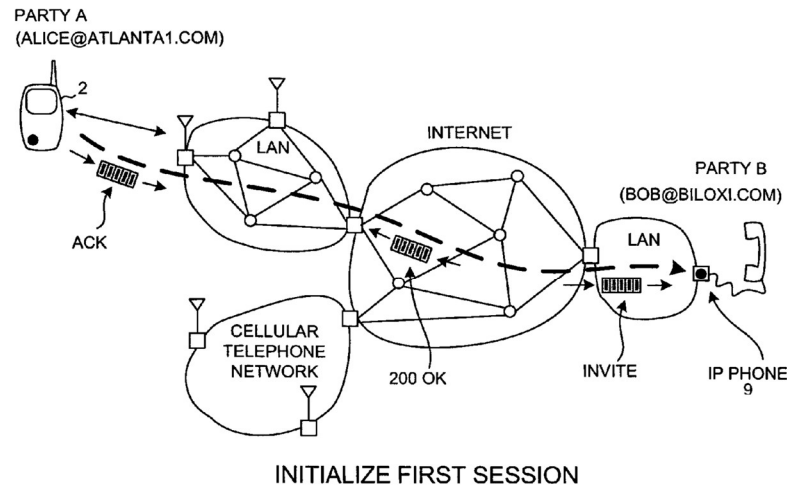


FIG. 5

Kottilingal at Figure 5.

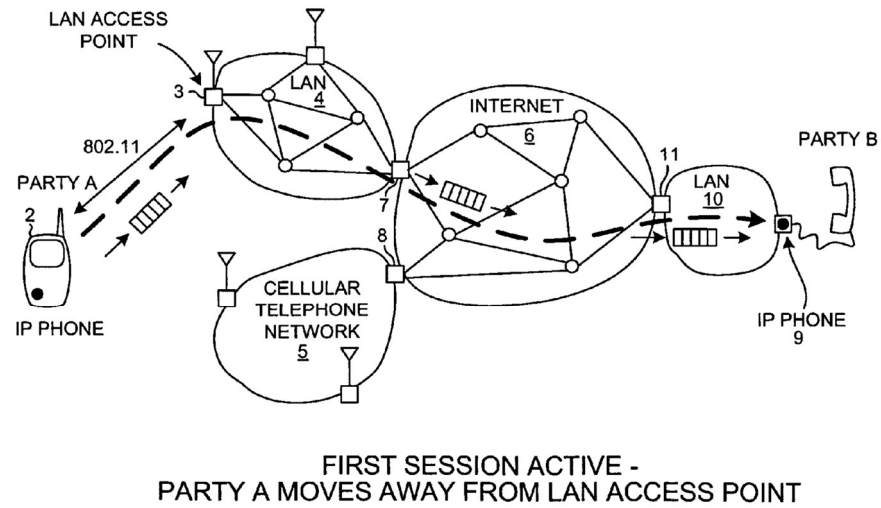


FIG. 6

Kottilingal at Figure 6.

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
 VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
 MAX-FORWARDS: 70
 TO: BOB <SIP:BOB@BILOXI.COM>
 FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
 CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
 CSEQ: 314160 SPAWN
 CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
 CONTENT-TYPE: APPLICATION/SDP
 CONTENT-LENGTH: 142

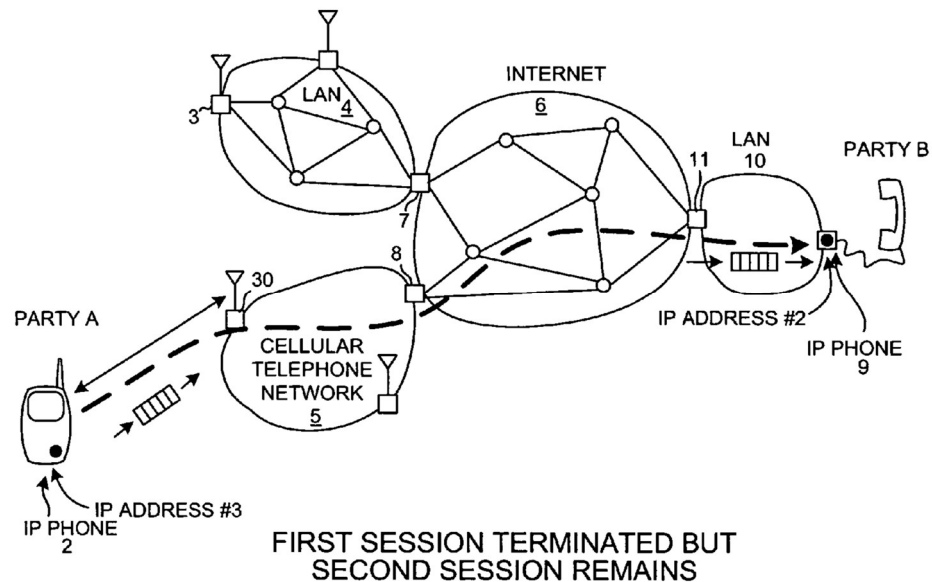
MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
 MASTER TO: BOB <SIP:BOB@BILOXI.COM>
 MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
 MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
 MASTER CSEQ: 314159 INVITE
 MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

SO CALLEE CAN
 IDENTIFY WITH
 FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24. “Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61. “Where, for example, the data payload VoIP packets contain voice data for a conversation, the communication device that receives the VoIP packets buffers VoIP payloads received in the second session in a FIFO (first in first out) buffer behind the payloads of the VoIP packets received in the first session. The VoIP packets are ordered in the FIFO according to the RTP sequence number and timestamps. The output of the FIFO buffer is converted into sound that is heard by the user of the communication device.” Kottilingal at 2:62–3:3. “In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31. “FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[d]	said transmission uses the Real Time Protocol (RTP); said transmission uses the Session Initiation Protocol (SIP); and	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>

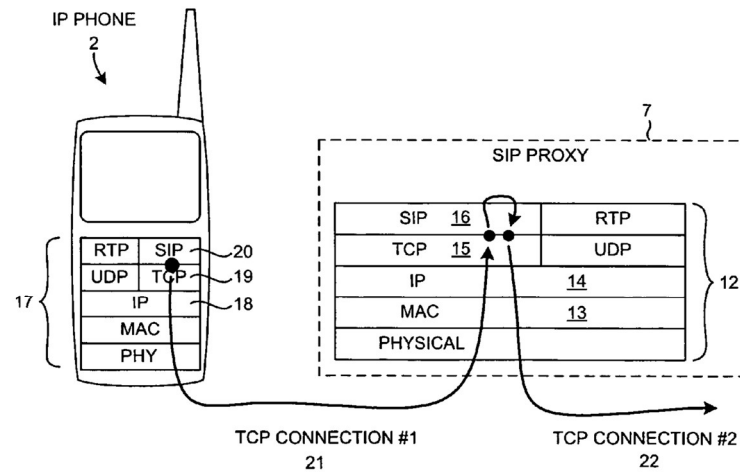


FIG. 2

Kottilingal at Figure 2.

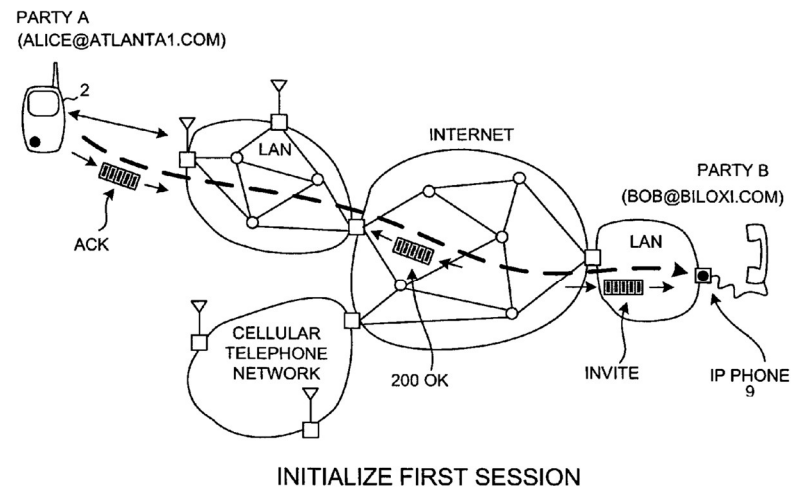
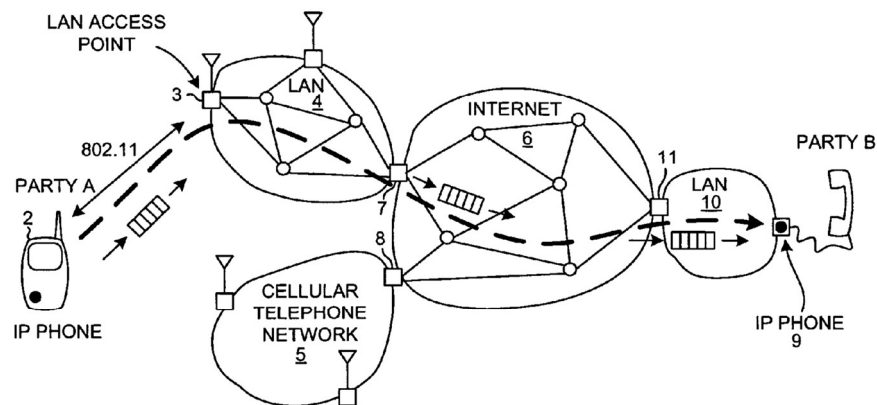


FIG. 5

Kottilingal at Figure 5.



FIRST SESSION ACTIVE -
PARTY A MOVES AWAY FROM LAN ACCESS POINT

FIG. 6

Kottilingal at Figure 6.

```

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
MAX-FORWARDS: 70
TO: BOB <SIP:BOB@BILOXI.COM>
FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
CSEQ: 314160 SPAWN
CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
CONTENT-TYPE: APPLICATION/SDP
CONTENT-LENGTH: 142

```

```

MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
MASTER TO: BOB <SIP:BOB@BILOXI.COM>
MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
MASTER CSEQ: 314159 INVITE
MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

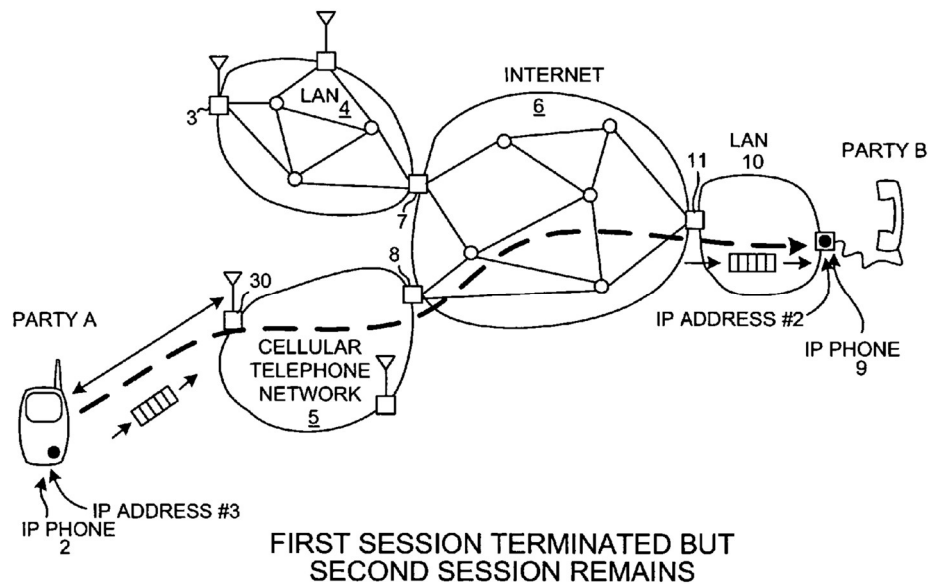
```

SO CALLEE CAN
IDENTIFY WITH
FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

“Regardless of the reason for desiring to switch from the initially used air interface to the other air interface, PARTY A's mobile communication device sends a SPAWN SIP message to PARTY B's target communication device. The SPAWN SIP message is communicated using SIP (Session Initialization Protocol) over TCP (Transmission Control Protocol) over IP. The target responds by sending a 200 OK SIP message that contains a spawn identifier. The mobile communication device then sets up a second session across the other air interface by sending a SIP INVITE request across the other interface to the target. The SIP INVITE request contains the spawn identifier. The second session is initialized and both the first and second sessions are active VoIP sessions. Neither session involves a circuit-switched link. The target communication device uses the spawn identifier received in the SIP INVITE request to associate the first and second sessions.” Kottilingal at 2:25–41.

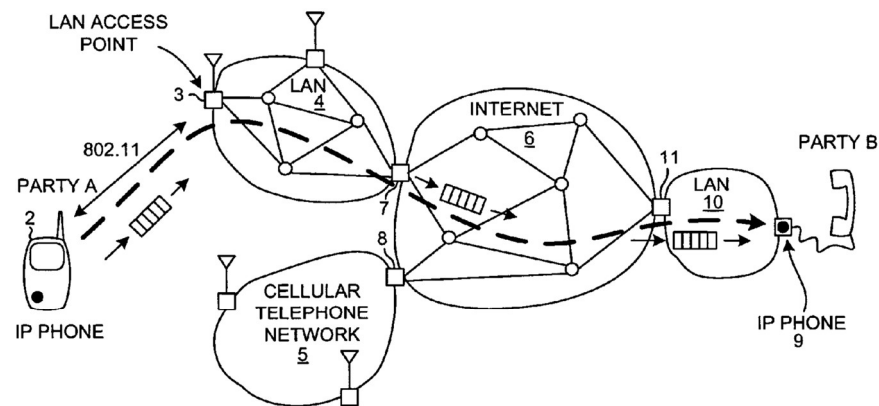
“FIG. 2 illustrates IP phone 2 and SIP proxy 7. A stack 12 of protocol processing layers executes on the hardware platform of SIP proxy 7. There is one such stack of protocol processing layers executing on each of the SIP proxies 7, 8 and 11. Stack 12 includes, among other layers, a MAC layer 13, an IP layer 14, a TCP layer 15, and a SIP layer 16. MAC stands for “Media Access Control”. IP stands for “Internet Protocol”. TCP stands for “Transmission Control Protocol”. SIP stands for “Session Initialization Protocol”. The processor within IP phone 2 executes a similar stack 17 of protocol processing layers. Stack 17 includes an IP layer 18, a TCP layer 19 and a SIP

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		layer 20. Because each of the stacks 12 and 17 includes an IP layer and a TCP layer, a TCP connection can be established between the IP phone 2 and SIP proxy 7. In FIG. 2, the black dot in IP phone 2 represents IP address #1 of IP phone 2. The black dots in SIP proxy 7 represent IP addresses. The leftmost arrow represents a first TCP connection 21 that terminates in IP phone 2 and in SIP proxy 7. Whereas IP protocol communication is only a best-efforts communication, the use of TCP in addition to IP allows the reliable communication of information between IP phone 2 and SIP proxy 7 across the TCP connection 21. In the same way that TCP connection 21 is established and maintained between IP phone 2 and SIP proxy 7, a second TCP connection 22 is established and maintained between SIP proxy 7 and another device on the Internet that has an IP address and that has a stack involving IP and TCP protocol processing layers. SIP proxy 11 (see FIG. 1) is one such device. In FIG. 2, the rightmost arrow represents a second TCP connection 22 that is terminated on one end by SIP proxy 7 and on another end by SIP proxy 11.” Kottilingal at 6:7–36. “FIG. 5 illustrates this SIP transaction involving the SIP INVITE request, the 200 OK message, and the ACK. Although all three SIP messages of the transaction are illustrated as propagating through the networks at the same time, ” Kottilingal at 7:64–67. “the SIP messages are actually sent and received one at a time as described above. The result of the SIP transaction is the initialization of a first SIP session between IP address #1 (of PARTY A) and IP address #2 (of PARTY B). An initialized SIP session that has not been terminated is said to be “active”, regardless of whether data payloads are being communicated in the session or not. In the present example, once the first SIP session has been initialized, first VoIP/VIP (voice over IP or video over IP) IP packets having voice and/or video data payloads are communicated between IP address #1 of IP phone 2 and IP address #2 of IP phone 9 in UDP (User Datagram Protocol) packets in accordance with the RTP (Real-Time Protocol) protocol. Data payloads for the first session are communicated using RTP over UDP over IP, whereas control packets for the first session are communicated using SIP over TCP over IP.” Kottilingal at 8:1–16. “FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 7 is a diagram that illustrates the SPAWN SIP message.” Kottilingal at 8:45–46. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47. “IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone 2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[e]	the SIP signaling sequence of the transmission is maintained after said switching from the first network path to the one of a plurality of alternative network paths.	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>



FIRST SESSION ACTIVE -
PARTY A MOVES AWAY FROM LAN ACCESS POINT

FIG. 6

Kottilingal at Figure 6.

```

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
MAX-FORWARDS: 70
TO: BOB <SIP:BOB@BILOXI.COM>
FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
CSEQ: 314160 SPAWN
CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
CONTENT-TYPE: APPLICATION/SDP
CONTENT-LENGTH: 142

```

```

MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
MASTER TO: BOB <SIP:BOB@BILOXI.COM>
MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
MASTER CSEQ: 314159 INVITE
MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

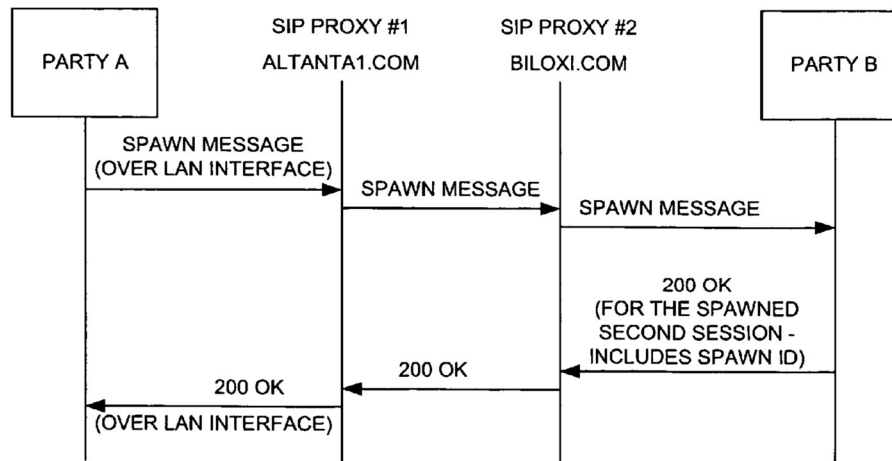
```

SO CALLEE CAN
IDENTIFY WITH
FIRST SESSION

SPAWN MESSAGE

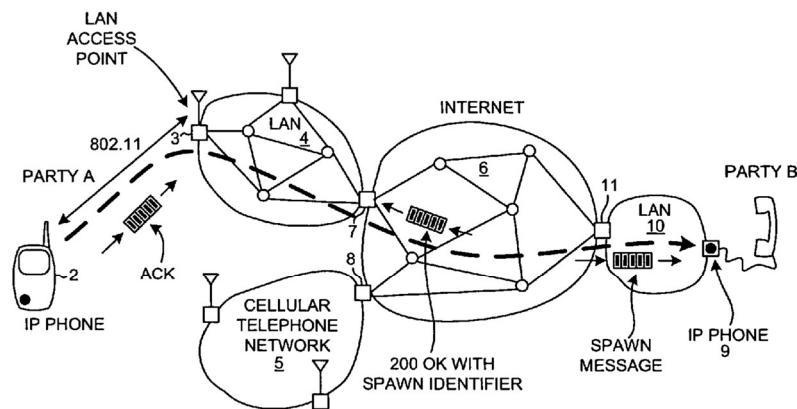
FIG. 7

Kottilingal at Figure 7.



