IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

SAMSUNG ELECTRONICS AMERICA, INC.,

SAMSUNG ELECTRONICS CO., LTD.,

Petitioners,

v.

KONINKLIJKE KPN N.V.,

Patent Owner.

Case No. IPR2025-00504

U.S. Patent No. 8,549,151

DECLARATION OF KEVIN C. ALMEROTH, PH.D., IN SUPPORT OF PETITION FOR *INTER PARTES* REVIEW OF U.S. PATENT NO. 8,549,151

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I. INTRODUCTION

1. I, Kevin C. Almeroth, Ph.D., submit this declaration to state my opinions on the matter described below.

2. I have been retained by Petitioners Samsung Electronics Co., Ltd. and Samsung Electronics America, Inc. ("Petitioners"), as an independent expert in this proceeding before the United States Patent and Trademark Office. Although I am being compensated at my usual and customary rate of $850 per hour, no part of my compensation depends on the outcome of this proceeding, and I have no other interest in this proceeding.

3. I understand this proceeding involves U.S. Patent No. 8,549,151 (the "'151 patent"), and I have been asked to consider the validity of certain claims of the '151 patent based on certain prior art references. I have also been asked to consider the state of the art and prior art available as of December 21, 2007. Based on the prior art discussed in this declaration, it is my opinion that claims 1-20 and 22-23 of the '151 patent are unpatentable for the reasons provided below.

II. QUALIFICATIONS AND BACKGROUND

4. I am currently a Professor Emeritus in the Department of Computer Science at the University of California, Santa Barbara (UCSB). While at UCSB, I held faculty appointments and was a founding member of the Computer Engineering (CE) Program, Media Arts and Technology (MAT) Program, and the Technology

Management Program (TMP). I also served as the Associate Director of the Center for Information Technology and Society (CITS) from 1999 to 2012. I have been a faculty member at UCSB since July 1997.

5. I hold three degrees from the Georgia Institute of Technology: (1) a Bachelor of Science degree in Information and Computer Science (with minors in Economics, Technical Communication, and American Literature) earned in June 1992; (2) a Master of Science degree in Computer Science (with specialization in Networking and Systems) earned in June 1994; and (3) a Doctor of Philosophy (Ph.D.) degree in Computer Science (Dissertation Title: Networking and System Support for the Efficient, Scalable Delivery of Services in Interactive Multimedia System, minor in Telecommunications Public Policy) earned in June 1997. During my education, I have taken a wide variety of courses, as demonstrated by my minor. My undergraduate degree also included a number of courses more typical of a degree in electrical engineering including digital logic, signal processing, and telecommunications theory.

6. One of the major concentrations of my research over the past 30+ years has been the delivery of multimedia content and data between computing devices, including various network architectures. In my research, I have studied large-scale content delivery systems, and the use of servers located in a variety of geographic locations to provide scalable delivery to hundreds or thousands of users

simultaneously. I have also studied smaller-scale content delivery systems in which content is exchanged between individual computers and portable devices. My work has emphasized the exchange of content more efficiently across computer networks, including the scalable delivery of content to many users, mobile computing, satellite networking, delivering content to mobile devices, and network support for data delivery in wireless networks.

7. In 1992, the initial focus of my research was on the provision of interactive functions (e.g., VCR-style functions like pause, rewind, and fastforward) for near video-on-demand systems in cable systems; in particular, how to aggregate requests for movies at a cable head-end and then how to satisfy a multitude of requests using one audio/video stream broadcast to multiple receivers simultaneously. This research has continually evolved and resulted in the development of techniques to scalably deliver on-demand content, including audio, video, web documents, and other types of data, through the Internet and over other types of networks, including over cable systems, broadband telephone lines, and satellite links. 6. An important component of my research has been investigating the challenges of communicating multimedia content, including video, between computers and across networks including the Internet. Although the early Internet was used mostly for text-based, non-real time applications, the interest in sharing multimedia content, such as video, quickly developed. Multimedia-based

applications ranged from downloading content to a device to streaming multimedia content to be instantly used. One of the challenges was that multimedia content is typically larger than text-only content, but there are also opportunities to use different delivery techniques since multimedia content is more resilient to errors. I have worked on a variety of research problems and used a number of systems that were developed to deliver multimedia content to users. One content-delivery method I have researched is the one-to-many communication facility called "multicast," first deployed as the Multicast Backbone, a virtual overlay network supporting one-to-many communication. Multicast is one technique that can be used on the Internet to provide streaming media support for complex applications like video-on-demand, distance learning, distributed collaboration, distributed games, and large-scale wireless communication. The delivery of media through multicast often involves using Internet infrastructure, devices and protocols, including protocols for routing and TCP/IP.

8. Starting in 1997, I worked on a project to integrate the streaming media capabilities of the Internet together with the interactivity of the web. I developed a project called the Interactive Multimedia Jukebox (IMJ). Users would visit a web page and select content to view. The content would then be scheduled on one of a number of channels, including delivery to students in Georgia Tech dorms delivered

via the campus cable plant. The content of each channel was delivered using multicast communication.

9. In the IMJ, the number of channels varied depending on the capabilities of the server including the available bandwidth of its connection to the Internet. If one of the channels was idle, the requesting user would be able to watch their selection immediately. If all channels were streaming previously selected content, the user's selection would be queued on the channel with the shortest wait time. In the meantime, the user would *See* what content was currently playing on other channels, and because of the use of multicast, would be able to join one of the existing channels and watch the content at the point it was currently being transmitted.

10. The IMJ service combined the interactivity of the web with the streaming capabilities of the Internet to create a jukebox-like service. It supported true Video-on-Demand when capacity allowed, but scaled to any number of users based on queuing requested programs. As part of the project, we obtained permission from Turner Broadcasting to transmit cartoons and other short-subject content. We also connected the IMJ into the Georgia Tech campus cable television network so that students in their dorms could use the web to request content and then view that content on one of the campus's public access channels.

11. More recently, I have also studied issues concerning how users choose content, especially when considering the price of that content. My research has examined how dynamic content pricing can be used to control system load. By raising prices when systems start to become overloaded (i.e., when all available resources are fully utilized) and reducing prices when system capacity is readily available, users' capacity to pay as well as their willingness can be used as factors in stabilizing the response time of a system. This capability is particularly useful in systems where content is downloaded or streamed on-demand to users.

12. As a parallel research theme, starting in 1997, I began researching issues related to wireless devices and sensors. In particular, I was interested in showing how to provide greater communication capability to "lightweight devices," i.e., small form-factor, resource-constrained (e.g., CPU, memory, networking, and power) devices. Starting in 1998, I published several papers on my work to develop a flexible, lightweight, battery-aware network protocol stack. The lightweight protocols we envisioned were similar in nature to protocols like Bluetooth, Universal Plug and Play (UPnP) and Digital Living Network Alliance (DLNA). 12. From this initial work, I have made wireless networking—including ad hoc, mesh networks and wireless devices—one of the major themes of my research. My work in wireless network spans the protocol stack from applications through to the encoding and exchange of data at the data link and physical layers. 13. At the application layer,

even before the large-scale "app stores" were available, my research looked at building, installing, and using apps for a variety of purposes, from network monitoring to support for traditional computer-based applications (e.g., content retrieval) to new applications enabled by ubiquitous, mobile devices. For example, my research has looked at developing applications for virally exchanging and tracking "coupons" through "opportunistic contact" (i.e., communication with other devices coming into communication range with a user). In many of the courses I have taught there is a project component. Through these projects I have supervised numerous efforts to develop new "apps" for download and use across a variety of mobile platforms.

13. Yet another theme of this research was monitoring wireless networks, in particular different variants of IEEE 802.11 compliant networks, to (1) understand the operation of the various protocols used in real-world deployments, (2) use these measurements to characterize use of the networks and identify protocol limitations and weaknesses, and (3) propose and evaluate solutions to these problems. I have successfully used monitoring techniques to study wireless data link layer protocol operation and to improve performance by enhancing the operation of such protocols. For wireless protocols, this research includes functions like network acquisition and channel bonding.

14. My recent work ties some of the various threads of my past research together. I have investigated content delivery in online social networks and proposed reputation management systems in large-scale social networks and marketplaces. On the content delivery side, I have looked at issues of caching and cache placement, especially when content being shared and the cache has geographical relevance. We were able to show that effective caching strategies can greatly improve performance and reduce deployment costs. Our work on reputation systems showed that reputations have economic value, and as such, creates a motivation to manipulate reputations. In response, we developed a variety of solutions to protect the integrity of reputations in online social networks. The techniques we developed for content delivery and reputation management were particularly relevant in peer-to-peer communication and recommendations for downloadable "apps."

15. As an important component of my research program, I have been involved in the development of academic research into available technology in the market place. One aspect of this work is my involvement in the Internet Engineering Task Force (IETF). The IETF is a large and open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet. I have been involved in various IETF groups including many content delivery-related working groups like the Audio Video Transport (AVT) group, the MBone Deployment

(MBONED) group, Source Specific Multicast (SSM) group, the Inter-Domain Multicast Routing (IDMR) group, the Reliable Multicast Transport (RMT) group, the Protocol Independent Multicast (PIM) group, etc. I have also served as a member of the Multicast Directorate (MADDOGS), which oversaw the standardization of all things related to multicast in the IETF. Finally, I was the Chair of the Internet2 Multicast Working Group for seven years.

16. My involvement in the research community extends to leadership positions for several academic journals and conferences. I am the co-chair of the Steering Committee for the ACM Network and System Support for Digital Audio and Video (NOSSDAV) workshop and on the Steering Committees for the International Conference on Network Protocols (ICNP), ACM Sigcomm Workshop on Challenged Networks (CHANTS), and IEEE Global Internet (GI) Symposium. I have served or am serving on the Editorial Boards of IEEE/ACM Transactions on Networking, IEEE Transactions on Mobile Computing, IEEE Network, ACM Computers in Entertainment, AACE Journal of Interactive Learning Research (JILR), and ACM Computer Communications Review. I have co-chaired a number of conferences and workshops including the IEEE International Conference on Network Protocols (ICNP), IEEE Conference on Sensor, Mesh and Ad Hoc Communications and Networks (SECON), International Conference on Communication Systems and Networks (COMSNETS), IFIP/IEEE International

Conference on Management of Multimedia Networks and Services (MMNS), the International Workshop On Wireless Network Measurement (WiNMee), ACM Sigcomm Workshop on Challenged Networks (CHANTS), the Network Group Communication (NGC) workshop, and the Global Internet Symposium, and I have served on the program committees for numerous conferences.

17. Furthermore, in the courses I taught at UCSB, a significant portion of my curriculum covered aspects of the Internet and network communication including the physical and data link layers of the Open System Interconnect (OSI) protocol stack, and standardized protocols for communicating across a variety of physical media such as cable systems, telephone lines, wireless, and high-speed Local Area Networks (LANs). The courses I have taught also cover most major topics in Internet communication, including data communication, multimedia encoding, and mobile application design. My research and courses have covered a range of physical infrastructures for delivering content over networks, including cable, Integrated Services Digital Network (ISDN), Ethernet, Asynchronous Transfer Mode (ATM), fiber, and Digital Subscriber Line (DSL). For a complete list of courses I have taught, *See* my curriculum vitae (CV), attached as Exhibit 2006.

18. In addition, I co-founded a technology company called Santa Barbara Labs that was working under a sub-contract from the U.S. Air Force to develop very accurate emulation systems for the military's next generation internetwork. Santa

Barbara Labs' focus was in developing an emulation platform to test the performance characteristics of the network architecture in the variety of environments in which it was expected to operate, and, in particular, for network services including IPv6, multicast, Quality of Service (QoS), satellite-based communication, and security. Applications for this emulation program included communication of a variety of multimedia-based services, including video conferencing and video-on-demand.

19. In addition to having co-founded a technology company myself, I have worked for, consulted with, and collaborated with companies for nearly 30 years. These companies range from well-established companies to start-ups and include IBM, Hitachi Telecom, Turner Broadcasting System (TBS), Bell South, Digital Fountain, RealNetworks, Intel Research, Cisco Systems, and Lockheed Martin.

20. I am a Member of the Association of Computing Machinery (ACM) and a Fellow of the Institute of Electrical and Electronics Engineers (IEEE).

21. Additional details about my employment history, fields of expertise, courses taught, and publications are further included in my CV attached as Exhibit 1010.

III. LEGAL STANDARDS

22. In forming my opinions and considering the subject matter of the '151 patent and its claims in light of the prior art, I am relying on certain legal principles

that counsel in this case explained to me. My understanding of these concepts is summarized below.

23. I understand that the claims define the invention. I also understand that an unpatentability analysis is a two-step process. First, the claims of the patent are construed to determine their meaning and scope. Second, after the claims are construed, the content of the prior art is compared to the construed claims.

24. I understand that a claimed invention is only patentable when it is new, useful, and non-obvious in light of the "prior art." That is, the invention, as defined by the claims of the patent, must not be anticipated or rendered obvious by the prior art.

A. Claim Construction

25. I understand that the United States Patent and Trademark Office interprets claim terms in an *inter-partes* review proceeding under the same claim construction standard that is used in a United States federal court. I understand that under this standard, the meaning of claim terms is considered from the viewpoint of one of ordinary skill in the art at the time of the alleged invention.

26. I understand that claim terms are generally given their ordinary and customary meaning as understood by one of ordinary skill in the art in light of the specification and the prosecution history pertaining to the patent. I understand, however, that claim terms are generally not limited by the embodiments described

in the specification.

27. I understand that in addition to the claims, specification, and prosecution history, other evidence may be considered to ascertain the meaning of claim terms, including textbooks, encyclopedias, articles, and dictionaries. I have been informed that this other evidence is often less significant and less reliable than the claims, specification, and prosecution history.

B. Anticipation Under 35 U.S.C. § 102

28. I understand that under 35 U.S.C. § 102, a patent claim is invalid if its subject matter was patented or described in a printed publication before the effective filing date of the claimed invention. I have been told that this is referred to as invalidity by anticipation. I have been told that a patent claim is anticipated under § 102 if a single prior art reference discloses all limitations of the claimed invention.

C. Obviousness Under 35 U.S.C. § 103

29. I understand that a patent claim is invalid as obvious if the claimed invention would have been obvious to a person of ordinary skill in the art ("POSA") at the time the claimed invention was made. This means that even if all of the elements of the claim cannot be found in a single prior art reference that would anticipate the claim, a person of ordinary skill in the field who knew about all the prior art would have come up with the claimed invention. I understand that in an obviousness determination, the person of ordinary skill in the art is presumed to have

knowledge of all material prior art. I understand that whether a claim is obvious is based upon the determination of several factual issues.

30. I understand that obviousness is a determination of law based on underlying determinations of fact. I understand that these factual determinations include the scope and content of the prior art, the level of ordinary skill in the art, the differences between the claimed invention and the prior art, and secondary considerations of non-obviousness.

31. In considering obviousness, I understand that one must determine the scope and content of the prior art. I understand that, in order to be considered as prior art to a patent being considered, a prior art reference must be reasonably related to the claimed invention of that patent. A reference is reasonably related if it is in the same field as the claimed invention or is from another field to which a person of ordinary skill in the art would look to solve a known problem.

32. I understand that one must determine what differences, if any, existed between the claimed invention and the prior art.

33. I understand that a patent claim composed of several elements is not proved obvious merely by demonstrating that each of its elements was independently known in the prior art. In evaluating whether such a claim would have been obvious, one may consider whether a reason has been identified that would have prompted a person of ordinary skill in the art to combine the elements or concepts from the prior

art in the same way as in the claimed invention. There is no single way to define the line between true inventiveness on the one hand (which is patentable) and the application of common sense and ordinary skill to solve a problem on the other hand (which is not patentable). For example, market forces or other design incentives may be what precipitated a change, rather than true inventiveness.

34. I understand that whether a prior art reference renders a patent claim unpatentable as obvious is determined from the perspective of a person of ordinary skill in the art at the time of the alleged invention. I have been told that there is no requirement that the prior art contain an express suggestion to combine known elements to achieve the claimed invention, but a suggestion to combine known elements to achieve the claimed invention may come from the prior art, as filtered through the knowledge of one skilled in the art. In addition, I have been told that the inferences and creative steps a person of ordinary skill in the art would employ are also relevant to the determination of obviousness.

35. I understand that there is no rigid rule that a reference or combination of references must contain a "teaching, suggestion, or motivation" to combine references. But I also understand that the "teaching, suggestion, or motivation" test can be a useful guide in establishing a rationale for combining elements of the prior art. I have been told that this test poses the question as to whether there is an express or implied teaching, suggestion, or motivation to combine prior art elements in a

way that realizes the claimed invention, and that it seeks to counter impermissible hindsight analysis.

36. I understand that one may consider, e.g., whether (1) the change was merely the predictable result of using prior art elements according to their known functions, or whether it was the result of true inventiveness; (2) there is some teaching or suggestion in the prior art to make the modification or combination of elements claimed in the patent; (3) the claimed innovation applies a known technique that had been used to improve a similar device or method in a similar way; (4) the claimed invention would have been obvious to try, meaning that the claimed innovation was one of a relatively small number of possible approaches to the problem with a reasonable expectation of success by those skilled in the art; (5) the invention merely substituted one known element for another known element in order to obtain predictable results; (6) the invention merely applies a known technique to a known device, method, or product to yield predictable results; or (7) known work in the field may have prompted variations of use of the same inventions in the same or different fields due to market forces or design incentives that would have been predictable to a person of ordinary skill in the art.

37. I understand that any assertion of secondary considerations of non-obviousness must be accompanied by a nexus between the merits of the invention and the evidence offered.

IV. PERSON OF ORDINARY SKILL IN THE ART (POSA)

38. I am informed that patentability must be analyzed from the perspective of "one of ordinary skill in the art" in the same field as the '151 patent at the time of the invention. As previously discussed, the relevant time of invention is the patent's priority date, which is no earlier than December 21, 2007. Moreover, I understand that whether or not a patent application provides "written description" support for patent claims must also be assessed from the perspective of a person of ordinary skill in the art.

39. I am also informed that several factors are considered in assessing the level of ordinary skill in the art, including (1) the types of problems encountered in the art; (2) the prior art solutions to those problems; (3) the rapidity with which innovations are made; (4) the sophistication of the technology; and (5) the education level of active workers in the field.

40. A person of ordinary skill in the art pertinent to the '151 patent would have had at least a Bachelor of Science degree in electrical engineering, computer engineering, computer science, or an equivalent field, and at least two years of experience with cellular telecommunications systems and related protocols, and multimedia systems in wireless networks. Additional education could substitute for professional experience and vice versa. A person of ordinary skill in the art would also be able to understand and apply the prior art discussed herein.

41. Although I surpass this definition of one of ordinary skill in the art now and at the priority date of the ’151 patent, my analysis regarding the ’151 patent has been based on the perspective of one ordinary skill in the art as of the priority date of the ’151 patent.

42. I am also familiar with the knowledge of the person of ordinary skill in the art as of the priority date of the ’151 patent. I am able to opine on how the person of ordinary skill in the art would have understood the disclosure and claims of the ’151 patent, the disclosures of the prior art, the motivation to combine the prior art, and what combinations would have been obvious to one of ordinary skill in the art.

V. TECHNICAL BACKGROUND

43. I first begin with a summary of the technology involved with the ’151 patent.

A. Introduction to the ’151 Technology

1. Prior Standard Technology

44. In my opinion, a POSA at the time would recognize and understand the standard protocols used in cellular standards, including their purpose, structure, and interaction within the broader communication system. This includes familiarity with widely adopted protocols such as SIP for signaling, RTP for media transport, and RTSP for streaming control, as well as an understanding of how these protocols work together to enable seamless real-time communication and multimedia delivery over

cellular networks. Indeed, anyone at the time with a level of familiarity with standardized communication technologies would at least recognize SIP, RTP, and RTSP and understand that their use and capabilities are both well-documented, along with their use being wide-spread.

2. History and Overview of the Session Invitation Protocol (SIP)

45. The development of the Session Invitation Protocol (SIP) was formalized with the publication of RFC 2543 (Ex.1014) in 1999 by the Internet Engineering Task Force (IETF). RFC 2543 established SIP as a signaling protocol designed to create, modify, and terminate multimedia sessions, leveraging a simple, text-based structure modeled after HTTP. *See* Ex.1014, 1-2. It introduced a transaction-oriented framework that, *inter alia*, used INVITE, ACK, and 200 OK messages for session setup. *Id.*, 28-29, 76. Within this framework, SIP relied on the Session Description Protocol (SDP) for media session negotiation, enabling endpoints to communicate parameters such as codecs, media types, and transport information. *Id.*, 136-144. Gateva (Ex.1005) and Whitehead (Ex.1006) (described in more detail *infra*) both provide extensive disclosure of the different SIP messages and their SDP payloads. *See, e.g.*, Ex.1004, Ex.1005.

46. In 2002, the IETF published RFC 3261 (Ex.1016) in 2002, replacing RFC 2543 and introducing advancements on the original specification. Ex.1016, 1. Most notably, RFC 3261 adopted the offer/answer model, for the structured

negotiation of media session parameters using SDP. *Id.*, 9, 15. This model provided clear rules for exchanging session descriptions: (1) an SDP is usually included in the INVITE request to describe the initiating party's desired session parameters; (2) the receiving party would respond with a 200 OK containing an SDP answer that finalized the negotiated parameters; (3) and, finally an ACK message is used to confirm acceptance. Ex.1006, 21-25 (providing discussion on standard SIP messaging schemes).

3. History and Overview of the Real-Time Streaming Protocol (RTSP)

47. The formalization of the Real-Time Streaming Protocol (RTSP) occurred with the publication of RFC 2326 (Ex.1008) in 1998 by the Internet Engineering Task Force (IETF). *See, e.g.,* Ex.1008, 1. RTSP was designed to facilitate the control of real-time multimedia streams, offering functionality such as play, pause, seek, and stop for streaming media. *See, e.g.,* Ex.1008, 12-13. Drawing inspiration from HTTP, RTSP employed a similar text-based structure but diverged by introducing stateful interactions and mechanisms optimized for continuous media delivery. *Id*. It supported both live and on-demand streaming use cases, providing a foundation for many early streaming technologies. *Id.*., 63, 69.

48. RFC 2326 introduced features that enabled session control over real-time media. *See, e.g.,* Ex.1008. Using methods such as SETUP, PLAY, PAUSE, and TEARDOWN, clients could establish and manage streaming sessions with

servers. *See, e.g.*, Ex.1008, 12-13. Media transport parameters were negotiated during the SETUP phase, allowing for compatibility with protocols like RTP (Real-time Transport Protocol) and ensuring efficient data delivery. *Id.*, 1, 5, 33-34. RTSP also relied on external description formats, such as the Session Description Protocol (SDP), to describe media characteristics, including codec information, bitrates, and media access points. *Id.*, 8; *see also* Ex.1015, [6] (SDP standard). In my experience, these descriptions were often included within RTSP messages to provide clients and servers with the necessary details to coordinate streaming. *Id.*

4. History and Overview of Real-Time Transport Protocol (RTP)

49. The Real-time Transport Protocol (RTP) was formalized with the publication of RFC 1889 in 1996, establishing a protocol for transporting real-time audio, video, and other time-sensitive data over packet-switched networks. Ex.1017, 1.3-5. In other words, RTP was used for providing media streams. RTP provided essential mechanisms for media synchronization and stream identification using sequence numbers, timestamps, and payload type identifiers. *Id.*, 3-5. Accompanying RTP in the RFC was the Real-time Transport Control Protocol (RTCP), which allowed participants to monitor quality of service (QoS) and provided session control feedback, including reports on packet loss and jitter. *Id.*

50. In 2003, RFC 3550 replaced RFC 1889, introducing RTP version 2. *See, e.g.*, Ex.1018. This update clarified ambiguities in the protocol, standardizing

the use of sequence numbers, timestamps, and other key fields to ensure consistent implementation, however "[m]ost of the text in [RFC3550] is identical to RFC 1889 which it obsoletes." *Id.*, 1, 3-5.