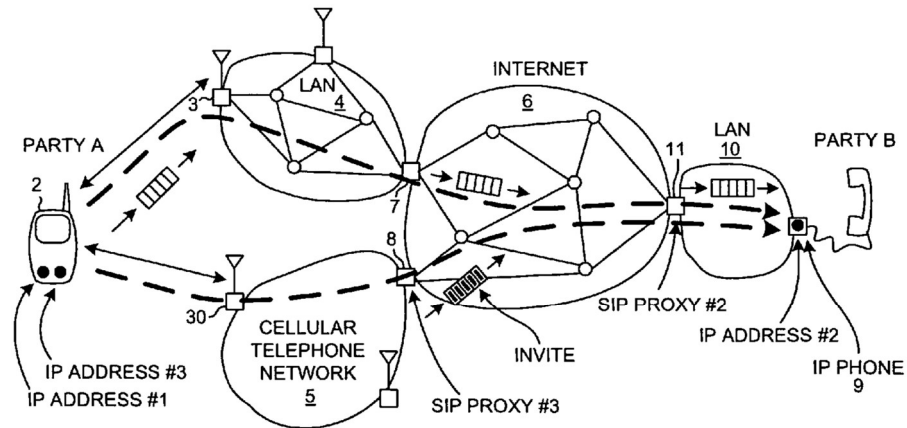
SPAWN MESSAGE
FIG. 8

Kottilingal at Figure 8.



SPAWN MESSAGE
FIG. 9

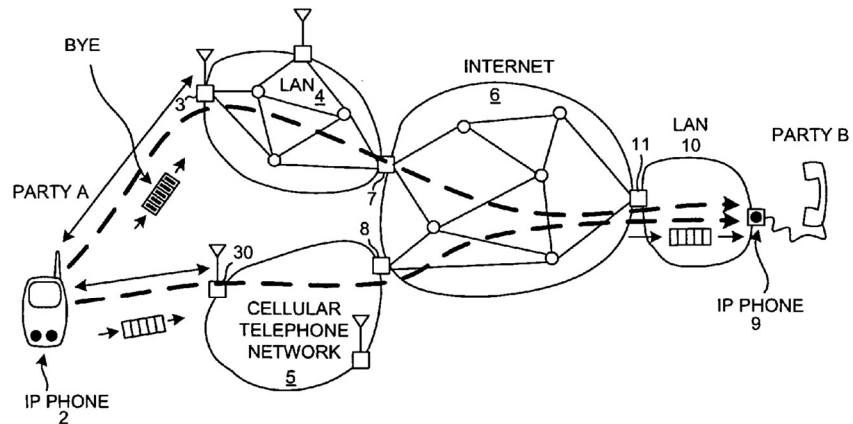
Kottilingal at Figure 9.



INVITE FOR SECOND SESSION

FIG. 10

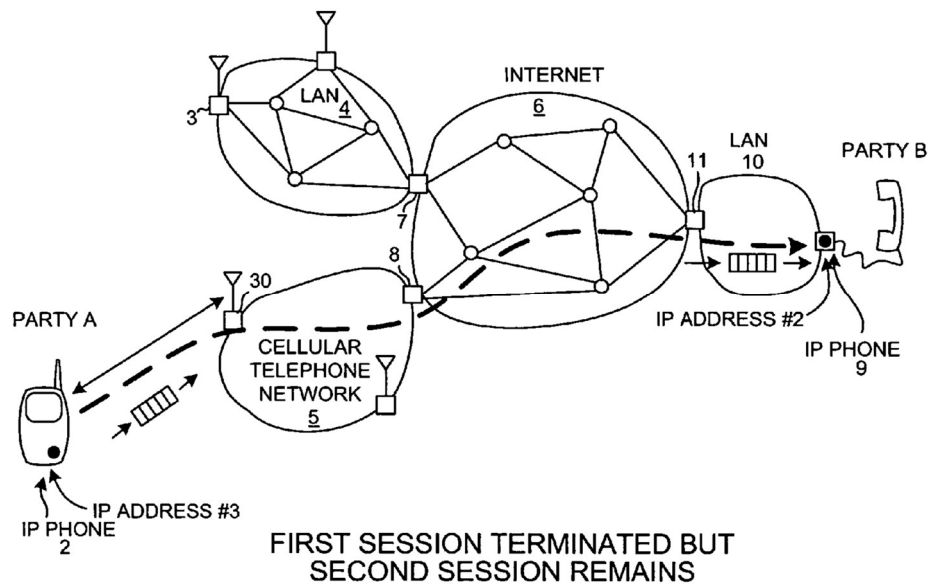
Kottilingal at Figure 10.



BYE PACKET SENT ON FIRST SESSION

FIG. 16

Kottilingal at Figure 16.



Kottilingal at Figure 17.

“Regardless of the reason for desiring to switch from the initially used air interface to the other air interface, PARTY A's mobile communication device sends a SPAWN SIP message to PARTY B's target communication device. The SPAWN SIP message is communicated using SIP (Session Initialization Protocol) over TCP (Transmission Control Protocol) over IP. The target responds by sending a 200 OK SIP message that contains a spawn identifier. The mobile communication device then sets up a second session across the other air interface by sending a SIP INVITE request across the other interface to the target. The SIP INVITE request contains the spawn identifier. The second session is initialized and both the first and second sessions are active VoIP sessions. Neither session involves a circuit-switched link. The target communication device uses the spawn identifier received in the SIP INVITE request to associate the first and second sessions.” Kottilingal at 2:25–41.

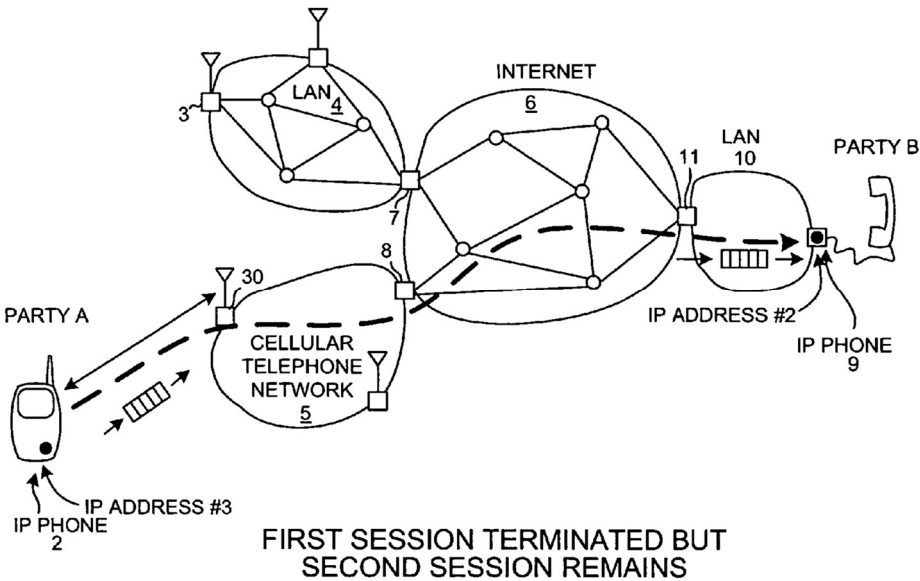
“FIG. 5 illustrates this SIP transaction involving the SIP INVITE request, the 200 OK message, and the ACK. Although all three SIP messages of the transaction are illustrated as propagating through the networks at the same time,” Kottilingal at 7:64–67.

“the SIP messages are actually sent and received one at a time as described above. The result of the SIP transaction is the initialization of a first SIP session between IP address #1 (of PARTY A) and IP address #2 (of PARTY B). An initialized SIP session that has not been terminated is said to be

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“active”, regardless of whether data payloads are being communicated in the session or not. In the present example, once the first SIP session has been initialized, first VoIP/VIP (voice over IP or video over IP) IP packets having voice and/or video data payloads are communicated between IP address #1 of IP phone 2 and IP address #2 of IP phone 9 in UDP (User Datagram Protocol) packets in accordance with the RTP (Real-Time Protocol) protocol. Data payloads for the first session are communicated using RTP over UDP over IP, whereas control packets for the first session are communicated using SIP over TCP over IP.” Kottilingal at 8:1–16. “FIG. 7 is a diagram that illustrates the SPAWN SIP message.” Kottilingal at 8:45–46. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
		of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service." Kottilingal at 12:1-47. "IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone 2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased." Kottilingal at 12:48-67.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“The second session can be set up using any suitable method. In the example presently described, the second session is set up using the method described above in connection with FIGS. 7-11. IP phone 2 communicates a SIP SPAWN message that identifies the first session to IP phone 9. IP phone 9 responds by communicating a SIP OK message back to IP phone 2. The SIP OK message includes a SPAWN identifier. IP phone 2 then communicates a SIP INVITE message to IP phone 9, where the SIP INVITE message includes the spawn identifier. IP phone 9 receives the SIP INVITE message, sets up a second session, and uses the SIP INVITE message to associate the second session with the first session.” Kottilingal at 13:1–12. “Voice data is therefore communicated in interleaved fashion across the two sessions. In accordance with the RTP protocol, each VoIP/VIP packet has an RTP timestamp as well as an RTP sequence number. IP phone 9 receives the VoIP/VIP packets coming into the IP phone 9 across the first and second sessions, and uses the RTP timestamps and/or RTP sequence numbers to order the data payloads so as to reconstruct the media stream. In the example described above wherein the four VoIP/VIP packets carry voice data of a single media stream, the data payloads are ordered by IP phone 9 in the following order: first data payload, second data payload, third data payload, fourth data payload. IP phone 9 uses the RTCP sequence numbers and RTCP time stamps to properly order the data payloads. The properly ordered data payloads constitute the reconstructed media stream.” Kottilingal at 13:31–45. “receiving at the first device a first SIP OK message from the second device via the first wireless communication link, the first SIP OK message including a SPAWN identifier that comprises a hash of a FROM field, a TO field, a CALL-ID field, and a CSEQ number of the first application layer session;” Kottilingal at 16:43–48. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
2	The method of claim 1 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of a stream quality event.	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>  Kottilingal at Figure 17. “Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47. “IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
3	The method of claim 1 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	Kottilingal expressly or inherently discloses this claim element. See below re claim 17 which is incorporated by reference herein.
4	The method of claim 1 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of an administrator move.	Kottilingal expressly or inherently discloses this claim element. See below re claim 18 which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
5	The method of claim 1 wherein a SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Kottilingal expressly or inherently discloses this claim element. See below re claim 19 which is incorporated by reference herein.
6	The method of claim 1 wherein the source device is a mobile device.	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i> "A first mobile device has two air interfaces. The first mobile device uses its first interface to transmit VoIP/VIP packets to a second device in a first session. If a quality of service decreases, then the first device causes a second session to be set up where the second session uses the second air interface. Both the first and second air interfaces are then used to communicate packets of a single media stream from the first device to the second device. The packets are transmitted in the first and second sessions in interleaved fashion, where the ratio of packets communicated in one session versus the other per unit time is adjusted in accordance with a quality of service afforded by one session versus a quality of service afforded by the other session. The second device orders the data payloads of the packets, thereby reconstructing the single media stream." Kottilingal at Abstract.

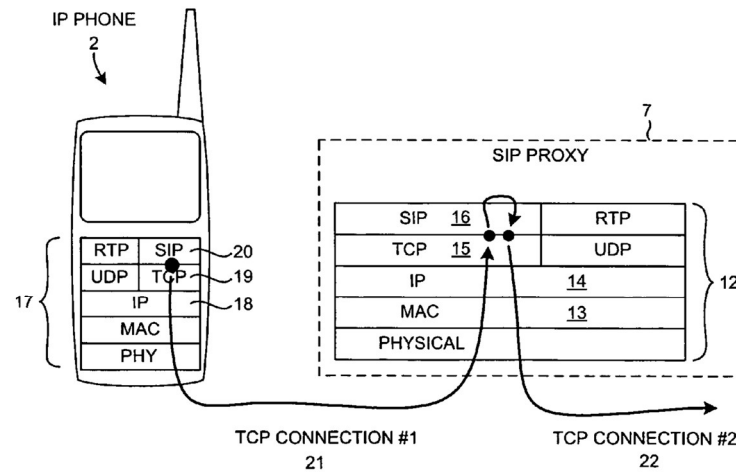


FIG. 2

Kottilingal at Figure 2.

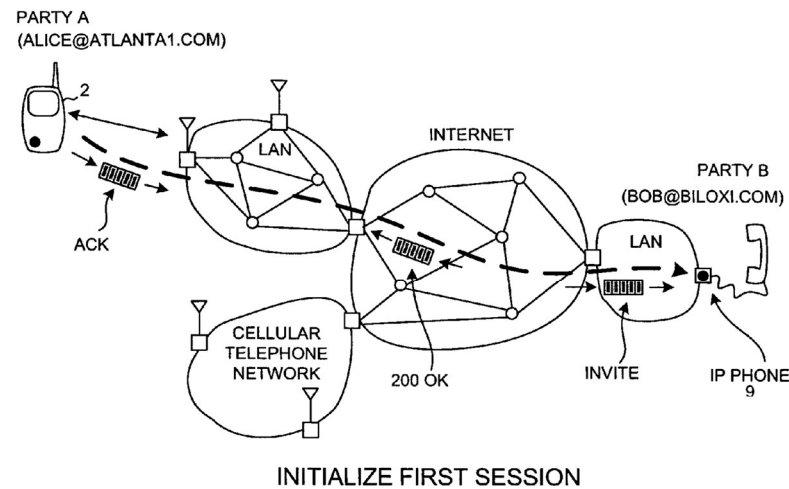
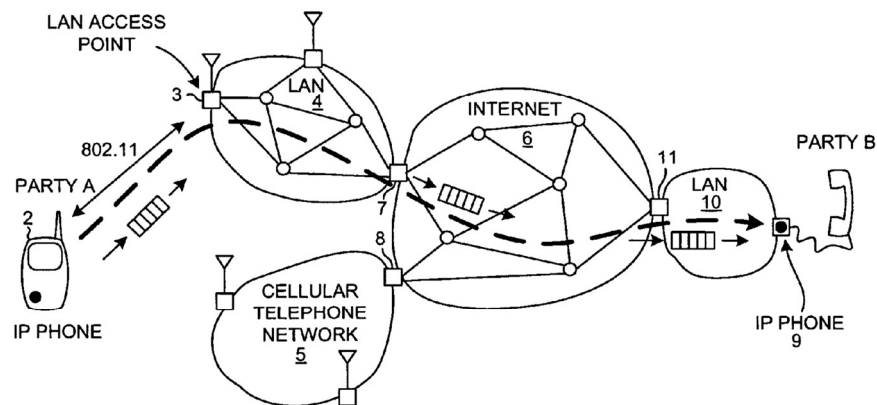


FIG. 5

Kottilingal at Figure 5.



FIRST SESSION ACTIVE -
PARTY A MOVES AWAY FROM LAN ACCESS POINT

FIG. 6

Kottilingal at Figure 6.

```

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
MAX-FORWARDS: 70
TO: BOB <SIP:BOB@BILOXI.COM>
FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
CSEQ: 314160 SPAWN
CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
CONTENT-TYPE: APPLICATION/SDP
CONTENT-LENGTH: 142

```

```

MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
MASTER TO: BOB <SIP:BOB@BILOXI.COM>
MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
MASTER CSEQ: 314159 INVITE
MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

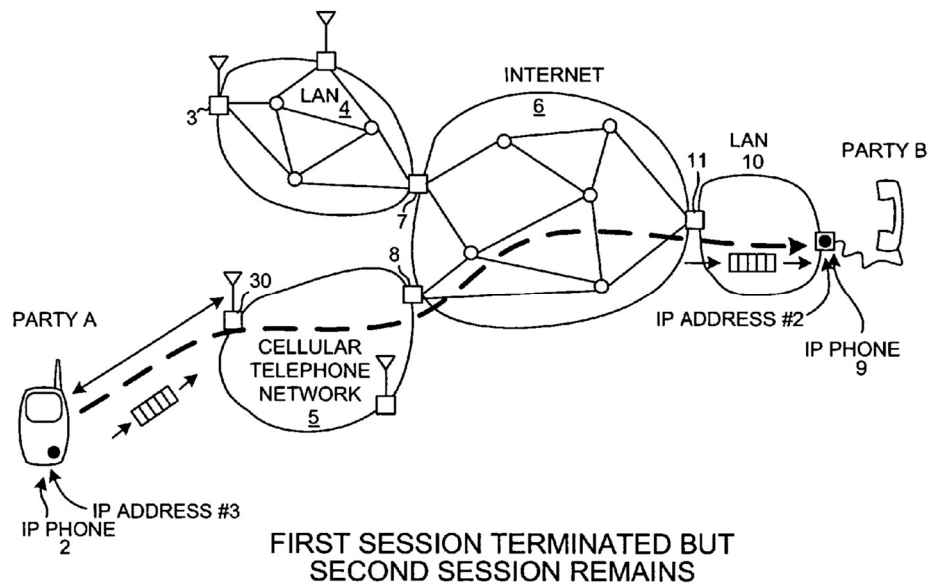
```

SO CALLEE CAN
IDENTIFY WITH
FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



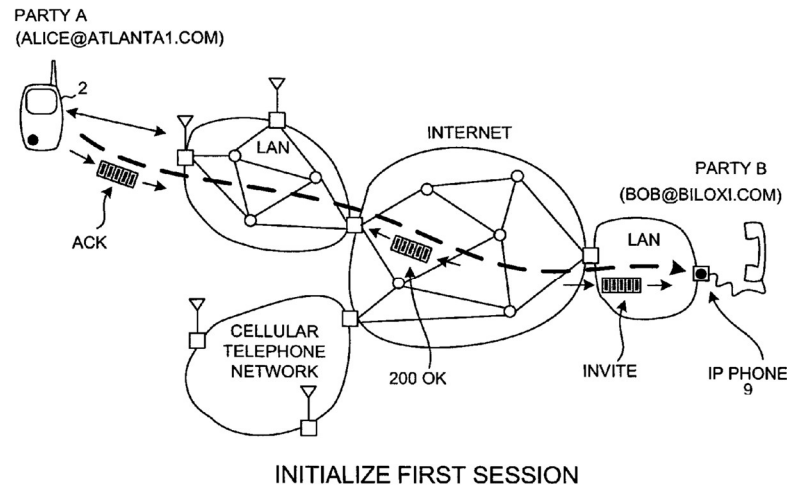
Kottilingal at Figure 17.