5. Integration of SIP and RTSP

51. The integration of SIP and RTSP was an obvious and widely known solution for addressing interoperability between session initiation and media control protocols, particularly as multimedia applications became increasingly sophisticated. Such systems were known as of the priority date of the '151 patent. Ex.1001, 1:31-32 ("Systems that provide an interface between SIP and RTSP domains . . . are known in the art."). Combining SIP's robust signaling capabilities with RTSP's media control functionality allowed developers to create systems that supported features like dynamic playback, adaptive streaming, and interactive user controls, meeting the demands of increasingly complex multimedia services.

52. At the time SIP was widely adopted as a signaling protocol and facilitated the negotiation of key session parameters—such as codec selection, transport protocols, and network resources—through its use of the SDP for setting up media session. Ex.1001, 7:32-36. While later obsoleted by RFC3261, the IETF standard document, RFC2543, provides an exemplary use of INVITE, ACK, and 200OK messages that is used in RFC3261, the typical session set-up using SIP "consists of two requests, an INVITE followed by ACK." Ex.1014 at 15; Both the

INVITE and ACK messages contain a "session description" (SDP) for, *inter alia*, confirming the media type to be used in the session. Ex.1014, 28-29. In this process, the sender of the INVITE message proposes its media capabilities and then confirms with an ACK containing "the final session description to be used by the callee." Ex.1014, 28-29. In addition, a "200 OK" message is sent by the receiver of the INVITE message to the sender between the INVITE and ACK message to indicate to the sender that the "callee has agreed to participate" and provides its capabilities. Ex.1014, 76. RFC2543 lays out this messaging scheme in the context of an SIP proxy, the messages highlighted in red reflect the normal order, type, and direction of the SIP messages used in a typical session set-up:

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                                  +....... cs.columbia.edu .......+
                                  :                               :
                                  : (~~~~~~~~~~)                  :
                                  : ( location )                  :
                                  : ( service  )                  :
                                  : (~~~~~~~~~~)                  :
                                  :     ^    |                    :
                                  :     | hgs@lab                 :
                                  :    2|   3|                    :
                                  :     |    |                    :
                                  : henning  |                    :
+.. cs.tu-berlin.de ..+ 1: INVITE :     |    |                    :
:                     :   henning@cs.col:   |   \/ 4: INVITE  5: ring :
: cz@cs.tu-berlin.de  =========================>(~~~~~~)=========>(~~~~~~) :
:                     <........................(      )<.........(      ) :
:                     : 7: 200 OK          :   (      )6: 200 OK (      ) :
:                     :                    :   ( work )          ( lab  ) :
:                     : 8: ACK             :   (      )9: ACK    (      ) :
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+.....................+                    +...............................+

 ====> SIP request
 ....> SIP response

  ^
  |     non-SIP protocols
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53. SDP played a vital role in enabling SIP clients to effectively share and confirm session details, such as media types and transport addresses, between devices initiating a multimedia session. Ex.1016, 9-10 (citing to RFC2327 describing SDP); Ex.1015 (RFC2327), 10-11, 19-20, 36-37. By including this information within the SDP portion of SIP messages, devices could negotiate compatible parameters to ensure seamless communication before establishing a session. (Ex.1014, 9, 147-157.) This negotiation process was essential to the functionality of SIP-based systems, as it provided a standardized framework for session setup.

54. SDP also supported the inclusion of Uniform Resource Locators (URLs), which specified the location of resources or endpoints within the network. Ex.1014, 20; Ex.1016, 36-37; Ex.1009, 3 (“The term "Uniform Resource Locator" (URL) refers to the subset of URI that identify resources via a representation of their primary access mechanism”).. These URLs, along with other SDP parameters, enabled devices to identify where media streams originated or were destined. This ensured proper routing and coordination of multimedia traffic, a critical aspect of reliable and efficient communication. URIs (Uniform Research Identifiers), such as URLs, and other SDP information allowed devices to identify where media streams originated or were destined, ensuring proper routing and coordination. Indeed, at the time it was well known that URLs are a type of URI.

55. RTSP was designed for managing and controlling media playback, provided commands such as SETUP, PLAY, and PAUSE, which allowed users to dynamically interact with media streams. Ex.1008, 12-13. These commands enabled granular control over media delivery, making RTSP highly effective for applications like Video on Demand (VoD) and live streaming. By offering direct mechanisms for media manipulation, RTSP allowed users to pause, resume, or reposition streams, providing flexibility that was essential for interactive media experiences.

56. Like SIP, RTSP also utilized SDP to negotiate session parameters, ensuring compatibility between the client and the media stream being controlled.

Ex.1008, 80-88. This shared reliance on SDP facilitated the exchange of critical information such as media types, codecs, and transport addresses, which were necessary to establish and manage sessions. However, despite this common foundation, SIP and RTSP were not natively interoperable. SIP was primarily focused on session signaling and setup, while RTSP was tailored to media stream control, creating a functional gap when both protocols were used together.

57. This lack of integration created alleged inefficiencies in systems where session setup via SIP needed to transition seamlessly into media control handled by RTSP. Without an intermediary, devices initiating sessions using SIP could not directly communicate playback or stream parameters to RTSP servers, limiting functionality and complicating workflows in applications like VoD and live streaming. For instance, the inability of SIP to pass necessary media control commands to RTSP servers often required additional infrastructure or bespoke solutions to bridge the protocols, adding complexity to the system.

58. The need for interoperability between SIP and RTSP became increasingly apparent as multimedia systems evolved and the IP Multimedia Subsystem (IMS) architecture emerged. IMS, standardized by the 3GPP, sought to establish a unified framework for delivering voice, video, and data services over IP networks. At the core of IMS was SIP, which served as the signaling backbone, enabling session initiation and management across a wide range of applications.

However, while SIP was well-suited for session signaling, IMS applications often required dynamic media stream control, a capability provided by RTSP. This highlighted the necessity for seamless integration between these two protocols to deliver comprehensive multimedia services.

59. To address this need, gateways were widely implemented to bridge SIP and RTSP. These gateways acted as translators, enabling the exchange of Uniform Resource Identifiers (URIs), SDP parameters, and other session details between the protocols. This integration allowed SIP clients to establish multimedia sessions that RTSP servers could manage with fine-grained playback controls. By combining SIP's robust signaling capabilities with RTSP's dynamic media management features, these gateways enabled a unified system architecture capable of handling both session setup and media interaction efficiently.

60. By leveraging the complementary strengths of SIP and RTSP, developers were able to design multimedia systems that delivered high-quality, interactive streaming experiences. SIP's signaling framework allowed applications to pre-negotiate network resources, ensuring that media streams could be delivered with adequate Quality of Service (QoS). Simultaneously, RTSP's ability to manage playback dynamically and respond to user interactions made it an ideal solution for applications like Video on Demand (VoD), live streaming, and educational platforms. This integration not only streamlined system design but also laid the

foundation for innovative multimedia solutions, enabling reliable, scalable, and highly interactive services that met the growing demands of users and service providers alike.

6. Other Relevant Literature

61. Below I discuss two references Whitehead and Avaya and their relevance to the petition.

62. Whitehead is an "An Evaluation of Session Initiation Protocol (SIP) for use in Streaming Media Applications" conducted by the IETF. This paper notes that it seeks "**to improve the integration of the Session Initiation Protocol (SIP) [RFC3261] to the Real Time Streaming Protocol (RTSP and RTSP v2) [RFC 2326 and IDRTSP]** especially in the context of converged media services. Indeed, Whitehead (a study conducted by a group that I consider, as well as any POSA would, to be a prominent and well-known group in the industry as it sets the communication standards) directly speaks to the fact that RTSP and SIP integration was well-known by POSAs and the leading solution for converged media services. Ex. 1006, 2. Whitehead further points out that "[h]istorically RTSP [RFC2326] has been the signaling protocol of choice for streaming media applications." *Id.*, 3. "While historically SIP [RFC3261] has been used for communication services, the protocol itself is flexible enough (by design) to signal a wide variety of media streams." *Id.*, 3, 8-9. The paper further describes the same purported inventive

solution brought by the ’151 patent whereby “a dual-protocol class of solution[]” is used wherein “SIP is used for session negotiation and another protocol (e.g., RTSP) is used for media stream control.” *Id.*, 3. Whitehead introduces several exemplary solutions (12-14), where “the switching between RTSP and the SIP is left to the application logic.” *Id.*, 12.

63. Avaya's whitepaper, “Enterprising with SIP—A Technology Overview,” (Ex.1006) highlights that SIP was well-known to POSAs at the time, as it was recognized by the industry as “a key technology for . . . converged communication architecture” at the time. *Id.*, 1. It also highlights the role of the Session Description Protocol (SDP) within the SIP messages for establishing multimedia sessions in enterprise communications. Ex.1006, 16-18. The paper emphasizes that SDP is a fundamental component of the SIP signaling process, as it provides the necessary media negotiation parameters, such content-type (which a POSA would understand to refer to the type of media, e.g., RTP, video, etc.). *Id.* This information allows endpoints to establish compatible media streams, ensuring seamless communication across diverse devices and networks. *Id.*, 15-16.

VI. OVERVIEW OF THE ’151 PATENT

A. Disclosure

64. The ’151 patent is directed to “a gateway” for “transmitting a multimedia stream from a first terminal and receiving an associated second

multimedia stream at a second terminal," and more specifically, transmitting between "SIP and RTSP domains" or clients using a "first," "second," and "third" protocol. Ex.1001, Abstract, 1:31-35, 5:10-14. As provided for in the dependent claims of the patent, each protocol can be the same protocol or different. *Id.*, cl. 7 ("third protocol is different from the second protocol"), 14 ("the first protocol and the second protocol are identical").

65. The '151 patent does not purport to invent a gateway capable of providing such an interface. Instead, the patent acknowledges that the prior art describes "[s]ystems that provide an interface between SIP and RTSP domains, usually referred to as SIP-RTSP gateways, ***are known in the art***." Ex.1001, 1:31-33.

66. The '151 patent purports to improve on these prior art systems by reducing the drawbacks of prior systems, *e.g.*, "if the transmitting-end . . . wants to initiate a multimedia streaming session, the receiving-end is not capable to respond to such an invitation." Ex.1001, 2:11-21. The '151 patent's purported solution is to have the gateway acts both as a SIP client and RTSP server and [] provide a trigger to the second terminal for connecting the second terminal to the gateway," as illustrated in Fig. 5(a):

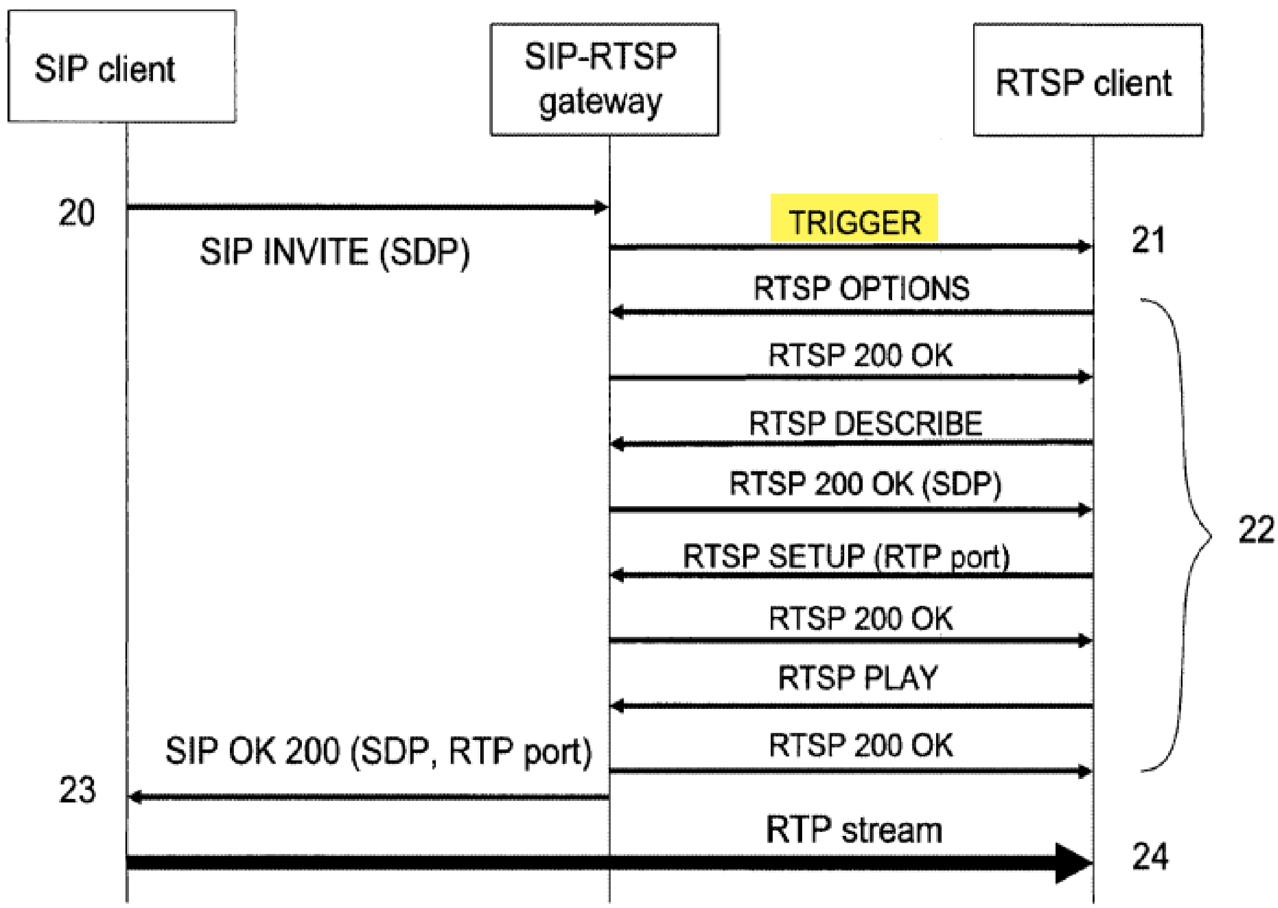


Figure 5 (a)

Ex.1001, Fig.5(a) (highlighting added).

B. Prosecution History

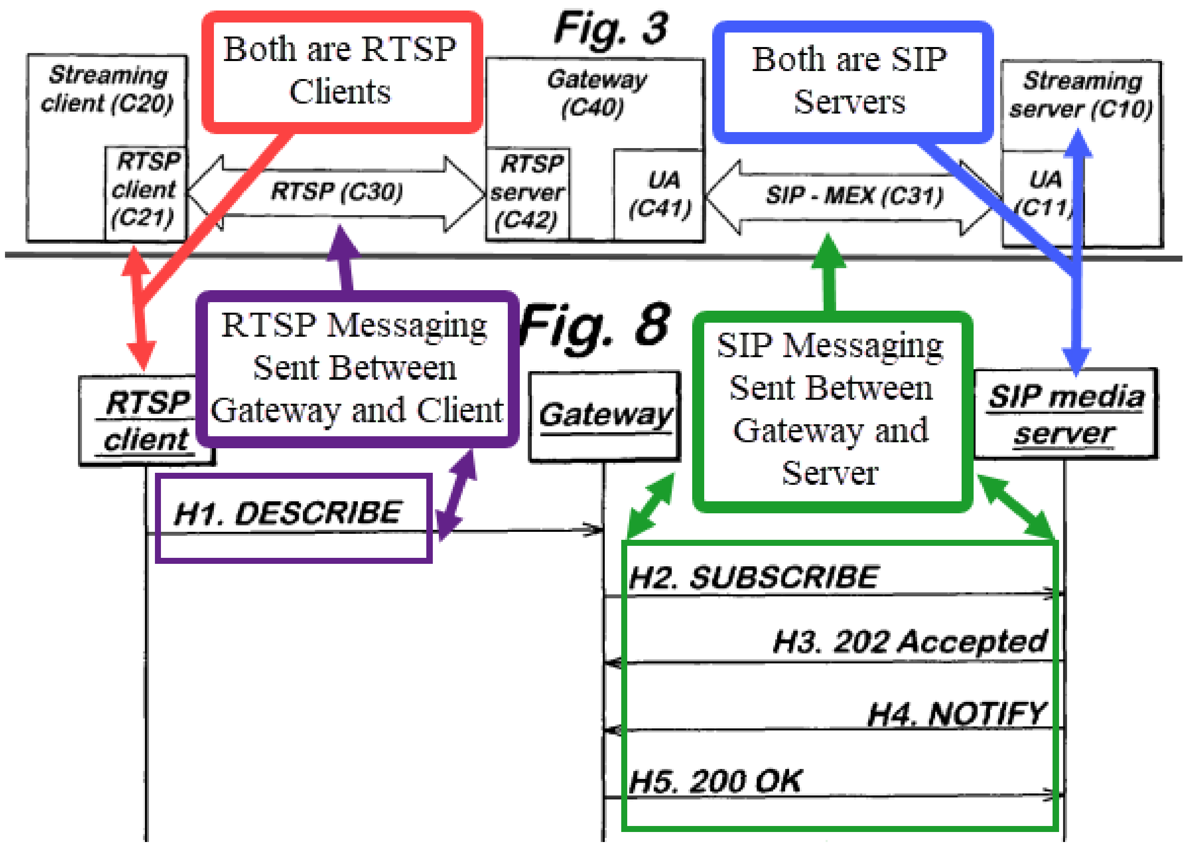
67. I have reviewed the prosecution history of the applications that led to the ’151 patent.

68. The application that led to the ’151 patent was filed on December 22, 2008, and claimed priority back to a provisional application filed on December 21, 2007. *See* generally Ex.1002. Selected portions of the prosecution are discussed below.

69. Specifically, in a Non-Final Rejection filed on October 16, 2012, the Examiner rejected the claims as unpatentable over Mela in view of RFC2326 (Ex.1008) and U.S. Patent No. 6,895,439 ("Isomaki") finding that Mela "teaches the invention substantially as claimed." Ex.1002, 62-104. Although, the Examiner acknowledged that Mela does not explicitly teach the step of initiating the exchange of the second media session information. The examiner further relied on Isomaki because "Mela and [RFC2326] does not explicitly teach using a third protocol." Ex.1002, 73. After considering arguments from the applicant, the Examiner called the applicant regarding an Examiner's amendment and applicant proposed an alternative amendment that the Examiner agreed would put the claims in a condition for allowance. Ex.1002, 10. The amendment added the following limitation to each independent claim: "wherein the first media stream is between the first terminal and a media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal." Ex.1002, 17-21. This amendment was unrelated to arguments made by applicant in Response to the October 16 Non-Final Rejection. Ex.1002, 62-64.

70. As the applicant pointed out, "Examiner relie[d] to a large extent on Figure 2 . . . , and for a minor part on the description of Fig. 8," in making its

rejections which was an error. Ex.1002, 44. In my opinion, Fig. 2 does not correspond to Fig. 8. Fig. 2 illustrates an ***RTSP Server*** and ***SIP Client*** while Fig. 8 depicts a ***RTSP Client*** and ***SIP Server*** creating mis-matches in the Examiner's invalidity read. Ex.1002, 11-12. It is unclear whether Examiner appreciated this error in allowance, but Petitioner's arguments below rely largely on the teachings of Fig. 8, and for a minor part on Fig. 3, which both illustrate the same embodiment wherein ***RTSP Clients*** and ***SIP Servers*** communicate through a gateway:



Ex.1004, Figs. 3, 8.

71. Further, Examiner only relied on Fig. 9 for the "H2—SUBSCRIBE" message which Examiner stated taught the claimed "trigger" message. Ex.1002, 69-

70. Petitioner's grounds do not rely on the "H2.SUBSCRIBE" message outside of acknowledging it as an unclaimed precursor step for practicing the limitations. Instead, Petitioner relies on the messages starting with "H4.NOTIFY" for its grounds which were not considered by the Examiner.

72. Lastly, the Examiner placed the claims in allowance over Mela based on adding an obvious limitation in its Examiner amendment. Ex.1002, 17-21. In my opinion the Examiner failed to consider that almost any system using the teachings of Mela would require buffering by a server to smooth out the data flow as otherwise the media stream would be jittery. Buffering was a well-established technique in multimedia systems to address issues caused by network variability, such as jitter and packet loss, which can disrupt the quality of real-time media delivery. Without sufficient buffering, fluctuations in data arrival times would result in uneven playback, making the media stream appear jittery and unreliable.

VII. OVERVIEW OF THE PRIOR ART

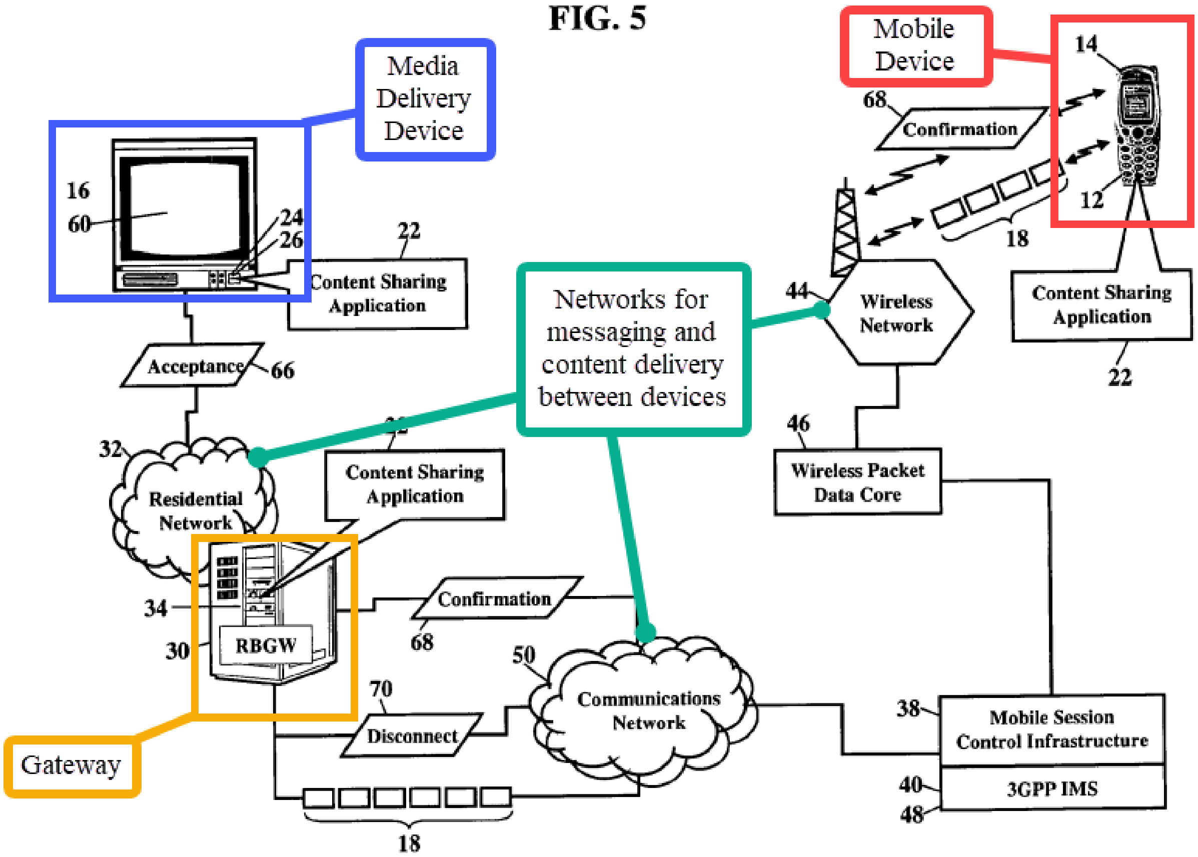
A. Alston (U.S. Patent Pub. No. 2006/0271687) – Ex.1004

73. I understand that Alston qualifies as prior art.

74. Alston focuses on a residential broadband gateway ("RBGW") capable of setting up sessions using known protocols that is able to trigger the receiving residential terminal and notify it of the availability of a new media stream from the sending terminal. *See generally* Ex.1004. As illustrated below, this gateway was

provided for in an 3GPP IMS system allowing cell phones to interact with at-home devices, *i.e.*, the standardized communication system for cellular and internet based communication at the time:

FIG. 5



Ex.1004, Fig. 5.