“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24.

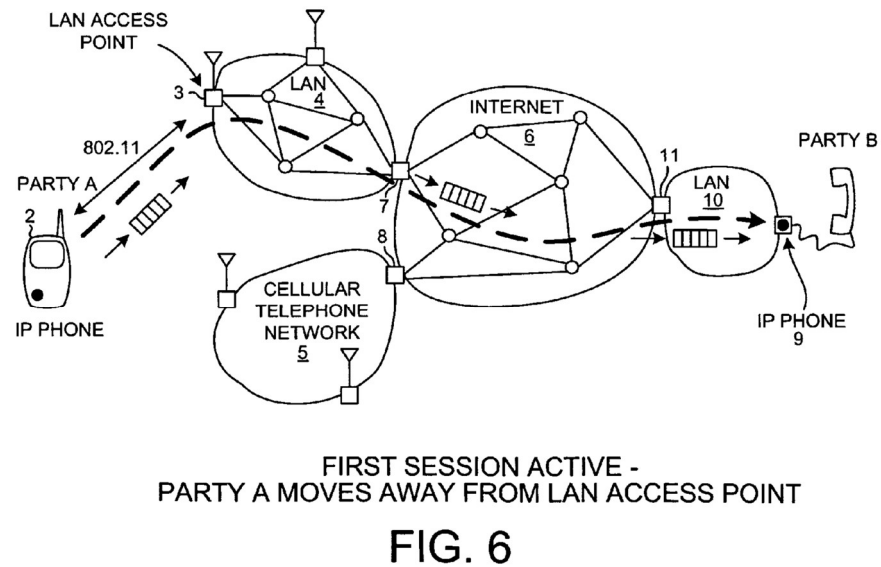
“In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31. “FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“Although IP phone 9 in the example described above is a landline IP phone, IP phone 9 is a mobile wireless communication device (for example, a cellular telephone) in another example. The first and second sessions can be initiated by either mobile or landline IP phones.” Kottilingal at 11:33–37. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
7	The method of claim 6 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>



Kottilingal at Figure 5.



Kottilingal at Figure 6.

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
 VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
 MAX-FORWARDS: 70
 TO: BOB <SIP:BOB@BILOXI.COM>
 FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
 CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
 CSEQ: 314160 SPAWN
 CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
 CONTENT-TYPE: APPLICATION/SDP
 CONTENT-LENGTH: 142

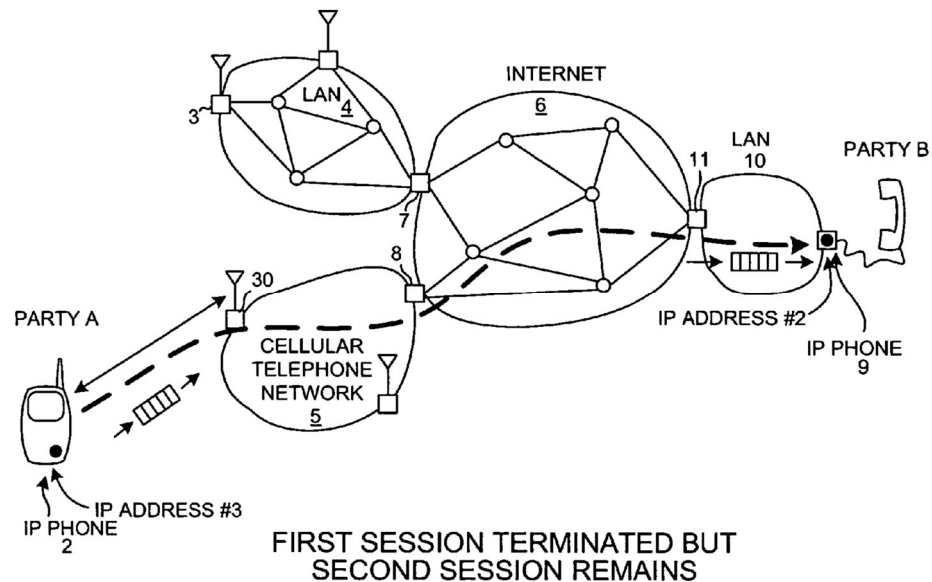
MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
 MASTER TO: BOB <SIP:BOB@BILOXI.COM>
 MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
 MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
 MASTER CSEQ: 314159 INVITE
 MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

SO CALLEE CAN
 IDENTIFY WITH
 FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24. “Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61. “In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47. “IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone

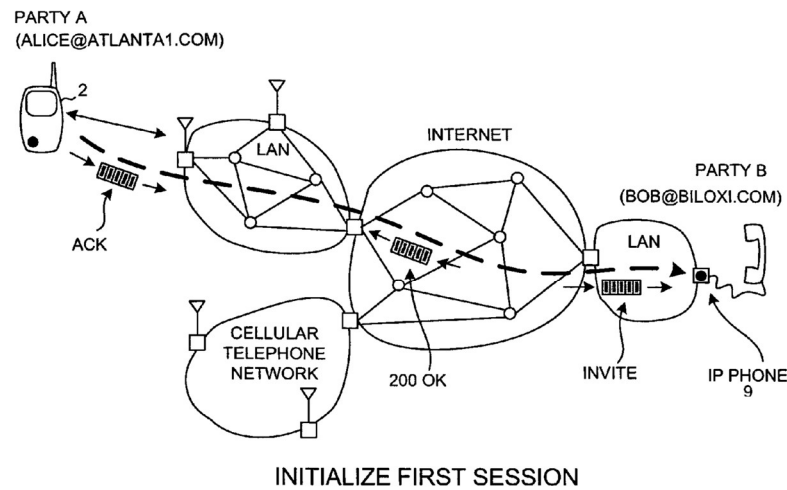
Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
8[Pre]	A communications network comprising:	To the extent the preamble is limiting, Kottilingal expressly or inherently discloses this claim element. See above re claim 1[Pre] which is incorporated by reference herein.
8[a]	a transmitting device; a receiving device; a first network device; a second network device;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[a] which is incorporated by reference herein.
8[b]	the transmitting device and receiving device operable to maintain a data transmission over a first network path that is made up of one or more network links where at least one of the network	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[a] which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
	links is to a local area network is remote from the transmitting device;	
8[c]	wherein the first network device is operable to switch the transmission from the first network path to one of a plurality of alternative network paths without disrupting the transmission; the second network device is operable to perform a Network Address Translation (NAT) on said transmission;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[b] and 1[c] which are incorporated by reference herein.
8[d]	said transmission contains audio or video content;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[c] which is incorporated by reference herein.
8[e]	said transmission is made using the Real Time Protocol (RTP); said transmission uses the Session Initiation Protocol (SIP); and	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[d] which is incorporated by reference herein.
8[f]	said transmission's SIP signaling sequence is maintained after said switching from the first network path to the one of a plurality of alternative network paths.	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[e] which is incorporated by reference herein.
9	The system of claim 8 wherein said switching from the first network path to the one of a plurality of alternative network	Kottilingal expressly or inherently discloses this claim element. See above re claim 2 which is incorporated by reference herein.

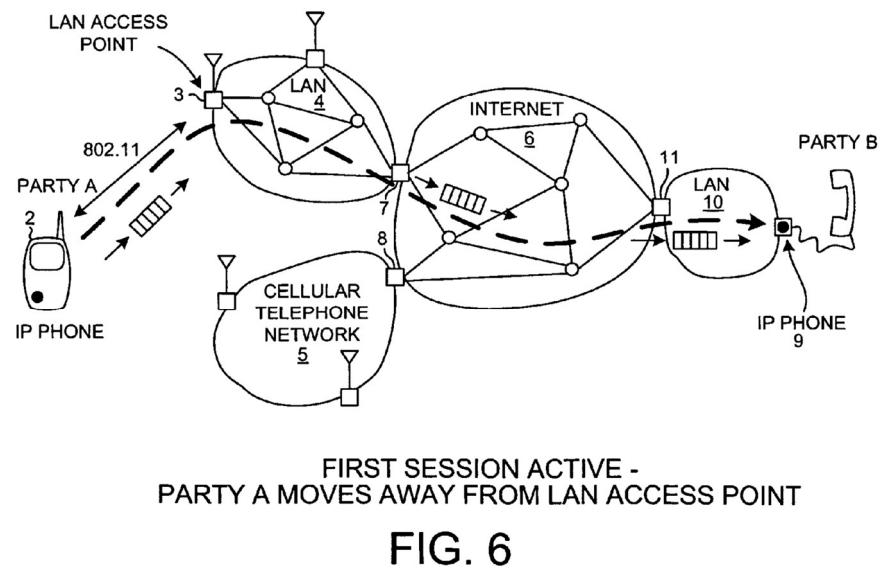
Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
	paths occurs as a result of a stream quality event.	
10	The system of claim 8 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	Kottilingal expressly or inherently discloses this claim element. See below re claim 17 which is incorporated by reference herein.
11	The system of claim 8 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of an administrator move.	Kottilingal expressly or inherently discloses this claim element. See below re claim 18 which is incorporated by reference herein.
12	The system of claim 8 wherein the SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Kottilingal expressly or inherently discloses this claim element. See below re claim 19 which is incorporated by reference herein.
13	The system of claim 8 wherein the transmitting device is a mobile device.	Kottilingal expressly or inherently discloses this claim element. See above re claim 6 which is incorporated by reference herein.
14	The system of claim 13 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Kottilingal expressly or inherently discloses this claim element. See above re claim 7 which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
15[pre]	A method of enhancing the reliability of a transmission from a source to a destination in a communications network using a network protocol that operates on top of the Internet Protocol (IP), the method comprising:	To the extent the preamble is limiting, Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i> See above re claim 1[pre] which is incorporated by reference herein.
15[a]	transmitting data between a source and destination over a first network path made up of one or more network links where at least one of the network links is to a local area network that is remote from the transmission source;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[a] which is incorporated by reference herein.
15[b]	switching the transmission from the first network path to one of a plurality of alternative network paths;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[b] which is incorporated by reference herein.
15[c]	wherein said transmission is not disrupted as a result of said switch to the one of a plurality of alternative network paths, said transmission contains audio or video data, and at least one Port Address Translation (PAT) is performed on said transmission;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[c] which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
15[d]	said transmission contains audio or video content;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[c] which is incorporated by reference herein.
15[e]	said transmission contains data used in the control of the audio or video transmission;	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>



Kottilingal at Figure 5.



Kottilingal at Figure 6.

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
 VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
 MAX-FORWARDS: 70
 TO: BOB <SIP:BOB@BILOXI.COM>
 FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
 CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
 CSEQ: 314160 SPAWN
 CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
 CONTENT-TYPE: APPLICATION/SDP
 CONTENT-LENGTH: 142

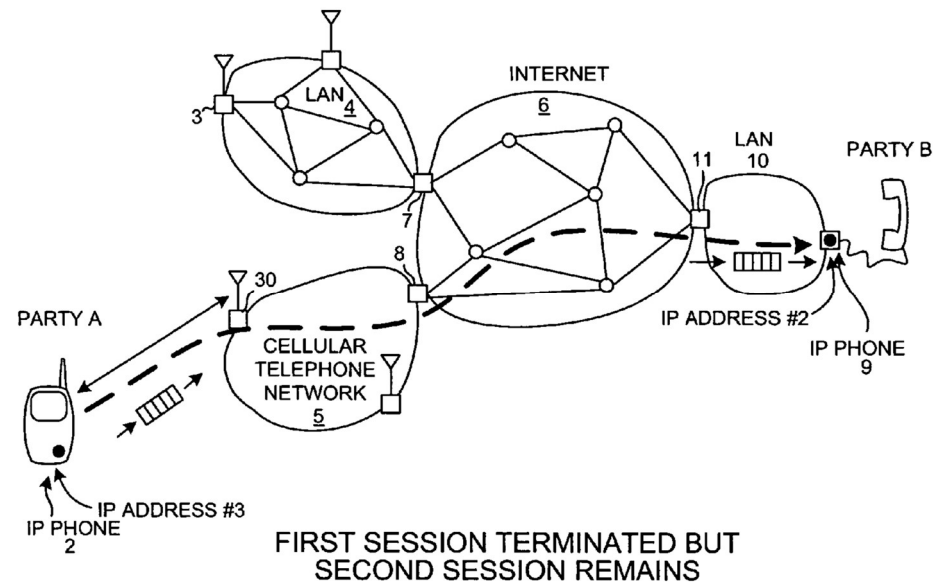
MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
 MASTER TO: BOB <SIP:BOB@BILOXI.COM>
 MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
 MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
 MASTER CSEQ: 314159 INVITE
 MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

SO CALLEE CAN
 IDENTIFY WITH
 FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

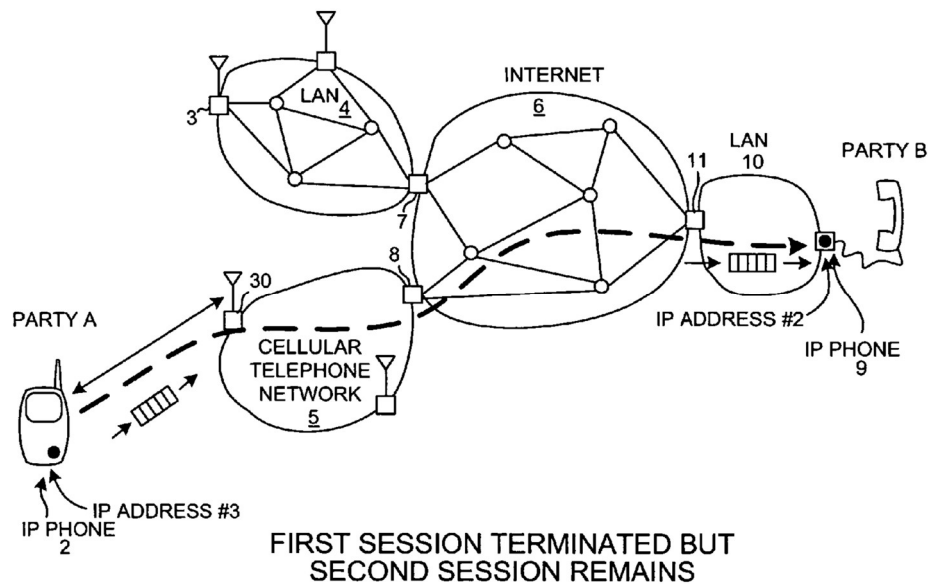
Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24. “Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61. “Where, for example, the data payload VoIP packets contain voice data for a conversation, the communication device that receives the VoIP packets buffers VoIP payloads received in the second session in a FIFO (first in first out) buffer behind the payloads of the VoIP packets received in the first session. The VoIP packets are ordered in the FIFO according to the RTP sequence number and timestamps. The output of the FIFO buffer is converted into sound that is heard by the user of the communication device.” Kottilingal at 2:62–3:3. “In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31. “FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone 2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
15[f]	said transmission uses the Real Time Protocol (RTP); said transmission further uses the Session Initiation Protocol (SIP); and	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[d] which is incorporated by reference herein.
15[g]	the SIP signaling sequence of the transmission is maintained after	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[e] which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
	said switching from the first network path to the one of a plurality of alternative network paths.	
16	The method of claim 15 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of a stream quality event.	Kottilingal expressly or inherently discloses this claim element. See above re claim 2 which is incorporated by reference herein.
17	The method of claim 15 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>



Kottilingal at Figure 17.

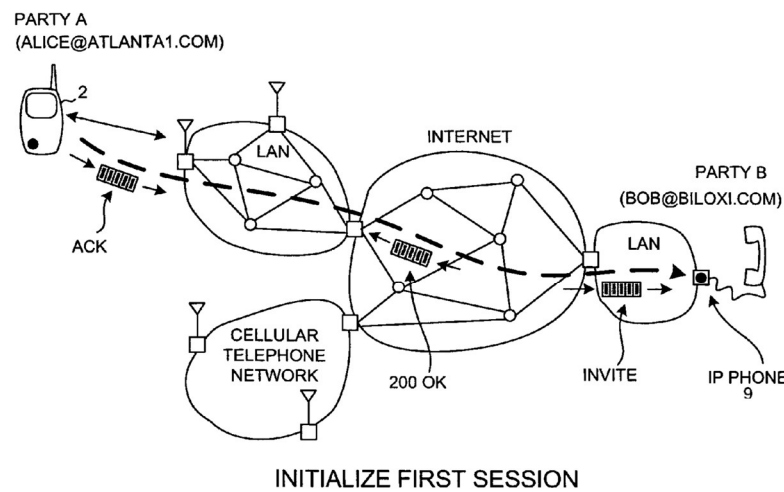
“Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61.

“FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back

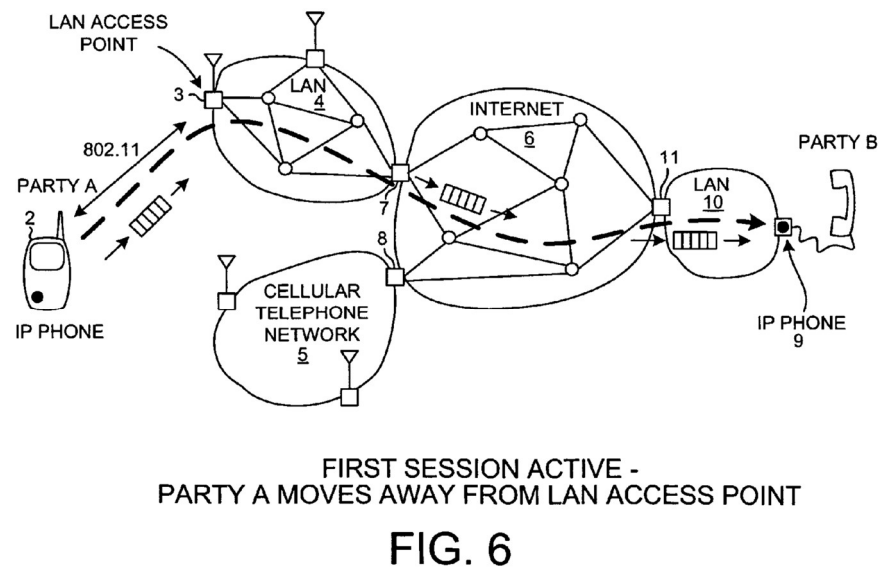
Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47. “IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone 2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
18	The method of claim 15 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i>

as a result of an administrator move.



Kottilingal at Figure 5.



Kottilingal at Figure 6.

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
 VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
 MAX-FORWARDS: 70
 TO: BOB <SIP:BOB@BILOXI.COM>
 FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
 CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
 CSEQ: 314160 SPAWN
 CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
 CONTENT-TYPE: APPLICATION/SDP
 CONTENT-LENGTH: 142

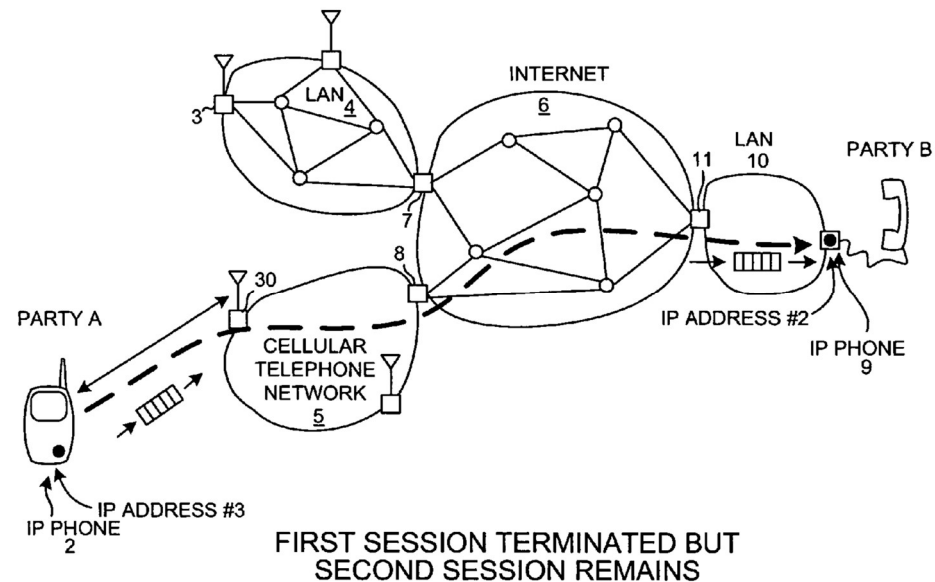
MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
 MASTER TO: BOB <SIP:BOB@BILOXI.COM>
 MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
 MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
 MASTER CSEQ: 314159 INVITE
 MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

SO CALLEE CAN
 IDENTIFY WITH
 FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24. “Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61. “In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		“FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both. IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47. “IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
19	The method of claim 15 wherein a SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Kottilingal expressly or inherently discloses this claim element. <i>See, e.g.,</i> “A first mobile device has two air interfaces. The first mobile device uses its first interface to transmit VoIP/VIP packets to a second device in a first session. If a quality of service decreases, then the first device causes a second session to be set up where the second session uses the second air interface. Both the first and second air interfaces are then used to communicate packets of a single media stream from the first device to the second device. The packets are transmitted in the first and second sessions in interleaved fashion, where the ratio of packets communicated in one session versus the other per unit time is adjusted in accordance with a quality of service afforded by one session versus a quality of service afforded by the other session. The second device orders the data payloads of the packets, thereby reconstructing the single media stream.” Kottilingal at Abstract.

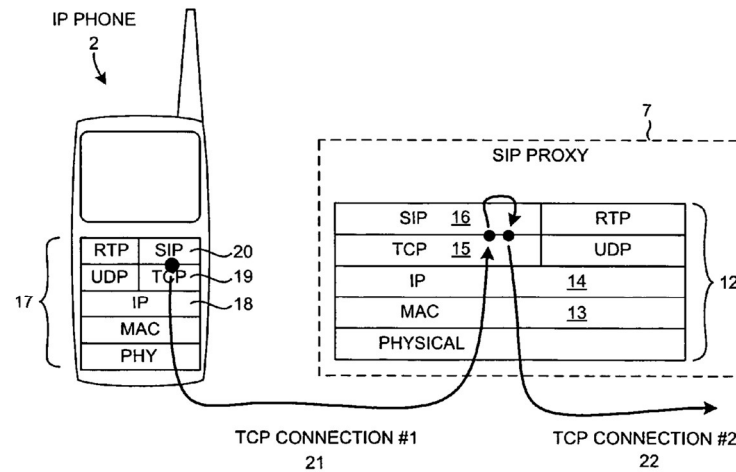


FIG. 2

Kottilingal at Figure 2.

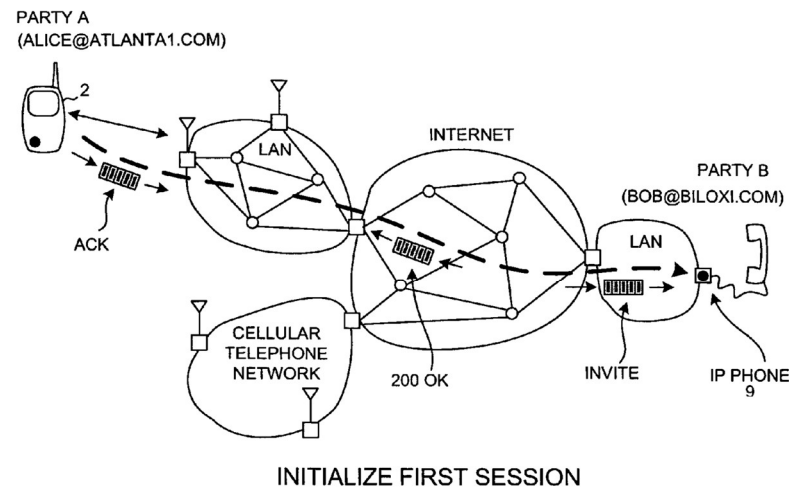
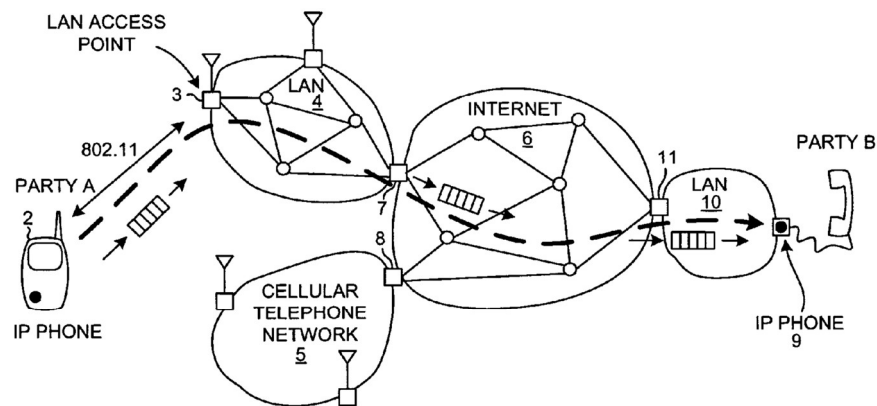


FIG. 5

Kottilingal at Figure 5.



FIRST SESSION ACTIVE -
PARTY A MOVES AWAY FROM LAN ACCESS POINT

FIG. 6

Kottilingal at Figure 6.

```

SPAWN SIP:BOB@BILOXI.COM SIP/2.0
VIA: SIP/2.0/UDP PC33.ATLANTA2.COM;BRANCH=ADFS
MAX-FORWARDS: 70
TO: BOB <SIP:BOB@BILOXI.COM>
FROM: ALICE <SIP:ALICE@ATLANTA2.COM>;TAG=876767776
CALL-ID: D7878-0-A2@PC33.ATLANTA2.COM
CSEQ: 314160 SPAWN
CONTACT: SIP:ALICE@PC33.ATLANTA2.COM.
CONTENT-TYPE: APPLICATION/SDP
CONTENT-LENGTH: 142

```

```

MASTER VIA: PC33.ATLANTA1.COM;BRANCH=Z9HG4BK776ASDHDS
MASTER TO: BOB <SIP:BOB@BILOXI.COM>
MASTER FROM: ALICE <SIP:ALICE@ATLANTA1.COM>;TAG=1928301774
MASTER CALL-ID: A84B4C76E66710@PC33.ATLANTA1.COM
MASTER CSEQ: 314159 INVITE
MASTER CONTACT: <SIP:ALICE@PC33.ATLANTA.COM>

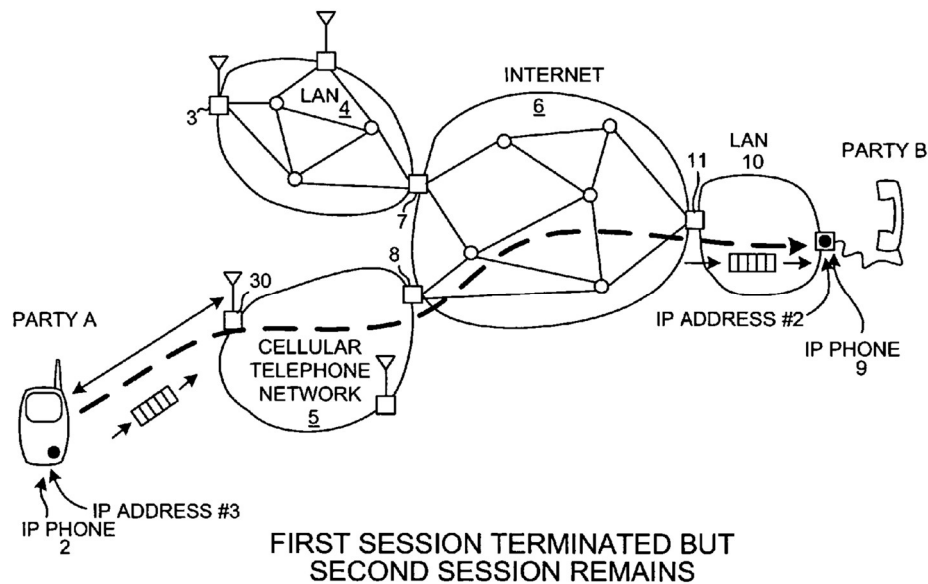
```

SO CALLEE CAN
IDENTIFY WITH
FIRST SESSION

SPAWN MESSAGE

FIG. 7

Kottilingal at Figure 7.



Kottilingal at Figure 17.

“It is then desired to continue the call using the other air interface of the mobile communication device. This may, for example, be due to the air interface initially being used being a short range wireless LAN interface. PARTY A may move out of the coverage area of the short range wireless LAN. It is desired to continue the call by switching to using the longer range cellular telephone air interface. Alternatively, it may be desired to switch from using the first air interface to the second air interface where the first interface is a longer range cellular telephone interface and the second air interface is a shorter range wireless LAN interface. Initially the cellular telephone air interface is used, but then PARTY A moves into the coverage area of the wireless LAN. If, for example, PARTY A's cellular telephone provider charges to carry a voice conversation on its cellular telephone network, then it may be desirable for PARTY A to stop using the cellular air interface and to continue the call using the less expensive wireless LAN air interface.” Kottilingal at 2:7–24.

“Once the second session is initialized, the mobile communication device stops transmitting VoIP packets for the media stream in the first session and transmits subsequent VoIP packets for the media stream in the second session. In some embodiments, a handoff control packet is sent from the mobile communication device to the target to alert the target that subsequent VoIP packets will no longer be received in the first session but rather will be received in the second session. In other embodiments, the target determines that the second session is now being used to communicate data payload VoIP packets because data payload VoIP packets for the media stream are no longer being

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		received by the target in the first session but rather data payload VoIP packets for the media stream are now being received by the target in the second session. Regardless of how the target determines that VoIP packets are now being communicated in the second session, the flow of VoIP packets in both directions between PARTY A's communication device and PARTY B's communication device now occurs in the second session and not the first session.” Kottilingal at 2:42–61. “In a VoIP/VIP call, the mobile communication device uses its first air interface to communicate VoIP/VIP packets in a first session from the mobile communication device to a second IP device. Software in the mobile communication device makes periodic assessments on a quality of service afforded by the first session. If the assessments indicate that the quality of service has decreased to an undesirable or unacceptable level, and if it is possible to use the second air interface, then the mobile communication device causes a second session to be set up between the mobile communication device and the second IP device, where the second session involves communication using the second air interface of the mobile communication device.” Kottilingal at 3:19–31. “FIG. 6 illustrates the communication of some of these first VoIP/VIP IP data payload packets between IP phone 2 and IP phone 9. This communication involves wireless communication in accordance with the 802.11 protocol between IP phone 2 and access point 3 of LAN 4. 802.11 is a relatively short range RF communication protocol. In the present example, PARTY A moves farther and farther away from access point 3. IP phone 2 detects the signal strength of RF transmissions received from access point 3. In the present example, an indication of the detected signal strength is available on the IP phone 2 as an RSSI (Radio Signal Strength Indicator) signal that is output from the receiver amplifier of the 802.11 transceiver in IP phone 2. The receiver amplifier of the 802.11 transceiver within access point 3 also detects the signal strength of RF transmissions received from IP phone 2 and this detected signal strength is reported from access point 3 back to IP phone 2. IP phone 2 is therefore cognizant of the strength of received transmissions in both directions. As PARTY A and IP phone 2 move away from access point 3, the detected signal strength of the 802.11 wireless link between IP phone 2 and access point 3 degrades until a threshold is reached. Once the threshold is reached, IP phone 2 determines that IP phone 2 should initialize a second SIP session using its long range cellular telephone transceiver. IP phone 2 initializes the second SIP session by sending a novel SIP message called a SPAWN message from PARTY A to PARTY B in the first session.” Kottilingal at 8:18–44. “FIG. 8 is a diagram that illustrates the communication of this SPAWN SIP message from PARTY A to PARTY B. In the same manner as the INVITE SIP message is communicated from PARTY A