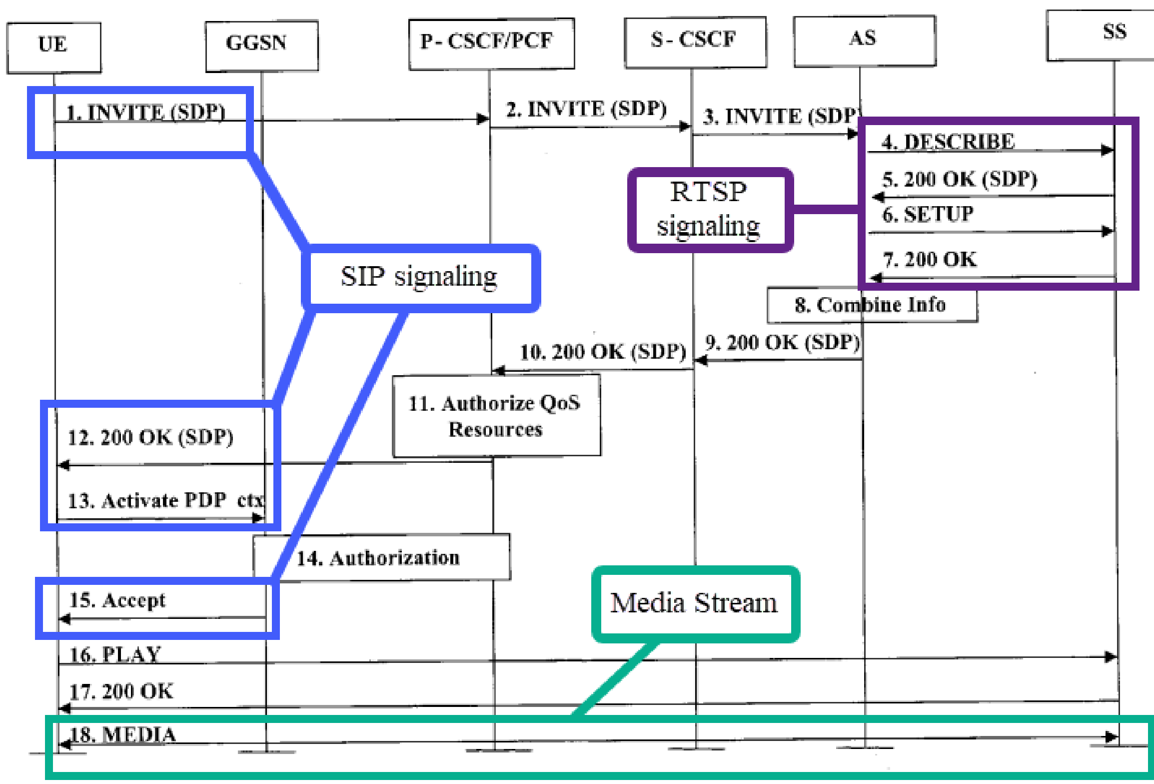
75. Alston's gateway uses conventional protocol messaging to exchange media session information between the sending terminal, gateway, and receiving terminal. Ex.1004, [0028]. Myself and any other POSA would recognize this as the conventional process under the standards at the time for setting up a media stream—sharing and confirming the media session parameters between terminals using SDP. Ex.1004, [0028]. The gateway further can store the media stream locally, so that the

content can be retrieved later through a local media stream instead of via the IMS system. Ex.1004, [0046].

B. Gateva (U.S. Patent Pub. No. 2004/0184432) – Ex.1005

76. I understand that Gateva qualifies as prior art.

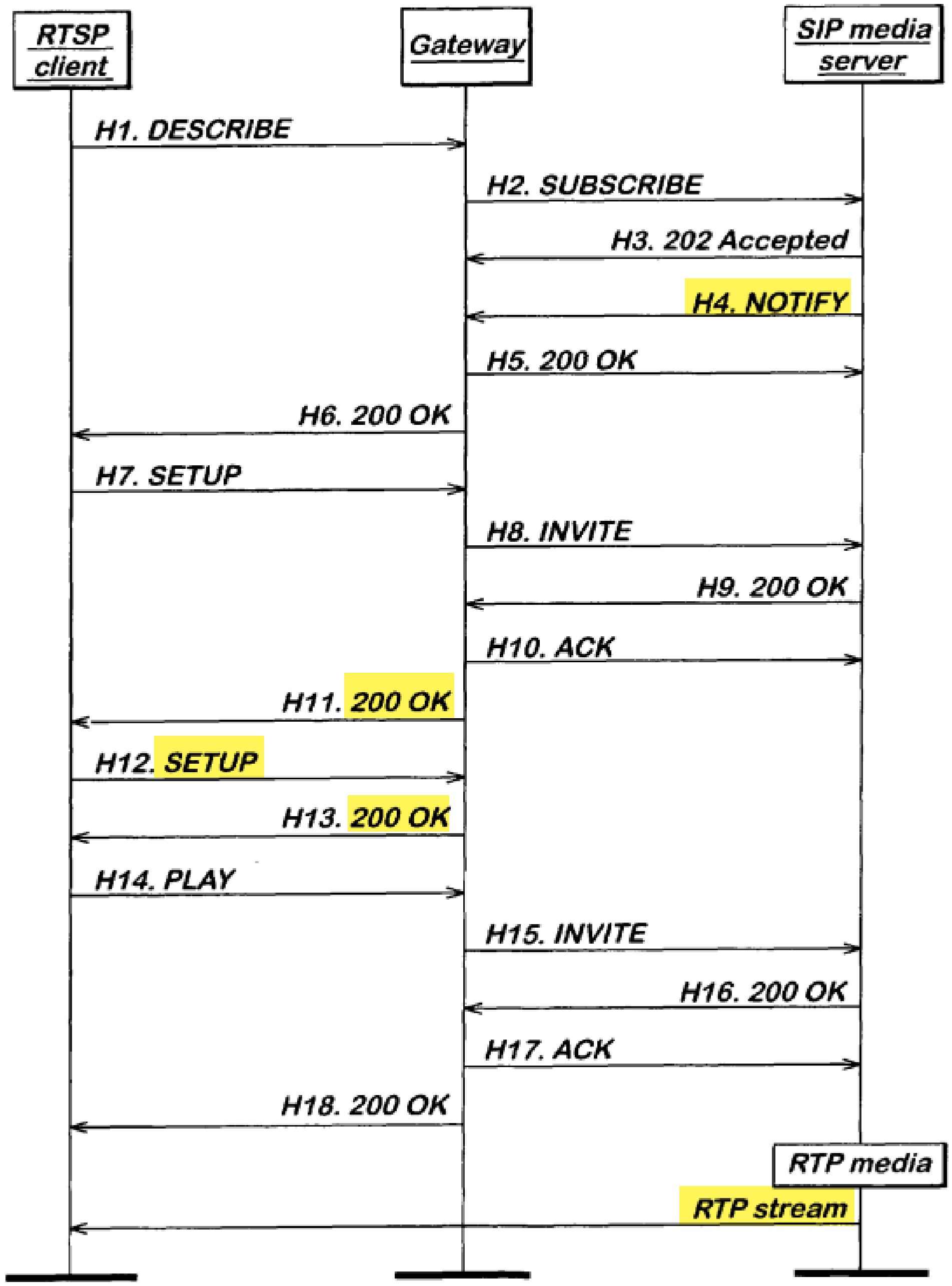
77. Gateva focuses on a method for controlling streaming services but also discloses well-known methods for RTSP and SIP terminals to communicate within an IMS system. For example, Fig. 2 illustrates "a terminal (UE)" using SIP signaling (steps 1, 12, 13, and 15) and a "streaming server (SS)" using RTSP signaling (steps 4-7) to set-up a media stream (step 18):



C. Mela (U.S. Patent No. 7,724,691) – Ex.1007

78. I understand that Mela qualifies as prior art.

79. Mela focuses on a communication system to facilitate communication between two or more entities such as user equipment and/or other nodes associated with the system, including via a SIP-RTSP gateway. Ex.1007, Abstract. As previously mentioned, in my opinion, certain embodiments of Mela were substantively considered by the Examiner during prosecution of the '151 Patent and other family members, however; in my opinion, the Examiner appears to have misunderstood Mela's solution for RTSP-SIP communication. Mela discloses a multi-step process that allows the SIP client to trigger the RTSP client when it wishes to send a stream by sending an SIP Message to ***notify*** the Gateway and RTSP client that a stream is available for it to stream. Ex.1007, Fig. 8, 20:22-27-23. This causes the gateway to send the trigger, a "***200 OK***" message, to the RTSP client, which triggers the ***SETUP*** and exchange of media session information in the ***SETUP*** and ***200 OK*** messages between the gateway and RTSP client that confirms the parameters for the RTP stream to be sent. Ex.1007, *Id*. This process is illustrated below:



Ex.1007, Fig. 8 (highlighting added). As shown above, Mela's solution allows for the SIP terminal that is sending media to notify and trigger the receiving RTSP terminal because of the pre-existing relationship set-up at the beginning steps, wherein the gateway is configured to change an RTSP DESCRIBE message into a

SIP SUBSCRIBE message that lets the SIP server know it wants a subscription to updates on new media streams. Ex.1007, 20:22-27-23. As stated above, the Examiner mistook the SUBSCRIBE message as the claimed "trigger" and DESCRIBE message as the claimed "first protocol" used "to exchange first media session information."

80. Moreover, Mela provides the explicit contents for each of the messages in the exchange as follows, which provides the specific information being exchanged, in my opinion these messages are exemplary of common protocol messages at the time and the contents of each disclosed message was also obvious and commonly known use at the time as Mela explicitly states that the fields message "were common to" RTSP and SIP for each message, thus each field was defined in RTSP and SIP (20:35-23:16):

> The request for streaming media by the RTSP client is announced by message H1 DESCRIBE from the RTSP client to the gateway.
>
> **<u>Message H1</u>**
>
> 0 DESCRIBE rtsp://media-gw.nokia.com/twister RTSP/1.0
> CSeq: 1
>
> This message is a RTSP message and it indicates what the RTSP client is requesting and where from, also version of RTSP is indicated to be 1.0
>
> . . .
>
> **<u>Message H2</u>**
>
> SUBSCRIBE sip:twister@media.nokia.com;user=stream SIP/2.0

From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 1 SUBSCRIBE
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:131.228.237.22>
Expires: 0
Event: session-description
Accept: application/sdp
Content-length: 0

Message H3

SIP/2.0 202 Accepted
To: <sip:twister@media.nokia.com;user=stream>
From: <sip:user@nokia.com>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@131.228.54.140;user=stream>
Event: session-description
Cseq: 1 SUBSCRIBE
Content-length: 0

Message H4

NOTIFY sip:131.228.237.22 SIP/2.0
From: <sip:twister@media.nokia.com;user=stream>
To: <sip:user@nokia.com>
Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95
Contact: <sip:twister@131.228.54.140;user=stream>
CSeq: 1 NOTIFY
Event: session-description
Range: smtpe-25=00:00:00–02:16:13
Content-type: application/sdp
Content-length: 131
v=0
o=media-server 1804289383 2 IN IP4 131.228.54.140
s=Twister
a=control: sip:twister@media.nokia.com;user=stream

t=0 0

a=sendonly

m=audio 0 RTP/AVP 0

m=video 0 RTP/AVP 26

Message H5

SIP/2.0 200 OK

To: <sip:user@nokia.com>

From: <sip:twister@media.nokia.com;user=stream>

Call-ID: 1c5e3780-5834-11d6-a882-00e00307ab95

CSeq: 1 NOTIFY

Content-length: 0

The gateway sends information about the requested streaming media to the

RTSP client in message H6 —200 OK.

Message H6

RTSP/1.0 200 OK

CSeq: 1

Content-Type: application/sdp

Content-Length: 164

v=0

o=- 2890844256 2890842807 IN IP4 172.16.2.93

s=Twister

a=control:rtsp://media-gw.nokia.com/twister

t=0 0

m=audio 0 RTP/AVP 0

a=control:audio

m=video 0 RTP/AVP 26

a=control:video

Fields in this message are common to RTSP and will be clear to a person skilled in the art.

The RTSP Client now starts streaming the movie and sends message H**7**—SETUP to the gateway.

<u>Message H7</u>

SETUP rtsp://media-gw.nokia.com/twister/audio RTSP/1.0

CSeq: 2

Transport: RTP/AVP;unicast;client_port=8000–8001

Fields in this message are common to RTSP.