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		to PARTY B as described above, so too is the SPAWN SIP message communicated from IP phone 2, across the 802.11 wireless link to access point 3 on LAN 4, across LAN 4 to SIP proxy 7, across Internet 6 and to SIP proxy 11, and across LAN 11 and to PARTY B's IP phone 9. PARTY B's IP phone 9 receives the SPAWN SIP message and responds by sending a 200 OK SIP message back to PARTY A's IP phone 2. This 200 OK SIP message, however, includes a SPAWN ID (spawn identifier). IP phone 9 generates the SPAWN ID by generating a hash of the FROM field, the TO field, the CALL-ID field and the CSEQ number of the first session. The SPAWN-ID is stored in IP phone 9 for future reference to associate a second session with the first session. In the present example, the SPAWN ID is a sixteen-byte character string that follows a field name “SPAWN-ID:” in the 200 OK SIP message. The 200 OK SIP message that includes the SPAWN ID is forwarded back to IP phone 2 through SIP proxy 11 and SIP proxy 7.” Kottilingal at 8:47–67. “Although IP phone 9 in the example described above is a landline IP phone, IP phone 9 is a mobile wireless communication device (for example, a cellular telephone) in another example. The first and second sessions can be initiated by either mobile or landline IP phones.” Kottilingal at 11:33–37. “Next (step 101), PARTY A moves IP phone 2 away from the wireless access point 3 on first network 4 as explained above in connection with FIG. 6. As a consequence of the movement, a quality of service afforded by the wireless communication link between IP phone 2 and the wireless access point 3 is degraded. This decrease in quality of service may, for example, involve one or more of the following: a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 2 to IP phone 9 in the first session, a decrease in the data payload bit rate (bits per second) that can be communicated from IP phone 9 to IP phone 2 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 2 to IP phone 9 in the first session, a decrease in the total number of RTP packets communicated per unit time from IP phone 9 to IP phone 2 in the first session, an increase in a number of RTP packets lost during a communication in the first session from IP phone 2 to IP phone 9, an increase in a number of RTP packets lost during a communication in the first session from IP phone 9 to IP phone 2, an increase in jitter between the arrival of incoming packets either into IP phone 2 or into IP phone 9, an increase in the time required (sometimes referred to as latency) to communicate packets end-to-end from IP phone 2 to IP phone 9, an increase in the time required to communicate packets end-to-end from IP phone 9 to IP phone 2. The decrease in quality of service may be manifest in reduced RF signal strength as detected by IP phone 2, by access point 3, or both.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 (“Kottilingal”)
		IP phone 2 detects (step 101) the decrease in quality of service using one or more manifestation of quality of service. There are many ways that IP phone 2 can detect the decrease in quality of service. In one example, IP phone 2 receives RTCP (Real-Time Control Protocol) receiver report (RR) packets in the first session from IP phone 9. These RTCP receiver reports contain a total number of packets received onto IP phone 9 in the first session. IP phone 2 determines a number of packets received into IP phone 9 in an amount of time by subtracting the total number of packets value in an initial RTCP receiver report from the total number of packets value in a subsequent RTCP receiver report. IP phone 2 determines a time difference between the two RTCP reports, and then divides the determined number of packets received in the amount of time from the time difference, thereby obtaining an approximate number of packets received into IP phone 9 per unit time. This value in one embodiment is used as an assessment of quality of service.” Kottilingal at 12:1–47. “IP phone 2 performs this operation to detect a decrease in quality of service periodically on a regular basis to track the number of packets received onto IP phone 9 in the first session per unit time. When the number of packets received onto IP phone 9 per unit time drops below a threshold level, then IP phone 2 causes a second VoIP session (step 102) to be set up for communication between IP phone 2 and IP phone 9. The second session involves communication between IP phone 2 and an access point 3 on cellular telephone network 5 using the second air interface of IP phone 2. An alternative way of detecting a decrease in quality of service involves the use of the lost fraction (fraction of the packets lost) fields, the lost count fields, and/or the inter-arrival jitter fields of consecutive RTCP receiver reports. From the values in these fields, the total number of packets lost and the inter-arrival jitter is monitored over time. If the number of packets lost per unit time increases beyond a threshold level, or if the inter-arrival jitter increases beyond a threshold level, then a determination is made the quality of service has decreased.” Kottilingal at 12:48–67. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
		Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
20	The method of claim 15 wherein the source device is a mobile device.	Kottilingal expressly or inherently discloses this claim element. See above re claim 6 which is incorporated by reference herein.
21	The method of claim 20 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Kottilingal expressly or inherently discloses this claim element. See above re claim 7 which is incorporated by reference herein.
22[Pre]	A communications network comprising	To the extent the preamble is limiting, Kottilingal expressly or inherently discloses this claim element. See above re claim 1[Pre] which is incorporated by reference herein.
22[a]	a transmitting device; a receiving device; a first network device; a second network device;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[a] which is incorporated by reference herein.
22[b]	the transmitting device and receiving device operable to maintain a data transmission over a first network path that is made up of one or more network links where at least one of the network links is to a local area network is remote from the transmitting device;	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[a] which is incorporated by reference herein.
22[c]	wherein the first network device is operable to switch the	Kottilingal expressly or inherently discloses this claim element.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
	transmission from the first network path to one of a plurality of alternative network paths without disrupting the transmission; and said the second network device is operable to perform a Port Address Translation (PAT) on said transmission;	See above re claim 1[b] and 1[c] which are incorporated by reference herein.
22[d]	said transmission contains audio or video content	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[c] which is incorporated by reference herein.
22[e]	said transmission is made using the Real Time Protocol (RTP); said transmission further uses the Session Initiation Protocol (SIP); and	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[d] which is incorporated by reference herein.
22[f]	said transmission's SIP signaling sequence is maintained after said switching from the first network path to the one of a plurality of alternative network paths.	Kottilingal expressly or inherently discloses this claim element. See above re claim 1[e] which is incorporated by reference herein.
23	The system of claim 22 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of a stream quality event.	Kottilingal expressly or inherently discloses this claim element. See above re claim 2 which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. Patent No. 8,467,377 ("Kottilingal")
24	The system of claim 22 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	Kottilingal expressly or inherently discloses this claim element. See above re claim 17 which is incorporated by reference herein.
25	The system of claim 22 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of an administrator move.	Kottilingal expressly or inherently discloses this claim element. See above re claim 18 which is incorporated by reference herein.
26	The system of claim 22 wherein a SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Kottilingal expressly or inherently discloses this claim element. See above re claim 19 which is incorporated by reference herein.
27	The system of claim 22 wherein the transmitting device is a mobile device.	Kottilingal expressly or inherently discloses this claim element. See above re claim 6 which is incorporated by reference herein.
28	The system of claim 27 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Kottilingal expressly or inherently discloses this claim element. See above re claim 7 which is incorporated by reference herein.

**Exhibit A-26: Invalidity of U.S. Patent No. 8,009,554 in view of
U.S. 2005/0286466 (“Tagg”)**

Based upon Plaintiff’s Complaint, Infringement Contentions, and apparent claim constructions and application of the claims to Defendants’ accused products, as best as they can be deciphered, the reference charted below anticipates or at least renders obvious the asserted claims. These invalidity contentions are not an admission by Defendants that the accused products are covered by or infringe the asserted claims, particularly when these claims are properly construed and applied. These invalidity contentions are not an admission that Defendants concede or acquiesce to any claim construction implied or suggested by Plaintiff’s Complaint or Infringement Contentions. Nor are Defendants asserting any claim construction positions through these charts, including whether the preamble is a limitation. The portions of the prior art reference cited below are not exhaustive but are exemplary in nature.

U.S. 2005/0286466 (“Tagg”) was filed on May 12, 2005, published on December 29, 2005, issued as U.S. 9,049,042 on June 2, 2015, and claims priority to U.S. Application No. 09/706,406, filed on November 3, 2000. This patent is prior art under at least 35 U.S.C. § 102(b). As described in the following claim chart, the asserted claims of U.S. Patent No. 8,009,554 (the “’554 Patent”) are invalid as anticipated by Tagg.

To the extent that Tagg is found not to anticipate one or more of the asserted claims of the ’554 Patent, these claims are invalid as obvious in view of Tagg alone or in combination with other prior art references disclosed in Defendants’ Invalidity Contentions Cover Pleading and accompanying charts, including without limitation as set forth below.

Claim No.	’554 Patent	U.S. 2005/0286466 – (“Tagg”)
1[Pre]	A method of enhancing the reliability of a transmission from a source to a destination in a communications network using a network protocol that operates on top of the Internet Protocol (IP), the method comprising:	To the extent the preamble is limiting, Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> This reference describes a system in which mobile devices, including but not limited to cellular phones, provide voice and data communication services and, in particular, providing handover, handoff, and/or seamless variations thereof between multiple mobile phones over multiple communication networks (<i>e.g.</i>, GSM cellular networks and Wi-Fi networks). Tagg explains that handoffs can be beneficial to route calls over lower-cost networks and to increase the reliability and redundancy of a transmission. “A system for providing handoff for a mobile devices comprising a mobile phone programmed to automatically handover between differing data bearers and to optimally detect those bearers in a roaming environment keeping power consumption to a minimum. Repeating means for these

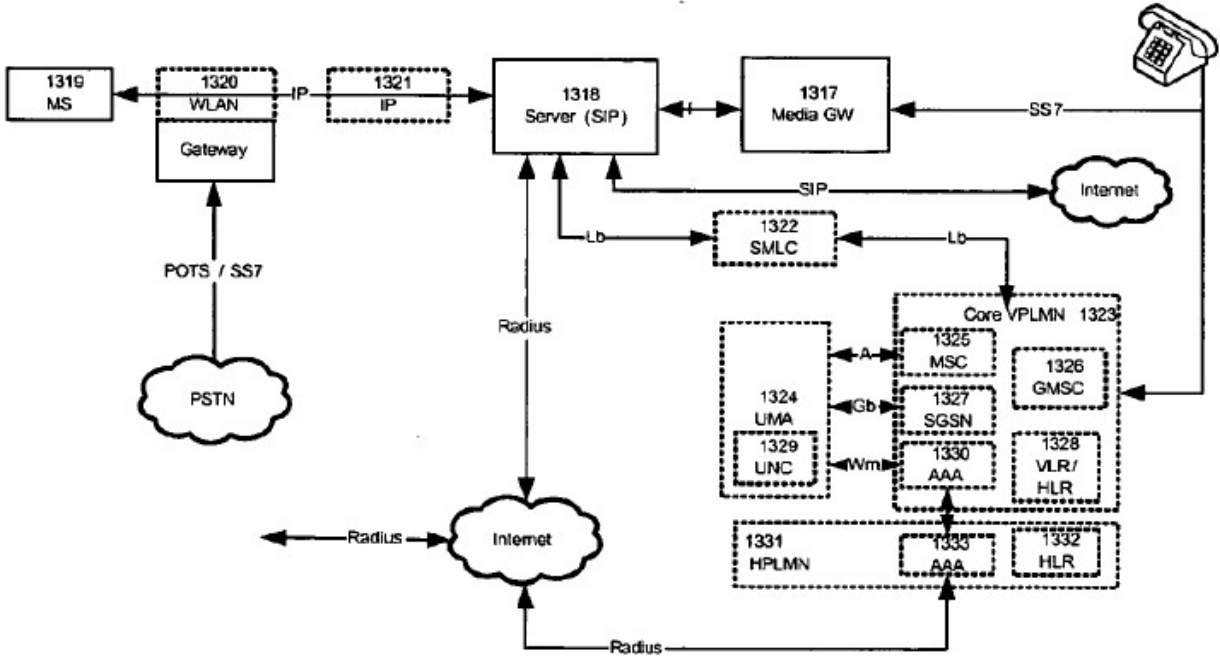
Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		mobile devices to extend the range of coverage and the protocol for that coverage.” Tagg at Abstract. “Another significant advantage of a packet based system is that the packets may follow different routing paths providing for a high degree of redundancy and therefore reliability. For example, a packet may be forwarded at the radio layer via the Bluetooth, Wi-Fi and UMTS link simultaneously during handoff/over in the invented system. While UMA systems are also packet based systems over the radio layer, those packets are only comprehensible by a UNC (UMA Network Controller) and must transit through it for translation back into core cellular protocols. This introduces a compulsory routing node and therefore a single point of failure. Must of the advantage of an IP based packet system is therefore lost.” Tagg at [0091]. The reference discloses enhancing the reliability of transmissions between mobile devices (<i>e.g.</i>, from a source to a destination) by initiating handoffs when a wireless communication device moves between different networks. The reference additionally discloses using various network protocols that operate on top of the Internet Protocol (IP), including various types of packet-based protocols as would be known to a person of ordinary skill in the art. “An embodiment of the present invention provides a method and system for connecting to a network—typically the Internet via a cooperative membership arrangement. Members of the cooperative networking service have an identifier that uniquely identifies them as a member of the service. A fixed member (host) enables its link capabilities for mobile users (clients) but directs those links to communicate with a secure piece of software—the Cooperative Tunneling Agent (CTA). Upon access by a mobile member, the CTA sends both identifiers and status information to a server. The server contains preference information on all members that can be referenced by their identifiers. The status information is compared to the preferences of the two members and, if the comparison is successful, a link to the Internet is provided to the mobile member. If the comparison fails the members may be notified and allowed to modify the preferences, make a one-off exception or reject the link. The CTA can work with or without reference to a server. When working independently of the server the comparison is made using data from the host and the client and the server is only accessed if additional information or service is required. The CTA ensures that the mobile user has no access to the fixed user's computer other than as a pipe thru to the Internet. To ensure that the mobile user's security and privacy are protected the data packets may be encrypted and sent via the fixed host to the server. The server may decrypts the data and sends it to the Internet. Return data is similarly dealt with. By this method the fixed member cannot eavesdrop on

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		the mobile member's data or voice traffic. Such encryption can be set at a variety of strengths according to user preference and applied selectively depending on the data type. Routing of the data may also be performed selectively according to data type.” Tagg at [0076]. “Once the link has been established the cooperative tunneling agent examines the preferences of the roaming member and the fixed member and compares these with a database stored on the server. If the comparison generates a match then standard IP packets are encrypted and sent from the roaming device 106, thru the fixed device 104 or 105 thru any present firewall 103 thru the Internet 102 to our server via a link to the Internet 111. Our server 101 decrypts the packets and sends them onto the Internet via the link. Return packets from the Internet are passed back to the server and are similarly routed back to the member via the encrypted links.” Tagg at [0209].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		Figure 1 illustrates the general layout of Tagg’s disclosed network, including IP communications over the Internet. Fig. 1 “These preferences may be appropriate for the corporation but wholly inappropriate for the member requesting the tunnel. Therefore the tunneling agent can encode all packets and sends them thru port 80 bypassing any corporate specific snooping of the data. If the UDP (“user datagram protocol”) port is available this may be preferentially used for sending VoIP and other real time packets.” Tagg at [0211].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		“FIG. 10 illustrates a wireless VoIP system for a wireless enabled home, office or public environment using commonly available systems such as GSM, SIP and UMTS (3G) but other similar systems might be used in substantially the same way for example CDMA (Code Division Multiple Access) and the H323 protocol. A mobile phone 1001 is present in the environment of a PC 1013, a wireless LAN access point 1002, GSM network 1004 and 3G network 1003. Each system is connected to the PSTN 1012 by differing routes.” Tagg at [0234]. “A call may be made via the PC 1013 or WLAN AP 1002 by way of one or more switches (or routers) 1005 connected to the Internet 1008 and through a SIP proxy server 1009 a SIP server 1010 and a media gateway 1011. For the 3G network 1003, packets must transit the telco packet switch network 1006 prior to reaching the Internet 1008, while the GSM network 1004 is connected via the telco circuit switch infrastructure 1007 directly to the PSTN 1012. The range may be extended and the protocols translated by a repeater 1014.” Tagg at [0235]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[a]	transmitting data between a source and destination over a first network path made up of one or more network links where at least one of the network links is to a local area network that is remote from the transmission source; and	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> <i>See</i> discussion with respect to 1[pre], above. This reference shows a wireless communication device can transmit data between a communication unit and a peer communication unit over multiple networks, including a wireless local area network (WLAN) and a cellular wide area network (cellular WAN or WAN). This reference explains that

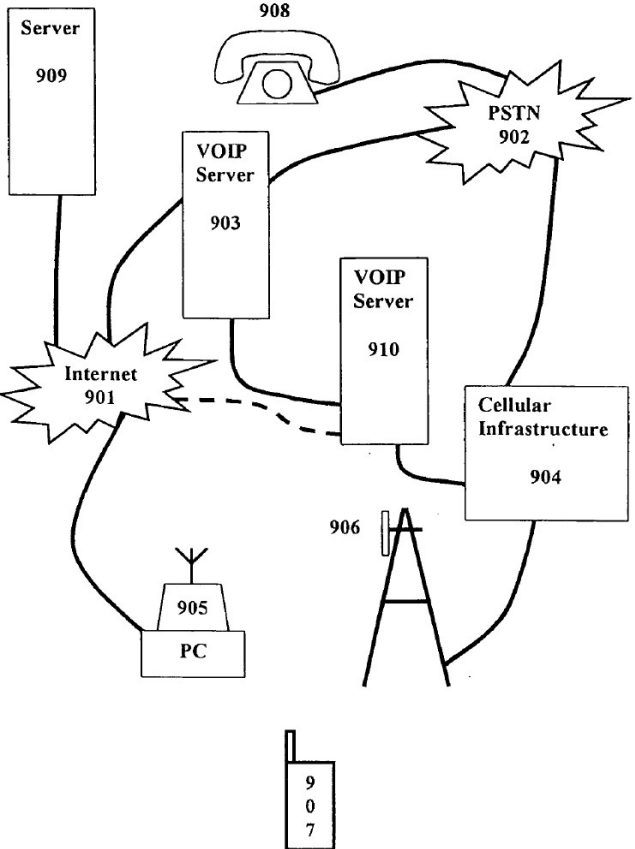
Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		the communication unit is a wireless communication unit, such that it is remote from both the WLAN and WAN. This reference further explains that both the WLAN and the WAN may be formed from network elements. The reference explains that the WLAN may be formed from large number of access points, as well as typical network routing equipment such as servers, concentrators, routers, and so on. The reference further explains that the WAN may be formed from a plurality of base transmitters, antenna towers, base site controllers, various registrars, billing servers, and so on. “The present invention relates to a system that provides VoIP over wireless LAN and WAN networks.” Tagg at [0002]. “The system attempts to look for the shortest signal path, thus the handset can uses peer-to peer-techniques to discover if another handset is in range. This may be achieved directly (for direct communication over the local radio link) where the telephone number is resolved to an IP address range and location and Bluetooth address or Wi-Fi MAC address. In this case the device or server can determine that the telephone number called is actually located in close proximity and route the traffic directly without even needing to transit an access point or repeater. The handset or server may track up the address range until it finds a short path for example or via a local repeater or communication over the LAN only. Failing this, if the SIP server detects that two phones are presently logged onto the system and that one phone is trying to ‘call’ the other by entering its phone number it may link the two phones by the shortest possible path without sending the packets into the PSTN. This in conjunction with presence services can give very rich user experience.” Tagg at [0015]. “A further advantage is that voice calls can be sent to the host and routed over the EP network using VoIP technology thus dramatically reducing costs. A further advantage is that voice calls can be routed thru the cellular system to a local VoIP server thus saving long distance cellular charges.” Tagg at [0078]. Figure 13b illustrates various network paths for the transmission between a source and a transmission, including a mobile subscriber 1319, a WLAN router/gateway 1320, SIP servers 1318, a media gateway 1317, and so on:

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		 Figure 13b (proposed invention). “b) In the proposed invention packets initially take the same path as in a UMA solution, originating from an MS 1319, traveling over a WLAN router 1320 via an IP connection 421, but then arriving first at our server that supports SIP 1318. The packets themselves are simpler in format (see FIG. 26) than in a UMA system as less protocol encapsulation is required. The SIP server is the main centralized control element in our system, interfacing with all the other systems and managing roaming, handoff and location management, but the handset can take control in a number of cases and obviate the necessity for a server, providing peer-to-peer capabilities that are not envisaged by UMA.” Tagg at [0265].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		“The SIP server in the invented system performs a number of tasks which are more fully described later in FIG. 15. To make a call the SIP server communicates via a media gateway function 1317 over SS7 or if the other system supports a SIP interface directly using SIP. A SIP connection to a cellular core network GMSC (Gateway Mobile Switching Centre) 1326 is shown for example. Authentication is provided using RADIUS (Remote Authentication Dial-In User Service) to ANY AAA (Access, Authorization, and Accounting) server and might include RADIUS proxy authentication calls to the cellular core network 1330, 1333 or any other AAA server that the user specifies (including one that is part of the SIP server 1318). The core network VPLMN (Visited Public Land Mobile Network) 1323 pictured is out of scope for our system, as is the UMA 1324, UNC 1329 block, so these are not further described. The WLAN router 1320 may contain a subset of the SIP server 1318 and media gateway 1317, such that it is able to route calls directly to the PSTN locally.” Tagg at [0266]. Tagg additionally discusses that a transmitting device is a mobile phone and is thereby remote from a network access. “The present Invention provides a means for wirelessly enabling a home, office or public environment such that commonly available hardware, such as PCs and mobile phones, may be easily and readily used in conjunction with simple low cost repeaters, PCs enabled as access points and one or more wireless access points, providing a complete telephony solution that allows a person to use one phone handset for all their telecommunications needs. The system solves a range of problems with previous systems.” Tagg at [0086]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
1[b]	switching the transmission from the first network path to one of a plurality of alternative network paths;	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> <i>See</i> discussion with respect to 1[pre] and 1[a], above. This reference discloses systems for providing a handoff for a mobile device, where a handoff switches transmissions from a mobile device from one network path to another network path. “A system for providing handoff for a mobile devices comprising a mobile phone programmed to automatically handover between differing data bearers and to optimally detect those bearers in a roaming environment keeping power consumption to a minimum. Repeating means for these mobile devices to extend the range of coverage and the protocol for that coverage.” Tagg at Abstract. “A number of methods have been proposed for implementing mobile voice and data systems so that they operate on both WWAN and WLAN based infrastructure. In implementing such mobile voice and data systems, these systems will have to deal with the problems of roaming and handoff between them. Roaming is the feature where a mobile phone can acquire a radio connection and signal to an authentication method to request a connection to services, while handoff is the feature where a mobile phone moves from one radio connection to the next maintaining a voice call or data connection with as little interruption as possible. When this interruption is not noticeable to the user the handoff is deemed “seamless.” Seamless handoff is not always an advantage. For example, when the user moves from a low cost connection to a high cost connection they generally want the transition to be made clear. The ability to do seamless handoff is an important feature of these systems even if, on occasion, it is deliberately switched off or the handoff made non-seamless through the use of notifications. When a mobile phone moves from one radio connection to another and the bearer technologies are the same we will hereafter refer to this as handover to differentiate it from handoff—for example when moving from one Wi-Fi access point to another.” Tagg at [0037]. “Another significant advantage of a packet based system is that the packets may follow different routing paths providing for a high degree of redundancy and therefore reliability. For example, a packet may be forwarded at the radio layer via the Bluetooth, Wi-Fi and UMTS link

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		simultaneously during handoff/over in the invented system. While UMA systems are also packet based systems over the radio layer, those packets are only comprehensible by a UNC (UMA Network Controller) and must transit through it for translation back into core cellular protocols. This introduces a compulsory routing node and therefore a single point of failure. Must of the advantage of an IP based packet system is therefore lost.” Tagg at [0091]. “In the previously available systems it has proven difficult to implement seamless handoff between cells or infrastructure. In both the UMA and CTP systems the audio is encoded in substantially different ways for the different radio paths—although efforts have been made in UMA to make these as similar as possible. As handoff occurs, the system must be re-configured as quickly as possible to implement the new coding scheme for audio transmission. In the invented system, the protocol is encoded in the same way regardless of which radio connection method is used. This means that without burdening system resources the handoff can be implemented quickly as no re-configuration is required or the packets can be simultaneously sent over both radio connections during the handoff to provide seamless handoff.” Tagg at [0120]. “FIG. 9 illustrates handoff between a fixed wireless, Internet based, VoIP system and the cellular system. A mobile user 907 is within range of two methods for placing a call; a PC running our cooperative networking service and a cellular tower. The call might be preferentially placed to either unit based on the users pre-set preferences or based on the current situation. Both links could be made or broken independently and exist simultaneously. In the case of connection made over the Internet voice packets are sent over the air using a wireless link such as Bluetooth or IEEE802. I to the host 905 These packets are routed thru the Internet 901 to a VoIP server 903. The VoIP server converts IP packets to a form suitable for use over the PSTN and handles making and breaking the connection to users. There are many VoIP servers, two are shown here, located in each region such that the call is placed at a local rate. If the user moves away from the fixed wireless link 905 or the Internet becomes congested such that call is no longer of sufficient quality, a factor that can be determined either automatically or by the user the call can be dialed on the cellular link. There are a number of methods that handoff could take. Firstly the call could be dropped by the VoIP server and a new call initiated end-to-end thru the cellular network to the destination 908. This would result in a break in the call. More suitably the mobile device would call a VoIP, either the existing server 903 or a server located near to the caller 910. The VoIP server would take information from the cooperative network server and route the call to the already established PSTN link and then to the destination 908 thus maintaining the call. A benefit of this is that the call is not dropped,

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		interrupted or reinitialized. A further benefit is that mobile device can place a call to a local VoIP server 910 which forwards the information to a VoIP server local to the receiving party thus saving cost. The call routing between the two VoIP servers can be via the public Internet (dashed line) or thru a proprietary network.” Tagg at [0227]. Figure 9 of this reference illustrates a handoff between a first network and a second network:  Fig. 9 “One of the problems that have to be accommodated when utilizing both WWAN and WLAN networks is the problem of handing off the call between the networks. With regard to FIG. 14, handoff cases will now be described with reference to a state diagram but firstly we should define

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[c]	wherein said transmission is not disrupted as a result of said switch to the one of a plurality of alternative network paths, said transmission contains audio or video data and at least one Network Address Translation (NAT) is performed on said transmission;	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> This reference discloses that a transmission is not disrupted as a result from a switch to one of a plurality of alternative network paths (<i>e.g.</i>, a “seamless handoff”). “Seamless handoff is where:— I. The user does not have to do anything II. There is no break in the call You can also have seamless handoff with the audio visual warnings but without requiring the user to actually press a button to choose what they want to do unless they choose not to handoff.” Tagg at [0269]. “As edge of range is detected by way of either, Receive Signal Strength Indication (RSSI), Bit Error Rate (BER) or Network Location Services (NLS), the phone will initiate handoff using either a seamless or non-seamless method. a) Seamless handoff is achieved by either rapidly switching the IP stream from the WLAN socket to the WWAN socket connection ‘break-before-make’ OR, copying the IP stream to two sockets and once end-to-end connectivity is proven stopping the stream to the WLAN socket ‘make-before-break’. b) A non seamless handoff is achieved by using ‘break-before-make’ handoff and notifying the user using some form of audio or visual indication that the WLAN connection is becoming or has become unavailable and giving them the option to either move back into coverage, accept the handoff, or terminate the call. This allows the user to move back into coverage and deliberately avoid handoff. Users rapidly learn the limits of range within their environment.” Tagg at [0280].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		This reference discloses that a transmission contains audio or video data. In addition, mobile devices with the capability to transmit audio or video data (or a combination of the two) would have been well known to a person of ordinary skill in the art as of the priority date of this reference. “The link can transport either data or voice. In the case of voice the CTA handles the VoIP (voice over IP) protocol. VoIP and Multimedia can be either sent via the complete path thru our secure server or split apart to take a more optimum routing.” Tagg at [0081]. “Multimedia subsystems within phones are also capable of recording and rendering pictures and video.” Tagg at [0089]. “Once a SIP (Session Initiation Protocol) connection has been made, the low level DSP (Digital Signal Processing) subsystem of the phone provides its audio streams to the high level multimedia subsystem within the phone which encodes the voice into RTP (Real-time Transport Protocol) packets. VoIP data (RTP stream plus signaling) is sent over the Bluetooth data link to an access point (AP), for example WLAN AP 1002, which routes the VoIP packets to a SIP server through the Internet 1008.” Tagg at [0238]. In addition, this reference discusses performing Network Address Translation (NAT) on the transmission. Further, performing NAT on a transmission would have been well known to a person of ordinary skill in the art as of the priority date of this reference. For example, routers were known to perform NAT to enable private IP networks to use a public network, like the Internet. Plaintiff’s Infringement Contentions rely on NAT processes of conventional equipment (e.g., a TWAG and Ports 500 and 4500/UDP, which Plaintiff contends is required for NAT traversal). “The WLAN repeater/translator further simplifies the home/office wiring problem by allowing wired phones to plug into the repeater with a suitable RJ45, RJ11 or BT plug, thus eliminating the need to provide a wiring system for fixed phones within the home and bonding the home system to the wireless VoIP infrastructure.” Tagg at [0101]. “Another significant problem that is overcome by the repeater is Network Address Translation (NAT) traversal. Many computer networks block certain ports and allow incoming IP only as a direct response to outgoing IP. The repeater makes regular contact with the system server and thus keeps an incoming channel open. Additionally the repeater may perform Network Address

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		Translation of Processing, search for and switch ports to an unblocked port on the server or take other action to maintain contract with the server. Since it is generally an AC powered, high power device these actions are not detrimental to its operation but would tax a phone in similar circumstances.” Tagg at [0102]. “FIG. 26 shows microSIP to SIP translation. In the case where the handset does not implement SIP, or implements a subset of SIP (eg. MicroSIP) the AP may need to implement a transcoding function to:— i) Provide Mobile IP mapping to allow multiple IP addresses from the different connection paths to be represented as a single IP end point to the SIP server. ii) Provide transcoding to the media streams so that any limitations in the media gateway or with the end point device associated with the SIP server can be mitigated. iii) Map audio data and subset of SIP/IP to full SIP/RTP for any endpoint device that does not support full SIP. iv) Provide IP routing and tunneling functions such as NAT traversal and VPN functions.” Tagg at [0338].

Figure 26 illustrates processes performed on an outgoing stream, including NAT traversal:

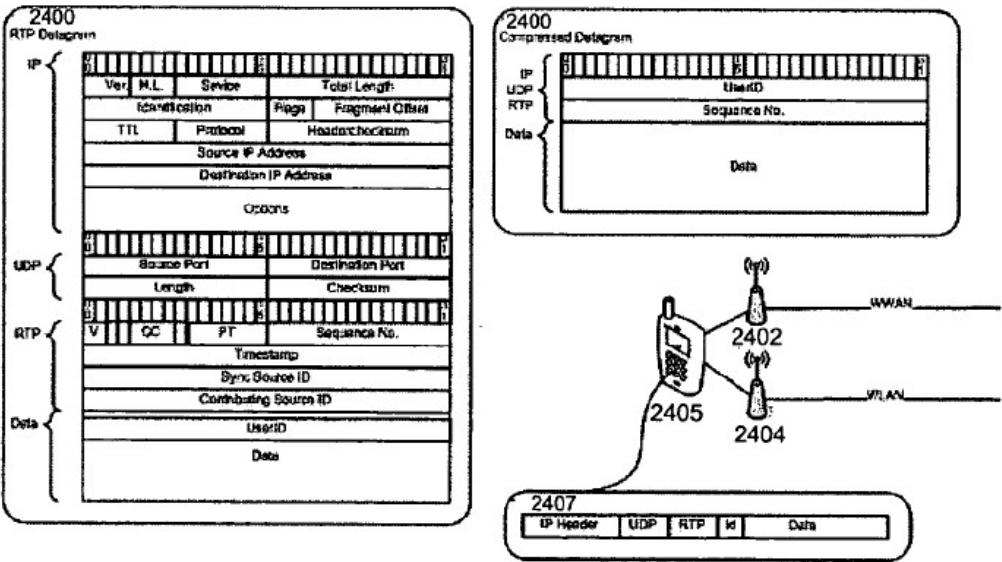


Figure 26

This reference additionally discusses the use of the User Datagram Protocol (UDP). Plaintiff alleges that certain UDP ports require NAT traversal in its Infringement Contentions.

“FIG. 24 illustrates the Network Address Processing required to process IP v4 RTP datagrams for the mobile roaming, handoff and handover. For this discussion, we will talk about RTP packets but a similar process is needed for the SIP and other packets. The RTP datagram 2400 is constructed in the normal way with IP, UDP, RTP and Data headers, hereafter referred to as an RTP packet. As can be seen each RTP packet contains source and destination information for both IP address and

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		port-together known as a socket. The phone sends RTP packets to one or more sockets during handoff, handover and other events and these packets may travel over WWAN 2402 or WLAN 2404 infrastructure prior to reaching the Internet and our server. RTP packets 2404 and 2406 may arrive from two different sockets and these need to be processes and de-duplicated into one stream prior to traveling on to the media gateway 2408. An important factor is that both the WLAN and WWAN paths will need authentication to allow streams to enter the SIP server and in addition the user name password SIP address should be authenticated against the SIM address in the phone.” Tagg at [0386]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[d]	said transmission uses the Real Time Protocol (RTP); said transmission uses the Session Initiation Protocol (SIP); and	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> <i>See</i> discussion with respect to 1[pre], 1[a], 1[b], and 1[c], above. This reference discloses that a transmission uses Real Time Protocol (RTP). In addition, a transmission from a wireless communication device using RTP would have been well known to a person of ordinary skill in the art as of the priority date of this reference. “The protocols used in the invention can be Session Initiation Protocol (SIP) and Real-time Transport Protocol (RTP) optimized for wireless transport. These are encoded at a high level within the protocol stack and transmitted down the protocol stack using IP within the phone to the different radio subsystems for transmission to the network. In this way, the audio or data protocols are radio layer agnostic. A benefit of this agnosticism is that a packet can be formatted and sent to

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		the appropriate socket (Bluetooth, Wi-Fi or UMTS) completely freely and simultaneously. The Network Address Processing in the server of the invention (a modified SIP server) will sort the streams back into a single stream for termination to the PSTN or forwarding to other parties. If the other party has a compatible client then this sorting is not needed as there is no essential restriction on having two streams (i.e. a socket pair rather than a single socket) connecting mobile subscribers in the SIP domain, and this approach can be used to improve QoS and the seamlessness of handover. A further advantage of the system using an all IP architecture is that handoff can be affected without interfacing with the low level stack components of a phone.” Tagg at [0096]. “FIG. 24 illustrates the Network Address Processing required to process IP v4 RTP datagrams for the mobile roaming, handoff and handover. For this discussion, we will talk about RTP packets but a similar process is needed for the SIP and other packets. The RTP datagram 2400 is constructed in the normal way with IP, UDP, RTP and Data headers, hereafter referred to as an RTP packet. As can be seen each RTP packet contains source and destination information for both IP address and port-together known as a socket. The phone sends RTP packets to one or more sockets during handoff, handover and other events and these packets may travel over WWAN 2402 or WLAN 2404 infrastructure prior to reaching the Internet and our server. RTP packets 2404 and 2406 may arrive from two different sockets and these need to be processes and de-duplicated into one stream prior to traveling on to the media gateway 2408. An important factor is that both the WLAN and WWAN paths will need authentication to allow streams to enter the SIP server and in addition the user name password SIP address should be authenticated against the SIM address in the phone.” Tagg at [0386]. “FIGS. 28 a-e illustrate the procedures carried out in both the handset and AP when making a call for different scenarios. These diagrams are representative of all the calling procedures possible and are by no means exhaustive. In particular, FIGS. 28 a and b show the instance where the SIP and RTP clients are both implemented in full on the AP. FIGS. 28 c and 28 d show the situation where the SIP and RTP clients are both implemented in full on the handset.” Tagg at [0395]. This reference discloses that a transmission uses Session Initiation Protocol (SIP). In addition, a transmission from a wireless communication device using SIP would have been well known to a person of ordinary skill in the art as of the priority date of this reference. “The protocols used in the invention can be Session Initiation Protocol (SIP) and Real-time Transport Protocol (RTP) optimized for wireless transport. These are encoded at a high level within

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		the protocol stack and transmitted down the protocol stack using IP within the phone to the different radio subsystems for transmission to the network. In this way, the audio or data protocols are radio layer agnostic. A benefit of this agnosticism is that a packet can be formatted and sent to the appropriate socket (Bluetooth, Wi-Fi or UMTS) completely freely and simultaneously. The Network Address Processing in the server of the invention (a modified SIP server) will sort the streams back into a single stream for termination to the PSTN or forwarding to other parties. If the other party has a compatible client then this sorting is not needed as there is no essential restriction on having two streams (i.e. a socket pair rather than a single socket) connecting mobile subscribers in the SIP domain, and this approach can be used to improve QoS and the seamlessness of handover. A further advantage of the system using an all IP architecture is that handoff can be affected without interfacing with the low level stack components of a phone.” Tagg at [0096]. “FIG. 24 illustrates the Network Address Processing required to process IP v4 RTP datagrams for the mobile roaming, handoff and handover. For this discussion, we will talk about RTP packets but a similar process is needed for the SIP and other packets. The RTP datagram 2400 is constructed in the normal way with IP, UDP, RTP and Data headers, hereafter referred to as an RTP packet. As can be seen each RTP packet contains source and destination information for both IP address and port-together known as a socket. The phone sends RTP packets to one or more sockets during handoff, handover and other events and these packets may travel over WWAN 2402 or WLAN 2404 infrastructure prior to reaching the Internet and our server. RTP packets 2404 and 2406 may arrive from two different sockets and these need to be processes and de-duplicated into one stream prior to traveling on to the media gateway 2408. An important factor is that both the WLAN and WWAN paths will need authentication to allow streams to enter the SIP server and in addition the user name password SIP address should be authenticated against the SIM address in the phone.” Tagg at [0386]. “FIGS. 28 a-e illustrate the procedures carried out in both the handset and AP when making a call for different scenarios. These diagrams are representative of all the calling procedures possible and are by no means exhaustive. In particular, FIGS. 28 a and b show the instance where the SIP and RTP clients are both implemented in full on the AP. FIGS. 28 c and 28 d show the situation where the SIP and RTP clients are both implemented in full on the handset.” Tagg at [0395]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
1[e]	the SIP signaling sequence of the transmission is maintained after said switching from the first network path to the one of a plurality of alternative network paths.	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> <i>See</i> discussion with respect to 1[pre], 1[a], 1[b], 1[c], and 1[d], above. This reference discloses that a transmission uses SIP in accordance with known techniques, such as disclosed in RFP 3261. “A number of methods have been proposed for implementing mobile voice and data systems so that they operate on both WWAN and WLAN based infrastructure. In implementing such mobile voice and data systems, these systems will have to deal with the problems of roaming and handoff between them. Roaming is the feature where a mobile phone can acquire a radio connection and signal to an authentication method to request a connection to services, while handoff is the feature where a mobile phone moves from one radio connection to the next maintaining a voice call or data connection with as little interruption as possible. When this interruption is not noticeable to the user the handoff is deemed “seamless.” Seamless handoff is not always an advantage. For example, when the user moves from a low cost connection to a high cost connection they generally want the transition to be made clear. The ability to do seamless handoff is an important feature of these systems even if, on occasion, it is deliberately switched off or the handoff made non-seamless through the use of notifications. When a mobile phone moves from one radio connection to another and the bearer technologies are the same we will hereafter refer to this as handover to differentiate it from handoff—for example when moving from one Wi-Fi access point to another.” Tagg at [0037].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		“b) In the proposed invention packets initially take the same path as in a UMA solution, originating from an MS 1319, traveling over a WLAN router 1320 via an IP connection 421, but then arriving first at our server that supports SIP 1318. The packets themselves are simpler in format (see FIG. 26) than in a UMA system as less protocol encapsulation is required. The SIP server is the main centralized control element in our system, interfacing with all the other systems and managing roaming, handoff and location management, but the handset can take control in a number of cases and obviate the necessity for a server, providing peer-to-peer capabilities that are not envisaged by UMA.” Tagg at [0265]. “The SIP server in the invented system performs a number of tasks which are more fully described later in FIG. 15. To make a call the SIP server communicates via a media gateway function 1317 over SS7 or if the other system supports a SIP interface directly using SIP. A SIP connection to a cellular core network GMSC (Gateway Mobile Switching Centre) 1326 is shown for example. Authentication is provided using RADIUS (Remote Authentication Dial-In User Service) to ANY AAA (Access, Authorization, and Accounting) server and might include RADIUS proxy authentication calls to the cellular core network 1330, 1333 or any other AAA server that the user specifies (including one that is part of the SIP server 1318). The core network VPLMN (Visited Public Land Mobile Network) 1323 pictured is out of scope for our system, as is the UMA 1324, UNC 1329 block, so these are not further described. The WLAN router 1320 may contain a subset of the SIP server 1318 and media gateway 1317, such that it is able to route calls directly to the PSTN locally.” Tagg at [0266]. “One of the problems that have to be accommodated when utilizing both WWAN and WLAN networks is the problem of handing off the call between the networks. With regard to FIG. 14, handoff cases will now be described with reference to a state diagram but firstly we should define what handoff and handover are and the different forms that they take. Handoff involves passing the user to a different network. Handover involves passing the user to a different radio antenna on the same network.” Tagg at [0268]. “As edge of range is detected by way of either, Receive Signal Strength Indication (RSSI), Bit Error Rate (BER) or Network Location Services (NLS), the phone will initiate handoff using either a seamless or non-seamless method.