The gateway receives the H**7** message from RTSP client and interprets message H**7** to a form that the SIP media server can understand. The gateway sends message H**8**—INVITE to the SIP media server. Communication between the gateway and the SIP media server is similar to previous examples, so that steps from H**8** to H**10** are similar to steps D**5** to D**7** in FIG. 4.

<u>Message H8</u>

INVITE sip:twister@media.nokia.com;user=stream SIP/2.0

From: <sip:user@nokia.com>

To: <sip:twister@media.nokia.com;user=stream>

Cseq: 1 INVITE

Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95

Contact: <sip:131.228.237.22>

Content-length: 0

Message H9

SIP/2.0 200 OK

To: <sip:twister@media.nokia.com;user=stream>

From: <sip:user@nokia.com>

Cseq: 1 INVITE

Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95

Contact: <sip:twister@131.228.54.140;user=stream>

Content-type: application/sdp

Content-length: 131

v=0

o=media-server 1804289383 321763 IN IP4 131.228.54.140

s=Twister

a=control: sip:twister@131.228.54.140;user=stream

t=0 0
c=IN IP4 131.228.54.140
a=sendonly
a=range:smtpe-25=00:00:00–02:16:13
m=audio 9004 RTP/AVP 0
a=rtpinfo: seq=92781365;rtptime=21375134
m=video 9002 RTP/AVP 26
a=rtpinfo: seq=9810092;rtptime=3450012
Message H10
ACK sip:twister@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 1 ACK
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Content-length: 83
v=0
o=– 816357754 1 IN IP4 172.21.164.126
s=–
t=0 0
c=IN IP4 172.21.164.126
a=inactive
m=audio 8000 RTP/AVP 0
m=video 0 RTP/AVP 26

The gateway sends confirmation to the RTSP client in message H**11**—200 OK.

Message H11
RTSP/1.0 200 OK
CSeq: 2
Transport: RTP/AVP;unicast;client_port=8000–8001;
source=131.228.54.140;server_port=9004–9005
Session: 12345678

Fields in this message are common to RTSP. The RTSP client sets up also the video session and sends message H**12** SETUP to the gateway

Message H12

SETUP rtsp://media-gw.nokia.com/twister/video RTSP/1.0
CSeq: 3
Transport: RTP/AVP;unicast;client_port=8002–8003

Fields in this message are common to RTSP. The Gateway confirms the request H**12** for adding of the video session by sending message H**13**—200 OK to the RTSP client.

Message H13

RTSP/1.0 200 OK
CSeq: 3
Transport: RTP/AVP;unicast;client_port=8002–8003;
source=131.228.54.140;server_port=9002–9003
Session: 12345678

Fields in this message are common to RTSP. The RTSP client requests that playing of the streaming media starts and sends message H**14** to the gateway.

Message H14

PLAY rtsp://media-gw.nokia.com/twister RTSP/1.0
CSeq: 4
Range: npt=0−
Session: 12345678

The gateway requests to the SIP media server that video streaming should also be included. The gateway sends message H**15**—INVITE to the SIP media server. Communication between the gateway and the SIP media server is similar to the previous examples, so that steps H**15** to H**17** are similar to steps H**8** to H**10**.

Message H15
INVITE sip:twister@131.228.54.140;user=stream SIP/2.0
From: <sip:user@nokia.com>
To: <sip:twister@media.nokia.com;user=stream>
Cseq: 2 INVITE
Call-ID: 87ed5e72-5834-11d6-a882-00e00307ab95
Contact: <sip:131.228.237.22>
Content-length: 0
v=0

81. Based on the teachings of Mela, including the messages in the preceding paragraph, a POSA would recognize that messages disclosed in Mela (H1-H15) are the same types of SIP and RTSP (including the INVITE, OK, ACK, DESCRIBE, SETUP) messages described in Gateva and Alston, and provided for by the SIP and RTSP standards.

D. Fajardo (U.S. Patent No. 8,068,460) – EX1019

82. I understand that Fajardo qualifies as prior art.

83. Fajardo discloses a "buffering system" and methods for "use[] in conjunction with existing mobility protocols [and] access protocols," (EX1019, Abstract), including "SIP [as] defined by the IETF." *Id.*, 4:44-4:59.

VIII. CLAIM CONSTRUCTION OF THE '151 PATENT

84. As I discussed above, I have been informed that for purposes of *inter-partes* reviews, the standard for claim construction of terms within the claims of the patent is the same as that applied in federal district court litigation. I have been asked

to assume that the claim terms otherwise have their plain and ordinary meaning to a person skilled in the art in light of the specification and the prosecution history.

85. As of this time, I am not aware of any term that requires specific construction for my opinions. To the extent Patent Owner suggests a narrow construction for a term, I reserve the right to respond to those opinions.

IX. SUMMARY OF GROUNDS

86. In the analysis that follows, I identify the following combinations of prior art that, in my opinion, anticipate and/or render obvious the ’151 patent claims:

Ground	Basis	Reference(s)	Claims
1	§103	Alston alone, or in view Fajardo	1-4, 7-10, 12-18, 22
2	§103	Alston (with or without Fajardo) in view of Gateva	1-18, 22
3	§103	Mela alone, or in view of Alston and/or Fajardo	1-5, 7-8, 10, 12-20, 22-23

X. GROUND I: CLAIMS 1-4, 7-10, 12-18, AND 22 ARE RENDERED OBVIOUS BY ALSTON ALONE OR IN VIEW OF FAJARDO

87. I provide a summary of this ground as follows: Ground I relies on the teachings from Alston in combination with the knowledge of a POSA regarding well-known, standardized, and established protocols and systems. Specifically, in my opinion, Alston discloses the use of SIP, a well-known protocol as evidenced above. As set out in the well-known standard documents, (such as the cited RFCs, Whitehead, and Avaya) describing the most current version of SIP, SIP was widely-recognized for its ability to set-up media session information by confirming the

parameters of the session with multiple exchanges of media session information (in the form of, *inter alia*, INVITE, OK, and ACK messages) between the sender and receiver. Moreover, as evinced by Alston and RFC2543, SIP's use in gateways was well-known and understood at the time. Accordingly, the '151 patent does not claim anything not already known in the art as of its priority date, and as further articulated below. To the extent Patent Owner argues that Alston does not disclose buffering, Fajardo discloses it, and it would be obvious to combine the two.

A. Claim 1

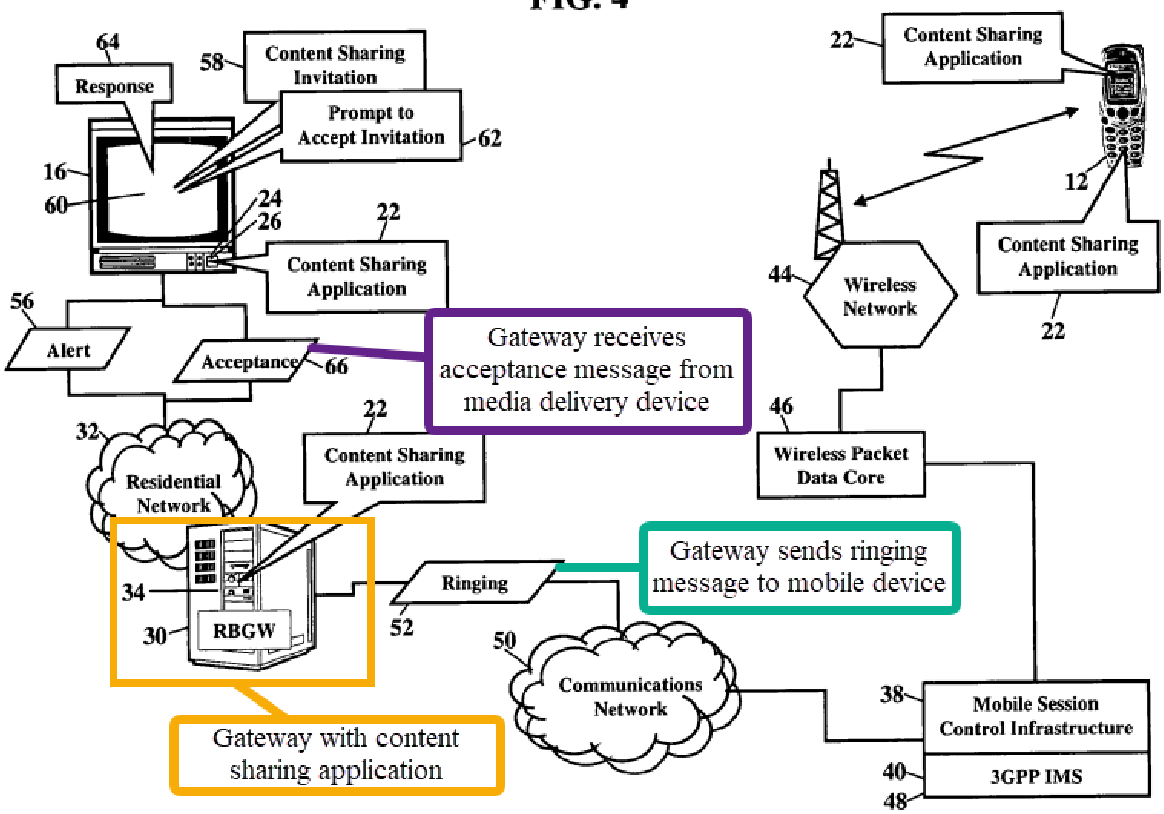
1. [1pre]: "A method, wherein a gateway is capable of communicating with a first terminal and a second terminal, the method comprising:"

88. In my opinion, Alston discloses or renders obvious this limitation, to the extent the preamble is limiting.

89. Alston describes a "residential broadband ***gateway*** (RBGW) 30" that uses a content sharing application ***capable of communicating with a first and second terminal***. Ex.1004, [0026], [0043]. For example, from a POSA would recognize that the media delivery device 16 is the second terminal and the mobile 12 is the first terminal as Fig. 4 illustrates Alston's teaching that the gateway 30 "receiv[es] an acceptance message 66" (Ex.1004, [0066]) from the media delivery device 16 (*i.e.* ***second terminal***) and sends "a ringing message 52 to the mobile

device 12" (*i.e.* ***first terminal***) (Ex.1004, [0032]):

FIG. 4



Id., Fig. 4.

2. [1a]: "the gateway or the first terminal using a first protocol to exchange first media session information between the first terminal and the gateway, wherein the exchange of the first media session information sets up a first media stream from the first terminal"

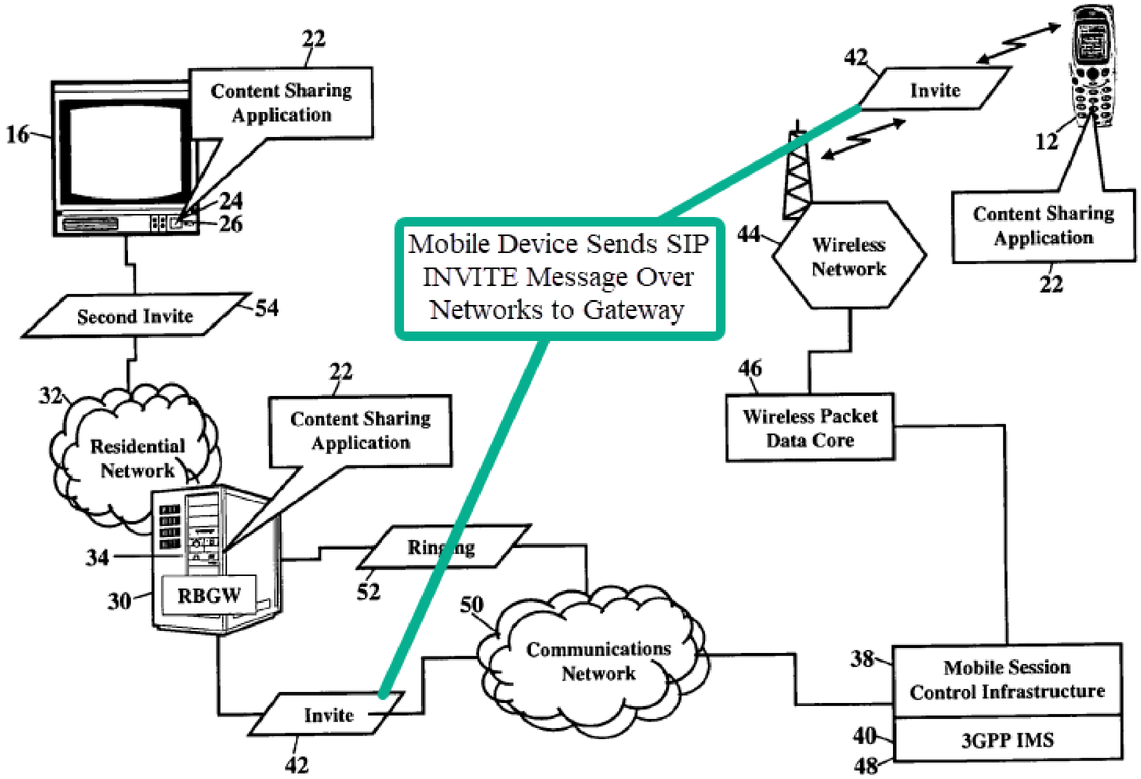
90. In my opinion, Alston discloses or renders obvious this limitation.

91. Alston teaches this limitation because Alston discloses that messages sent between the first terminal and gateway are SIP messages that contain media session information for setting up media sessions in accordance with SIP. *See*

Ex.1016, 1. The ’151 patent itself recognizes that it was well-known SIP was used to set up media sessions between terminals. EX1001, 1:26-28 (“In an IMS based architecture such multimedia streaming sessions between clients are set up using the Session Initiation Protocol (SIP)”).

92. With regard to the limitation “the gateway or the first terminal using a first protocol to exchange first media session information between the first terminal and the gateway,” Alston discloses that a “mobile device” (*i.e.* ***first terminal***) and ***gateway*** exchange SIP-protocol messages containing ***media session information***. Fig. 3 illustrates the “mobile device 12” (i.e., ***first terminal***) ***using a first protocol***, SIP, to “***send[] the SIP INVITE message*** 42” ***to the gateway***. Ex.1004, [0030]-[0032]; [0035].

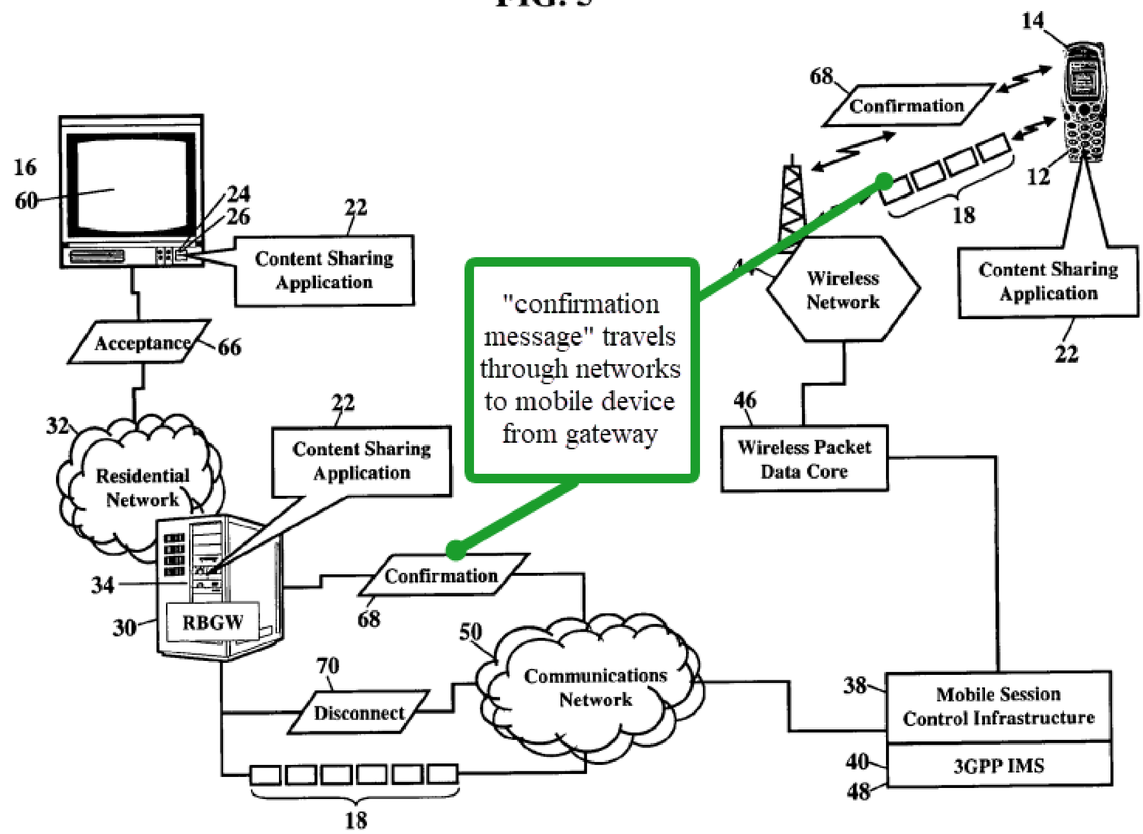
FIG. 3



Based on these disclosures of Alston and Fig. 3, in my opinion the limitation is disclosed by Alston.

93. If the media delivery device accepts the invite (*infra* Limitation [1b], [1c]), the "***gateway*** 30 ***sends the confirmation message*** 68 . . . ***to*** the mobile device 12," *i.e.*, ***the first terminal***, as illustrated in Fig. 68 below. Ex.1004, [0034]. Alston teaches the "confirmation message 68" is a "SIP OK message" which also uses SIP. Ex.1004, [0034]. In my opinion, a POSA would recognize that the SIP OK message uses SIP.

FIG. 5



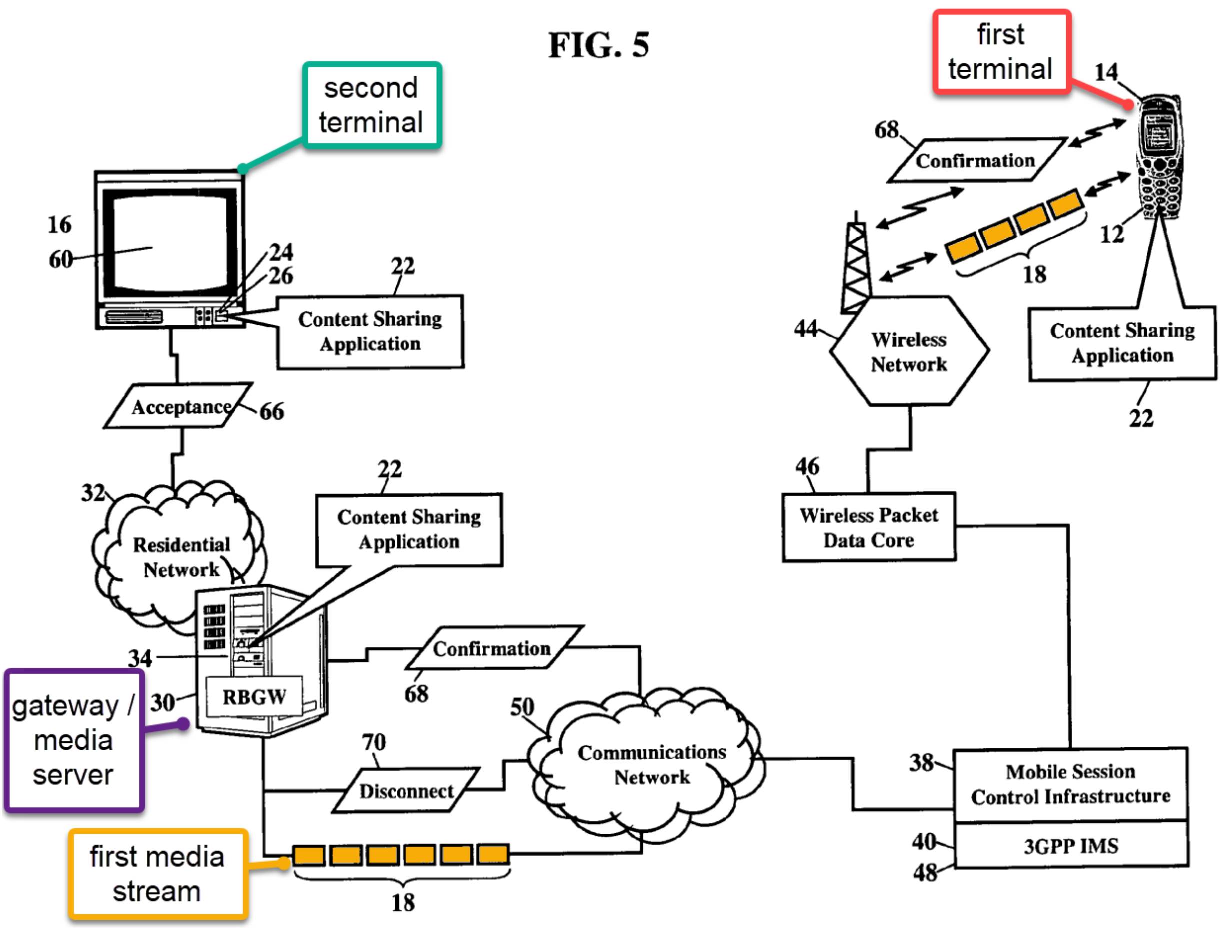
94. In my opinion, a POSA would understand, based on ordinary knowledge and Alston's teachings, that the use of SIP "INVITE" and "OK" messages contain ***media session information***. SIP is a well-understood protocol designed for setting up multimedia sessions (*i.e.*, media streams) which requires ***media session information to be exchanged*** in a series of messages (*i.e.*, SIP INVITE, OK, and ACK messages) as set out in the IETF standard RFC-3261, which defines SIP and was published in 2002, and as acknowledged by the '151 patent. Ex.1004, [0028] ("[a]s those of ordinary skill in the art understand, Session Initiation Protocol (SIP) is the Internet Engineering Task Force's standard for multimedia conferencing over IP"); *see, e.g.*, Ex.1016, 1, 7-9 (RFC3261); EX1001, 1:26-28

("[M]ultimedia streaming sessions between clients are set up using the Session Initiation Protocol (SIP)"). In my opinion, a POSA would understand that to set-up the media session using SIP, Alston's SIP "INVITE" and "OK" messages are used to ***exchange first media session information*** between the "gateway 30" and "mobile device 12" (*i.e.* ***first terminal***). Ex.1004, [0030]-[0035]; Ex.1016, 1, 7-9, 12-14. Therefore, a POSA would know that a SIP INVITE message includes an SDP/session description payload in the SIP INVITE for proposing a "session description" that includes the sender's media capabilities and other media-related information (e.g., media-type, URI, IP address, ports, etc.), in other words, ***media session information***, to the receiver to confirm common parameters for the media stream. Ex.1016, 1, 7-9, 11-14. Indeed, what is already evident to a POSA is confirmed by RF3261. Similarly, the 200 OK message also contains media session information as set-out in standard documents, although, again, it is already evident to a POSA that these messages must contain a session description (media session information) in order to confirm parameters of the media stream it wishes to set-up. Ex.1016, 13-14. In my opinion, a POSA would know that an INVITE message and 200 OK message are part of the "two-phase exchange provides basic negotiation capabilities and is based on a simple offer/answer model of SDP exchange." Ex.1016, 15. These "capabilities" include a determination of the media and media parameters (e.g., content-type) to be used, *i.e.*, the media type and additional first media session information. Ex.1016, 11, 15. As outlined above, the exchange of INVITE and 200 OK messages, such as disclosed in Alton, contains "session descriptions," allowing users to "agree on a set of compatible media types" and other information needed for setting up a session. Ex.1016, 1, 7-9. In addition, a POSA would have considered it obvious to combine Alston with RFC 3261. RFC 3261 is the well-known description of SIP, and a POSA would understand that Alston's express teachings of SIP messages would have been implemented using RFC 3261. Thus, the combination is merely applying a well-known technology that Alston expressly teaches should be used. This would in fact give effect to Alston's teachings. Moreover, a POSA would be motivated to look to RFC 3261 because it provides important and necessary details for implementing Alston's teachings.