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		a) Seamless handoff is achieved by either rapidly switching the IP stream from the WLAN socket to the WWAN socket connection ‘break-before-make’ OR, copying the IP stream to two sockets and once end-to-end connectivity is proven stopping the stream to the WLAN socket ‘make-before-break’. b) A non seamless handoff is achieved by using ‘break-before-make’ handoff and notifying the user using some form of audio or visual indication that the WLAN connection is becoming or has become unavailable and giving them the option to either move back into coverage, accept the handoff, or terminate the call. This allows the user to move back into coverage and deliberately avoid handoff. Users rapidly learn the limits of range within their environment.” Tagg at [0280]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
2	The method of claim 1 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of a stream quality event.	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> <i>See</i> 1[b] and 1[c]. “FIG. 9 illustrates handoff between a fixed wireless, Internet based, VoIP system and the cellular system. A mobile user 907 is within range of two methods for placing a call; a PC running our cooperative networking service and a cellular tower. The call might be preferentially placed to either unit based on the users pre-set preferences or based on the current situation. Both links could be made or broken independently and exist simultaneously. In the case of connection made over the Internet voice packets are sent over the air using a wireless link such as Bluetooth or IEEE802. I to the host 905 These packets are routed thru the Internet 901 to a VoIP server 903. The VoIP server converts IP packets to a form suitable for use over the PSTN and handles making and breaking the

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		connection to users. There are many VoIP servers, two are shown here, located in each region such that the call is placed at a local rate. If the user moves away from the fixed wireless link 905 or the Internet becomes congested such that call is no longer of sufficient quality, a factor that can be determined either automatically or by the user the call can be dialed on the cellular link. There are a number of methods that handoff could take. Firstly the call could be dropped by the VoIP server and a new call initiated end-to-end thru the cellular network to the destination 908. This would result in a break in the call. More suitably the mobile device would call a VoIP, either the existing server 903 or a server located near to the caller 910. The VoIP server would take information from the cooperative network server and route the call to the already established PSTN link and then to the destination 908 thus maintaining the call. A benefit of this is that the call is not dropped, interrupted or reinitialized. A further benefit is that mobile device can place a call to a local VoIP server 910 which forwards the information to a VoIP server local to the receiving party thus saving cost. The call routing between the two VoIP servers can be via the public Internet (dashed line) or thru a proprietary network.” Tagg at [0227]. (emphasis added) One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
3	The method of claim 1 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	Tagg expressly or inherently discloses this claim element. See below re claim 17 which is incorporated by reference herein.
4	The method of claim 1 wherein the switch from the first network	Tagg expressly or inherently discloses this claim element. See below re claim 18 which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
	path to the one of a plurality of alternative network paths occurs as a result of an administrator move.	
5	The method of claim 1 wherein a SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Tagg expressly or inherently discloses this claim element. See below re claim 19 which is incorporated by reference herein.
6	The method of claim 1 wherein the source device is a mobile device.	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> <i>See</i> 1[pre], 1[a], 1[b], 1[c], 1[d] and 1[e]. “FIG. 1 illustrates the general layout of the network. A roaming device, 106, which might be a mobile computer, PDA, cellular telephone or home appliance, comes within range of the fixed wireless link of a member, circles 107-110. The mobile device determines the connection methodologies available to it and their relative merits and then connects to the host using the best available standards.” Tagg at [0208]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
7	The method of claim 6 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> <i>See</i> 1[pre], 1[a], 1[b], 1[c], 1[d], and 1[e].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		Tagg discloses that a handover occurs when, for example, a communication unit moves out of the range of a cellular WAN (or vice versa). A person of ordinary skill in the art would understand that moving out of the range of a WAN can be considered “roaming,” as the user would move outside of a home WAN and into, for example, the WAN of another. “FIG. 9 illustrates handoff between a fixed wireless, Internet based, VoIP system and the cellular system. A mobile user 907 is within range of two methods for placing a call; a PC running our cooperative networking service and a cellular tower. The call might be preferentially placed to either unit based on the users pre-set preferences or based on the current situation. Both links could be made or broken independently and exist simultaneously. In the case of connection made over the Internet voice packets are sent over the air using a wireless link such as Bluetooth or IEEE802. I to the host 905 These packets are routed thru the Internet 901 to a VoIP server 903. The VoIP server converts IP packets to a form suitable for use over the PSTN and handles making and breaking the connection to users. There are many VoIP servers, two are shown here, located in each region such that the call is placed at a local rate. If the user moves away from the fixed wireless link 905 or the Internet becomes congested such that call is no longer of sufficient quality, a factor that can be determined either automatically or by the user the call can be dialed on the cellular link. There are a number of methods that handoff could take. Firstly the call could be dropped by the VoIP server and a new call initiated end-to-end thru the cellular network to the destination 908. This would result in a break in the call. More suitably the mobile device would call a VoIP, either the existing server 903 or a server located near to the caller 910. The VoIP server would take information from the cooperative network server and route the call to the already established PSTN link and then to the destination 908 thus maintaining the call. A benefit of this is that the call is not dropped, interrupted or reinitialized. A further benefit is that mobile device can place a call to a local VoIP server 910 which forwards the information to a VoIP server local to the receiving party thus saving cost. The call routing between the two VoIP servers can be via the public Internet (dashed line) or thru a proprietary network.” Tagg at [0227]. (emphasis added) “b) In the proposed invention packets initially take the same path as in a UMA solution, originating from an MS 1319, traveling over a WLAN router 1320 via an IP connection 421, but then arriving first at our server that supports SIP 1318. The packets themselves are simpler in format (see FIG. 26) than in a UMA system as less protocol encapsulation is required. The SIP server is the main centralized control element in our system, interfacing with all the other systems and managing roaming, handoff and location management, but the handset can take control in a number of cases

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		and obviate the necessity for a server, providing peer-to-peer capabilities that are not envisaged by UMA.” Tagg at [0265]. “FIG. 24 illustrates the Network Address Processing required to process IP v4 RTP datagrams for the mobile roaming, handoff and handover. For this discussion, we will talk about RTP packets but a similar process is needed for the SIP and other packets. The RTP datagram 2400 is constructed in the normal way with IP, UDP, RTP and Data headers, hereafter referred to as an RTP packet. As can be seen each RTP packet contains source and destination information for both IP address and port-together known as a socket. The phone sends RTP packets to one or more sockets during handoff, handover and other events and these packets may travel over WWAN 2402 or WLAN 2404 infrastructure prior to reaching the Internet and our server. RTP packets 2404 and 2406 may arrive from two different sockets and these need to be processes and de-duplicated into one stream prior to traveling on to the media gateway 2408. An important factor is that both the WLAN and WWAN paths will need authentication to allow streams to enter the SIP server and in addition the user name password SIP address should be authenticated against the SIM address in the phone.” Tagg at [0386]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
8[Pre]	A communications network comprising:	To the extent the preamble is limiting, Tagg expressly or inherently discloses this claim element. See above re claim 1[Pre] which is incorporated by reference herein.
8[a]	a transmitting device; a receiving device; a first network device; a second network device;	Tagg expressly or inherently discloses this claim element. See above re claim 1[a] which is incorporated by reference herein.
8[b]	the transmitting device and receiving device operable to maintain a data transmission over	Tagg expressly or inherently discloses this claim element. See above re claim 1[a] which is incorporated by reference herein.

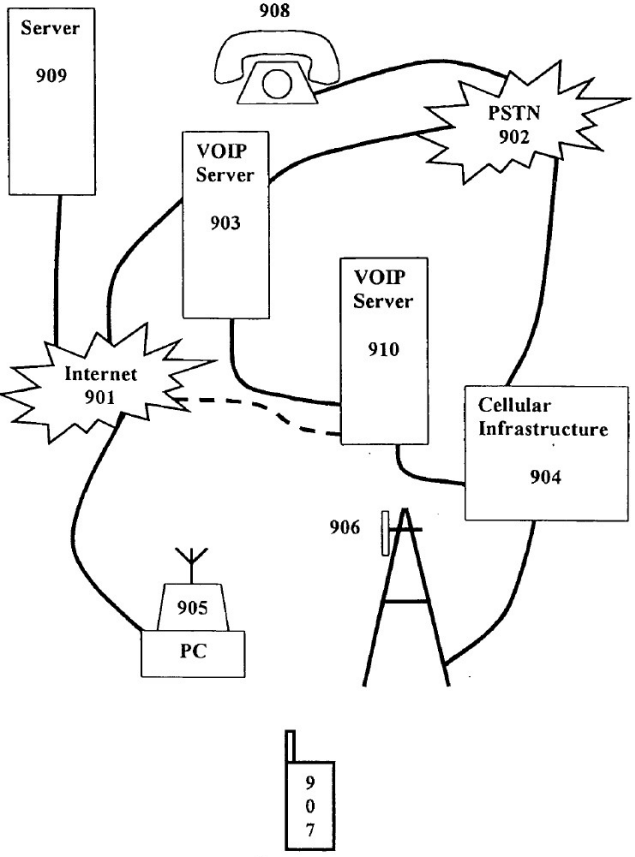
Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
	a first network path that is made up of one or more network links where at least one of the network links is to a local area network is remote from the transmitting device;	
8[c]	wherein the first network device is operable to switch the transmission from the first network path to one of a plurality of alternative network paths without disrupting the transmission; the second network device is operable to perform a Network Address Translation (NAT) on said transmission;	Tagg expressly or inherently discloses this claim element. See above re claim 1[b] and 1[c] which are incorporated by reference herein.
8[d]	said transmission contains audio or video content;	Tagg expressly or inherently discloses this claim element. See above re claim 1[c] which is incorporated by reference herein.
8[e]	said transmission is made using the Real Time Protocol (RTP); said transmission uses the Session Initiation Protocol (SIP); and	Tagg expressly or inherently discloses this claim element. See above re claim 1[d] which is incorporated by reference herein.
8[f]	said transmission's SIP signaling sequence is maintained after said switching from the first network path to the one of a plurality of alternative network paths.	Tagg expressly or inherently discloses this claim element. See above re claim 1[e] which is incorporated by reference herein.
9	The system of claim 8 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of a stream quality event.	Tagg expressly or inherently discloses this claim element. See above re claim 2 which is incorporated by reference herein.

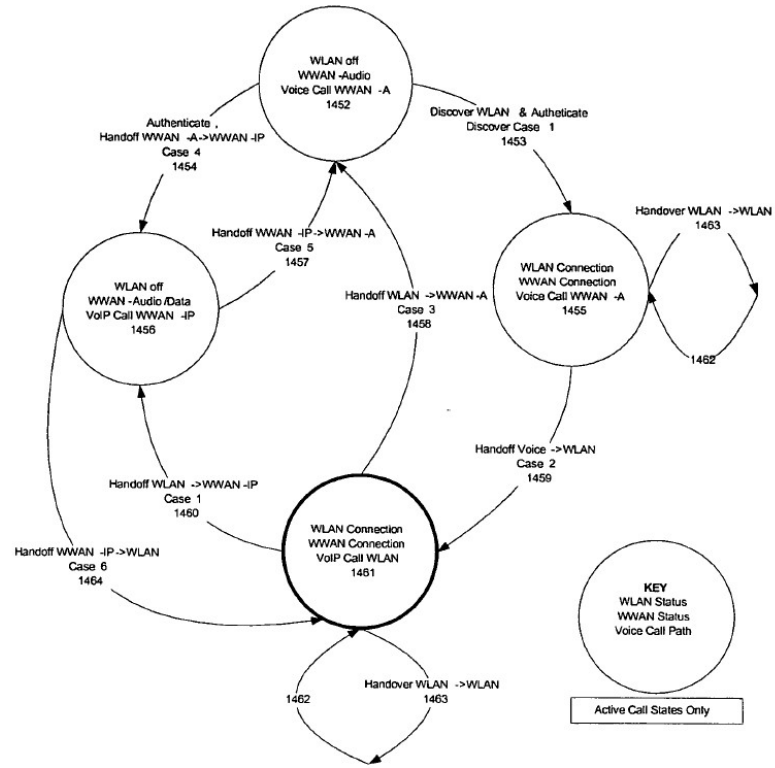
Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
10	The system of claim 8 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	Tagg expressly or inherently discloses this claim element. See below re claim 17 which is incorporated by reference herein.
11	The system of claim 8 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of an administrator move.	Tagg expressly or inherently discloses this claim element. See below re claim 18 which is incorporated by reference herein.
12	The system of claim 8 wherein the SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Tagg expressly or inherently discloses this claim element. See below re claim 19 which is incorporated by reference herein.
13	The system of claim 8 wherein the transmitting device is a mobile device.	Tagg expressly or inherently discloses this claim element. See above re claim 6 which is incorporated by reference herein.
14	The system of claim 13 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Tagg expressly or inherently discloses this claim element. See above re claim 7 which is incorporated by reference herein.
15[pre]	A method of enhancing the reliability of a transmission from a source to a destination in a communications network using a network protocol that operates on top of the Internet Protocol (IP), the method comprising:	To the extent the preamble is limiting, Tagg expressly or inherently discloses this claim element. <i>See</i> above re claim 1[pre] which is incorporated by reference herein.

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
15[a]	transmitting data between a source and destination over a first network path made up of one or more network links where at least one of the network links is to a local area network that is remote from the transmission source;	Tagg expressly or inherently discloses this claim element. <i>See</i> above re claim 1[a] which is incorporated by reference herein.
15[b]	switching the transmission from the first network path to one of a plurality of alternative network paths;	Tagg expressly or inherently discloses this claim element. <i>See</i> above re claim 1[b] which is incorporated by reference herein.
15[c]	wherein said transmission is not disrupted as a result of said switch to the one of a plurality of alternative network paths, said transmission contains audio or video data, and at least one Port Address Translation (PAT) is performed on said transmission;	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> This reference discloses systems for providing a handoff for a mobile device, where a handoff switches transmissions from a mobile device from one network path to another network path. “A system for providing handoff for a mobile devices comprising a mobile phone programmed to automatically handover between differing data bearers and to optimally detect those bearers in a roaming environment keeping power consumption to a minimum. Repeating means for these mobile devices to extend the range of coverage and the protocol for that coverage. Tagg at Abstract. “A number of methods have been proposed for implementing mobile voice and data systems so that they operate on both WWAN and WLAN based infrastructure. In implementing such mobile voice and data systems, these systems will have to deal with the problems of roaming and handoff between them. Roaming is the feature where a mobile phone can acquire a radio connection and signal to an authentication method to request a connection to services, while handoff is the feature where a mobile phone moves from one radio connection to the next maintaining a voice call or data connection with as little interruption as possible. When this interruption is not noticeable to the user the handoff is deemed “seamless.” Seamless handoff is not always an advantage. For example, when the user moves from a low cost connection to a high cost connection they generally want the

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		transition to be made clear. The ability to do seamless handoff is an important feature of these systems even if, on occasion, it is deliberately switched off or the handoff made non-seamless through the use of notifications. When a mobile phone moves from one radio connection to another and the bearer technologies are the same we will hereafter refer to this as handover to differentiate it from handoff—for example when moving from one Wi-Fi access point to another.” Tagg at [0037]. “Another significant advantage of a packet based system is that the packets may follow different routing paths providing for a high degree of redundancy and therefore reliability. For example, a packet may be forwarded at the radio layer via the Bluetooth, Wi-Fi and UMTS link simultaneously during handoff/over in the invented system. While UMA systems are also packet based systems over the radio layer, those packets are only comprehensible by a UNC (UMA Network Controller) and must transit through it for translation back into core cellular protocols. This introduces a compulsory routing node and therefore a single point of failure. Must of the advantage of an IP based packet system is therefore lost.” Tagg at [0091]. “In the previously available systems it has proven difficult to implement seamless handoff between cells or infrastructure. In both the UMA and CTP systems the audio is encoded in substantially different ways for the different radio paths—although efforts have been made in UMA to make these as similar as possible. As handoff occurs, the system must be re-configured as quickly as possible to implement the new coding scheme for audio transmission. In the invented system, the protocol is encoded in the same way regardless of which radio connection method is used. This means that without burdening system resources the handoff can be implemented quickly as no re-configuration is required or the packets can be simultaneously sent over both radio connections during the handoff to provide seamless handoff.” Tagg at [0120]. “FIG. 9 illustrates handoff between a fixed wireless, Internet based, VoIP system and the cellular system. A mobile user 907 is within range of two methods for placing a call; a PC running our cooperative networking service and a cellular tower. The call might be preferentially placed to either unit based on the users pre-set preferences or based on the current situation. Both links could be made or broken independently and exist simultaneously. In the case of connection made over the Internet voice packets are sent over the air using a wireless link such as Bluetooth or IEEE802. I to the host 905 These packets are routed thru the Internet 901 to a VoIP server 903. The VoIP server converts IP packets to a form suitable for use over the PSTN and handles making and breaking the connection to users. There are many VoIP servers, two are shown here, located in each region such

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		that the call is placed at a local rate. If the user moves away from the fixed wireless link 905 or the Internet becomes congested such that call is no longer of sufficient quality, a factor that can be determined either automatically or by the user the call can be dialed on the cellular link. There are a number of methods that handoff could take. Firstly the call could be dropped by the VoIP server and a new call initiated end-to-end thru the cellular network to the destination 908. This would result in a break in the call. More suitably the mobile device would call a VoIP, either the existing server 903 or a server located near to the caller 910. The VoIP server would take information from the cooperative network server and route the call to the already established PSTN link and then to the destination 908 thus maintaining the call. A benefit of this is that the call is not dropped, interrupted or reinitialized. A further benefit is that mobile device can place a call to a local VoIP server 910 which forwards the information to a VoIP server local to the receiving party thus saving cost. The call routing between the two VoIP servers can be via the public Internet (dashed line) or thru a proprietary network.” Tagg at [0227]. Figure 9 of this reference illustrates a handoff between a first network and a second network:

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		 The diagram illustrates a network architecture for handoff. At the top left is a 'Server 909' connected to an 'Internet 901' (represented by a starburst). A 'VOIP Server 903' is connected to both the 'Internet 901' and a 'PSTN 902' (starburst). A telephone icon '908' is connected to the 'PSTN 902'. Another 'VOIP Server 910' is connected to the 'Internet 901' and a 'Cellular Infrastructure 904'. A 'PC 905' with an antenna is connected to the 'Internet 901'. A mobile phone icon '906' is shown between the 'Internet 901' and the 'Cellular Infrastructure 904', with a dashed line indicating a connection path. Below the mobile phone is a small box containing the number '907'. Fig. 9 “One of the problems that have to be accommodated when utilizing both WWAN and WLAN networks is the problem of handing off the call between the networks. With regard to FIG. 14, handoff cases will now be described with reference to a state diagram but firstly we should define what handoff and handover are and the different forms that they take. Handoff involves passing the user to a different network. Handover involves passing the user to a different radio antenna on the same network.” Tagg at [0268].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		Figure 14 illustrates states for handoffs between various networks, including WLAN (e.g., Wi-Fi) and WWAN (e.g., cellular) networks.  Figure 14 One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
15[d]	said transmission contains audio or video content;	Tagg expressly or inherently discloses this claim element. <i>See</i> above re claim 1[c] which is incorporated by reference herein.
15[e]	said transmission contains data used in the control of the audio or video transmission;	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> This reference discloses a transmission with data to control audio and/or video content, as known in the art at the time of this application. As known in the art at the time of this reference, “data” content was well known to include control audio or video transmissions. “The link can transport either data or voice. In the case of voice the CTA handles the VoIP (voice over IP) protocol. VoIP and Multimedia can be either sent via the complete path thru our secure server or split apart to take a more optimum routing.” Tagg at [0081]. “Multimedia subsystems within phones are also capable of recording and rendering pictures and video.” Tagg at [0089]. “Once a SIP (Session Initiation Protocol) connection has been made, the low level DSP (Digital Signal Processing) subsystem of the phone provides its audio streams to the high level multimedia subsystem within the phone which encodes the voice into RTP (Real-time Transport Protocol) packets. VoIP data (RTP stream plus signaling) is sent over the Bluetooth data link to an access point (AP), for example WLAN AP 1002, which routes the VoIP packets to a SIP server through the Internet 1008.” Tagg at [0238]. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
15[f]	said transmission uses the Real Time Protocol (RTP); said transmission further uses the Session Initiation Protocol (SIP); and	Tagg expressly or inherently discloses this claim element. <i>See</i> above re claim 1[d] which is incorporated by reference herein.
15[g]	the SIP signaling sequence of the transmission is maintained after said switching from the first network path to the one of a plurality of alternative network paths.	Tagg expressly or inherently discloses this claim element. <i>See</i> above re claim 1[e] which is incorporated by reference herein.
16	The method of claim 15 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of a stream quality event.	Tagg expressly or inherently discloses this claim element. <i>See</i> above re claim 2 which is incorporated by reference herein.
17	The method of claim 15 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> Tagg discloses that a handover occurs when, for example, a communication unit moves out of the range of a cellular WAN (or vice versa) or when a signal quality of the cellular WAN (or vice versa) degrades. A person of ordinary skill in the art would understand that a signal quality may degrade when a cellular WAN (or a base station thereof) is overloaded with a large number of calls/data volume. A person of ordinary skill in the art would additionally know that one way to increase a signal quality is to route a call through a less busy base station. A person of ordinary

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		skill in the art would additionally know that load balancing was a commonly used term to refer to a process in which signal strength may be increases under periods of heavy load. “FIG. 9 illustrates handoff between a fixed wireless, Internet based, VoIP system and the cellular system. A mobile user 907 is within range of two methods for placing a call; a PC running our cooperative networking service and a cellular tower. The call might be preferentially placed to either unit based on the users pre-set preferences or based on the current situation. Both links could be made or broken independently and exist simultaneously. In the case of connection made over the Internet voice packets are sent over the air using a wireless link such as Bluetooth or IEEE802. I to the host 905 These packets are routed thru the Internet 901 to a VoIP server 903. The VoIP server converts IP packets to a form suitable for use over the PSTN and handles making and breaking the connection to users. There are many VoIP servers, two are shown here, located in each region such that the call is placed at a local rate. If the user moves away from the fixed wireless link 905 or the Internet becomes congested such that call is no longer of sufficient quality, a factor that can be determined either automatically or by the user the call can be dialed on the cellular link. There are a number of methods that handoff could take. Firstly the call could be dropped by the VoIP server and a new call initiated end-to-end thru the cellular network to the destination 908. This would result in a break in the call. More suitably the mobile device would call a VoIP, either the existing server 903 or a server located near to the caller 910. The VoIP server would take information from the cooperative network server and route the call to the already established PSTN link and then to the destination 908 thus maintaining the call. A benefit of this is that the call is not dropped, interrupted or reinitialized. A further benefit is that mobile device can place a call to a local VoIP server 910 which forwards the information to a VoIP server local to the receiving party thus saving cost. The call routing between the two VoIP servers can be via the public Internet (dashed line) or thru a proprietary network.” Tagg at [0227]. (emphasis added) One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
18	The method of claim 15 wherein the switch from the first network path to the one of a plurality of alternative network paths occurs as a result of an administrator move.	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> Tagg discloses that a handover occurs when, for example, a communication unit (and, therefore, a user/administrator of the communication unit) moves out of the range of a cellular WWAN (or vice versa) or when a signal quality of the cellular WWAN (or vice versa) degrades. A person of ordinary skill in the art would understand that this refers to an “administrator move,” as an administrator of a communication unit is physically moving to exit the range of a cellular WWAN or WLAN. “FIG. 9 illustrates handoff between a fixed wireless, Internet based, VoIP system and the cellular system. A mobile user 907 is within range of two methods for placing a call; a PC running our cooperative networking service and a cellular tower. The call might be preferentially placed to either unit based on the users pre-set preferences or based on the current situation. Both links could be made or broken independently and exist simultaneously. In the case of connection made over the Internet voice packets are sent over the air using a wireless link such as Bluetooth or IEEE802. I to the host 905 These packets are routed thru the Internet 901 to a VoIP server 903. The VoIP server converts IP packets to a form suitable for use over the PSTN and handles making and breaking the connection to users. There are many VoIP servers, two are shown here, located in each region such that the call is placed at a local rate. If the user moves away from the fixed wireless link 905 or the Internet becomes congested such that call is no longer of sufficient quality, a factor that can be determined either automatically or by the user the call can be dialed on the cellular link. There are a number of methods that handoff could take. Firstly the call could be dropped by the VoIP server and a new call initiated end-to-end thru the cellular network to the destination 908. This would result in a break in the call. More suitably the mobile device would call a VoIP, either the existing server 903 or a server located near to the caller 910. The VoIP server would take information from the cooperative network server and route the call to the already established PSTN link and then to the destination 908 thus maintaining the call. A benefit of this is that the call is not dropped, interrupted or reinitialized. A further benefit is that mobile device can place a call to a local VoIP server 910 which forwards the information to a VoIP server local to the receiving party thus saving cost. The call routing between the two VoIP servers can be via the public Internet (dashed line) or thru a proprietary network.” Tagg at [0227].

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		(emphasis added) One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
19	The method of claim 15 wherein a SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Tagg expressly or inherently discloses this claim element. <i>See, e.g.,</i> This reference discloses that a transmission uses Session Initiation Protocol (SIP). A transmission from a wireless communication device using SIP would have been well known to a person of ordinary skill in the art as of the priority date of this reference. This reference explicitly points to the RFC3261 as defining SIP and explaining that the signaling expressed in RFC3261 is utilized by this reference’s disclosure. “FIG. 24 illustrates the Network Address Processing required to process IP v4 RTP datagrams for the mobile roaming, handoff and handover. For this discussion, we will talk about RTP packets but a similar process is needed for the SIP and other packets. The RTP datagram 2400 is constructed in the normal way with IP, UDP, RTP and Data headers, hereafter referred to as an RTP packet. As can be seen each RTP packet contains source and destination information for both IP address and port-together known as a socket. The phone sends RTP packets to one or more sockets during handoff, handover and other events and these packets may travel over WWAN 2402 or WLAN 2404 infrastructure prior to reaching the Internet and our server. RTP packets 2404 and 2406 may arrive from two different sockets and these need to be processes and de-duplicated into one stream prior to traveling on to the media gateway 2408. An important factor is that both the WLAN and WWAN paths will need authentication to allow streams to enter the SIP server and in

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		addition the user name password SIP address should be authenticated against the SIM address in the phone.” Tagg at [0386]. (emphasis added) RFC3261, which defines SIP, explains authentication procedures that occur whenever a request to a SIP proxy server occurs. [RFC3261 at 22.3 – Proxy-to-User Authentication] Similarly, when a UAC sends a request to a proxy server, the proxy server MAY authenticate the originator before the request is processed. If no credentials (in the Proxy-Authorization header field) are provided in the request, the proxy can challenge the originator to provide credentials by rejecting the request with a 407 (Proxy Authentication Required) status code. The proxy MUST populate the 407 (Proxy Authentication Required) message with a Proxy- Authenticate header field value applicable to the proxy for the requested resource. The use of Proxy-Authenticate and Proxy-Authorization parallel that described in [17], with one difference. Proxies MUST NOT add values to the Proxy-Authorization header field. All 407 (Proxy Authentication Required) responses MUST be forwarded upstream toward the UAC following the procedures for any other response. It is the UAC’s responsibility to add the Proxy-Authorization header field value containing credentials for the realm of the proxy that has asked for authentication. If a proxy were to resubmit a request adding a Proxy-Authorization header field value, it would need to increment the CSeq in the new request. However, this would cause the UAC that submitted the original request to discard a response from the UAS, as the CSeq value would be different. When the originating UAC receives the 407 (Proxy Authentication Required) it SHOULD, if it is able, re-originate the request with the proper credentials. It should follow the same procedures for the display of the "realm" parameter that are given above for responding to 401. If no credentials for a realm can be located, UACs MAY attempt to retry the request with a username of "anonymous" and no password (a password of ""). The UAC SHOULD also cache the credentials used in the re-originated request. The following rule is RECOMMENDED for proxy credential caching: If a UA receives a Proxy-Authenticate header field value in a 401/407 response to a request with a particular Call-ID, it should incorporate credentials for that realm in all subsequent requests that contain the same Call-ID. These credentials MUST NOT be cached across dialogs; however, if a UA is configured with the realm of its local outbound proxy, when one exists, then the UA MAY cache credentials for that realm across dialogs. Note that this does mean a future request in a dialog could contain credentials that are not needed by any proxy along the Route header path. Any UA that wishes to authenticate itself

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		to a proxy server -- usually, but not necessarily, after receiving a 407 (Proxy Authentication Required) response -- MAY do so by including a Proxy-Authorization header field value with the request. The Proxy-Authorization request-header field allows the client to identify itself (or its user) to a proxy that requires authentication. The Proxy-Authorization header field value consists of credentials containing the authentication information of the UA for the proxy and/or realm of the resource being requested. A Proxy-Authorization header field value applies only to the proxy whose realm is identified in the "realm" parameter (this proxy may previously have demanded authentication using the Proxy-Authenticate field). When multiple proxies are used in a chain, a Proxy-Authorization header field value MUST NOT be consumed by any proxy whose realm does not match the "realm" parameter specified in that value. Note that if an authentication scheme that does not support realms is used in the Proxy-Authorization header field, a proxy server MUST attempt to parse all Proxy-Authorization header field values to determine whether one of them has what the proxy server considers to be valid credentials. Because this is potentially very time-consuming in large networks, proxy servers SHOULD use an authentication scheme that supports realms in the Proxy-Authorization header field. If a request is forked (as described in Section 16.7), various proxy servers and/or UAs may wish to challenge the UAC. In this case, the forking proxy server is responsible for aggregating these challenges into a single response. Each WWW-Authenticate and Proxy-Authenticate value received in responses to the forked request MUST be placed into the single response that is sent by the forking proxy to the UA; the ordering of these header field values is not significant. When a proxy server issues a challenge in response to a request, it will not proxy the request until the UAC has retried the request with valid credentials. A forking proxy may forward a request simultaneously to multiple proxy servers that require authentication, each of which in turn will not forward the request until the originating UAC has authenticated itself in their respective realm. If the UAC does not provide credentials for each challenge, the proxy servers that issued the challenges will not forward requests to the UA where the destination user might be located, and therefore, the virtues of forking are largely lost. When resubmitting its request in response to a 401 (Unauthorized) or 407 (Proxy Authentication Required) that contains multiple challenges, a UAC MAY include an Authorization value for each WWW-Authenticate value and a Proxy-Authorization value for each Proxy-Authenticate value for which the UAC wishes to supply a credential. As noted above, multiple credentials in a request SHOULD be differentiated by the "realm" parameter. It is possible for multiple challenges associated with the same realm to appear in the same 401 (Unauthorized) or 407 (Proxy Authentication Required). This can occur, for example, when multiple proxies within the same administrative domain, which use a common realm, are reached by a forking request. When it

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
		retries a request, a UAC MAY therefore supply multiple credentials in Authorization or Proxy-Authorization header fields with the same "realm" parameter value. The same credentials SHOULD be used for the same realm. One of ordinary skill would find this limitation disclosed either expressly or inherently in the teachings of this reference and its incorporated disclosures taken as a whole, or in combination with the state of the art at the time of the alleged invention. To the extent this reference is not found to teach this element explicitly, implicitly, or inherently, the element would have been obvious to one of ordinary skill in the art based on this reference, common sense, ordinary creativity of one of ordinary skill in the art, and the state of the art. Additionally, it would have been obvious to combine this reference with one or more other prior art references identified in Defendants’ Invalidity Contentions Cover Pleading, particularly, § V. Rather than repeat those disclosures here, they are incorporated by reference into this chart.
20	The method of claim 15 wherein the source device is a mobile device.	Tagg expressly or inherently discloses this claim element. <i>See above re claim 6 which is incorporated by reference herein.</i>
21	The method of claim 20 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Tagg expressly or inherently discloses this claim element. <i>See above re claim 7 which is incorporated by reference herein.</i>
22[Pre]	A communications network comprising	To the extent the preamble is limiting, Tagg expressly or inherently discloses this claim element. <i>See above re claim 1[Pre] which is incorporated by reference herein.</i>
22[a]	a transmitting device; a receiving device; a first network device; a second network device;	Tagg expressly or inherently discloses this claim element. <i>See above re claim 1[a] which is incorporated by reference herein.</i>
22[b]	the transmitting device and receiving device operable to maintain a data transmission over a first network path that is made up of one or more network links where at least one of the network	Tagg expressly or inherently discloses this claim element. <i>See above re claim 1[a] which is incorporated by reference herein.</i>

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
	links is to a local area network is remote from the transmitting device;	
22[c]	wherein the first network device is operable to switch the transmission from the first network path to one of a plurality of alternative network paths without disrupting the transmission; and said the second network device is operable to perform a Port Address Translation (PAT) on said transmission;	Tagg expressly or inherently discloses this claim element. See above re claim 1[b] and 1[c] which are incorporated by reference herein.
22[d]	said transmission contains audio or video content	Tagg expressly or inherently discloses this claim element. See above re claim 1[c] which is incorporated by reference herein.
22[e]	said transmission is made using the Real Time Protocol (RTP); said transmission further uses the Session Initiation Protocol (SIP); and	Tagg expressly or inherently discloses this claim element. See above re claim 1[d] which is incorporated by reference herein.
22[f]	said transmission's SIP signaling sequence is maintained after said switching from the first network path to the one of a plurality of alternative network paths.	Tagg expressly or inherently discloses this claim element. See above re claim 1[e] which is incorporated by reference herein.
23	The system of claim 22 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of a stream quality event.	Tagg expressly or inherently discloses this claim element. See above re claim 2 which is incorporated by reference herein.
24	The system of claim 22 wherein said switching from the first	Tagg expressly or inherently discloses this claim element.

Claim No.	'554 Patent	U.S. 2005/0286466 – (“Tagg”)
	network path to the one of a plurality of alternative network paths occurs as a result of stream load balancing.	See above re claim 17 which is incorporated by reference herein.
25	The system of claim 22 wherein said switching from the first network path to the one of a plurality of alternative network paths occurs as a result of an administrator move.	Tagg expressly or inherently discloses this claim element. See above re claim 18 which is incorporated by reference herein.
26	The system of claim 22 wherein a SIP server associated with the one of a plurality alternative network paths may authenticate the transmission before it is switched.	Tagg expressly or inherently discloses this claim element. See above re claim 19 which is incorporated by reference herein.
27	The system of claim 22 wherein the transmitting device is a mobile device.	Tagg expressly or inherently discloses this claim element. See above re claim 6 which is incorporated by reference herein.
28	The system of claim 27 wherein one or more of said plurality of alternative network paths become available to the mobile device during roaming.	Tagg expressly or inherently discloses this claim element. See above re claim 7 which is incorporated by reference herein.