95. With regard to the limitation "wherein the exchange of the first media session information sets up a first media stream from the first terminal," Alston discloses that the SIP INVITE message is sent to set up a first media stream from the first terminal. For example, Alston teaches that when the mobile device "desires to share content"—*i.e.*, ***a first media stream***—"with the subscriber's media delivery device . . . [t]he mobile device 12 sends an invite message 42 to the subscriber." Ex.1004, [0030]. Alston further teaches that, upon receipt of the "SIP OK" message, the mobile device initiates the process of "communicat[ing] the stream 18 of data containing any packetized data, such as 'live' or real time video images, video

clip(s), or image(s), and/or VoIP data" to the media delivery device that "visually and/or audibly presents the shared content," which is ***first media stream***. Ex.1004, [0034], Fig. 11, Fig. 12. Indeed, the standard documents confirm that these messages (and the exchange) are used for setting up a media stream from the first terminal that sends the INVITE message. *See*, e.g., Ex.1016, Abstract, Introduction. As I explain below for limitation [1e], and as shown in Figure 11, this ***first media stream*** is between the mobile device and a media server 96.

FIG. 5



96. I further note that the '151 patent itself teaches an embodiment that is the same as Alston's teachings for setting up a media stream. In particular, in Figure

5(a), the ’151 patent describes that a mobile device sends an SIP INVITE message, which is then forwarded by the gateway to the second terminal, which acknowledges the invite. The gateway then sends an SIP OK back to the mobile device, which results in set up of the media stream. Ex.1001, 11:53-12:48; Fig. 5(a).

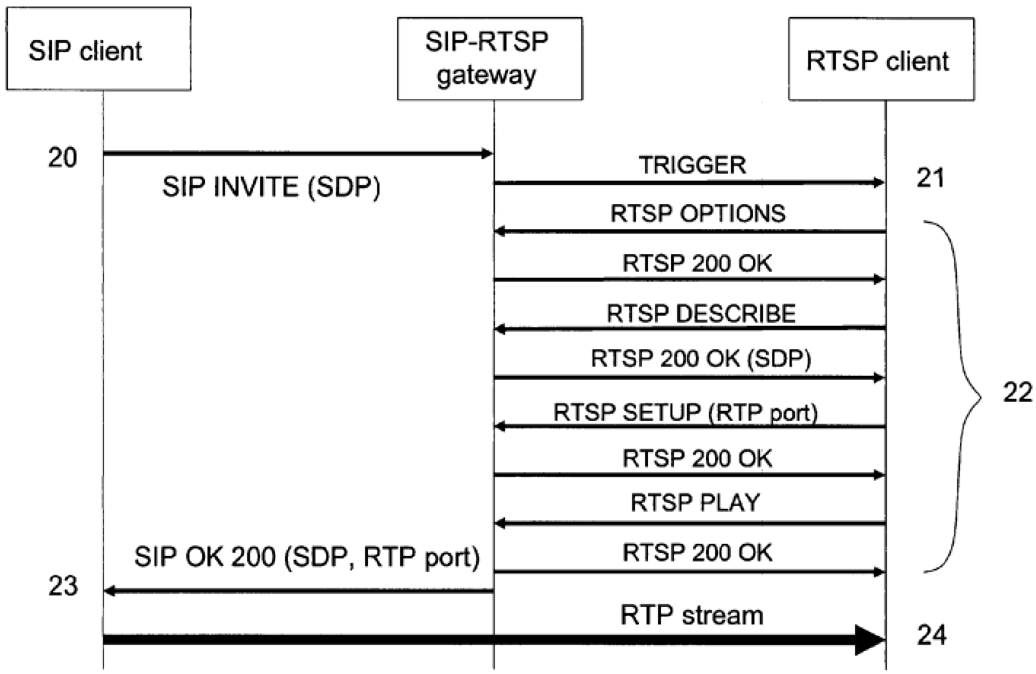


Figure 5 (a)

3. [1b]: “the second terminal using a second protocol to receive a trigger provided by the gateway, wherein the trigger is to initiate an exchange of second media session information between the gateway and the second terminal”

97. In my opinion, Alston discloses or renders obvious this limitation. To the extent not already addressed, in my opinion there is nothing in the claims that suggest that the “first,” “second,” and “third” cannot all be the same protocol. For example, dependent claim 14 and 15 are directed to limitations wherein the first and

second protocol are identical or both SIP. Alston teaches this limitation at least because Alston discloses that messages sent between the gateway and second terminal in response to the trigger messages are SIP messages which, in accordance with SIP, include media session information for setting up media sessions. Ex.1016, 11-16. As the standard illustrates, an "INVITE ***request***" message triggers the exchange of the media session information as "[e]ach transaction consists of a request that invokes a particular method, or function, on the server and at least one response." *Id.* And that INVITE invokes the 200.OK and ACK messages. *Id.* Based on the standard documents, the preferred method of practicing the standard using an INVITE message to trigger the exchange of media session information with the 200.OK and ACK messages. *Id.*

98. With regard to the limitation "the second terminal using a second protocol to receive a trigger provided by the gateway," Alston discloses that after the "gateway" receives the "invite message," the gateway's content sharing application sends an invite message, *i.e.* ***a trigger***, to the media delivery device, *i.e.* ***second terminal***. Ex.1004, [0032] ("content sharing application 22, operating in the residential broadband . . . sends a second invite message 54 to the media delivery device 16."). Alston expressly discloses that this second invite message may use "any" protocol, including "SIP" (*i.e.* ***second protocol***), based on its greater industry acceptance. "[T]he second invite message 54 may use any messaging protocol" but

"[b]ecause SIP has gained greater industry acceptance . . . the second invite message . . . utiliz[es] SIP" in Alston's exemplary embodiment. *Id.*, [0032]. The claims of the '151 patent allow for the first and second protocol to use the same SIP. *See* cls. 14, 15.

99. With regard to the limitation "wherein the trigger is to initiate an exchange of second media session information between the gateway and the second terminal," after receiving the "second invite message" (*i.e.*, ***the trigger***), "the processor 26 sends an acceptance message 66" to the gateway (Ex.1004, [0033]) which contains a session description (SDP), in other words, the second invite message ***initiates an exchange of the second media session information***. Ex.1016, 1-2. Indeed, this is the same process described above for the exchange of session information in Ground I, [1a]. As I stated there, which I incorporate here by reference, the trigger tells the second terminal it is requesting a stream and initiates the process of exchanging second media session information.

100. In accordance with RFC-3261, the acceptance message for accepting a media session invite is the OK message. EX1016, 12 (depicting "Bob's SIP Phone" sending "200 OK" in response to "SIP INVITE"); 15 ("[H]is SIP phone sends a 200 (OK) response to indicate that the call has been answered"). This OK message contains a session description (SDP) which contains ***second media session information*** including media type to be used by the terminal and gateway. EX1016,

15 (“The 200 (OK) contains a message body with the SDP media description of the type of session that Bob is willing to establish with Alice.”); 16 (“The Content-Type and Content-Length … contain[] Bob’s ***SDP media information***.”). As I discussed for limitation [1a], a POSA would have known and considered it obvious that Alston’s disclosure of SIP messaging includes the implementation details of RFC-3261, including the fact that a SIP OK message is sent to accept a SIP INVITE.

4. [1c]: “in response to receiving the trigger, the second terminal using a third protocol to initiate the exchange of the second media session information between the gateway and the second terminal, wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates”

101. In my opinion, Alston discloses or renders obvious this limitation.

102. Alston teaches this limitation because Alston discloses that messages sent between the gateway and second terminal in response to the trigger messages are SIP messages which would have included media session information as this was the standard way to use SIP at the time. As provided in the [1b] paragraphs directly above, the trigger (INVITE) initiates the sending of the 200 OK message, which uses SIP, a third protocol. To the extent not already addressed, in my opinion there is nothing in the claims that suggest that the “first,” “second,” and “third” cannot all be the same protocol. For example, dependent claim 14 and 15 are directed to limitations wherein the first and second protocol are identical or both SIP. Further,

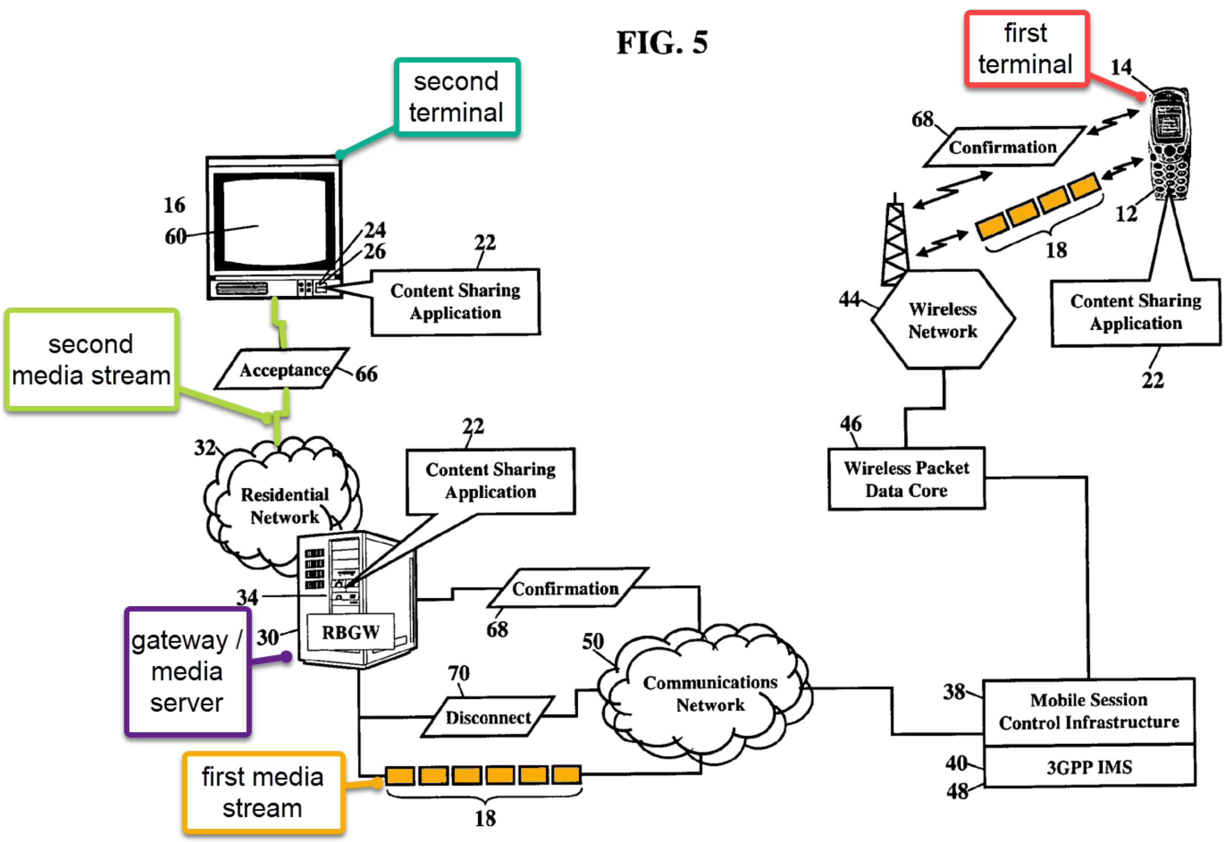
it is evident that the second terminal participates in the claim because it receives the stream and can also end the stream.

103. With regard to the limitation, "in response to receiving the trigger, the second terminal using a third protocol to initiate the exchange of the second media session information between the gateway and the second terminal," as I explained in Ground I, [1b], Alston teaches that the "second INVITE message" triggers the second terminal to exchange media session information with the gateway in the form of an "OK" message. The SIP OK message contains SDP, *i.e.*, media session information, including media type. Ex.1004, [0030]-[0034]; *see also* Ground I, Limitation [1b]. Specifically, as provided in Ground I, [1a] and [1b], the 200.OK/"OK" message contains media session information including the media type. Ex.1016, 1, 7-9, 11-16. The standard further explains that this exchange is for a "determination of the media and media parameters to be used," *i.e.*, the media type and additional first media session information. Ex.1014, 7.

104. With regard to the limitation "wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates," Alston discloses the acceptance message (*i.e.* ***exchange of second media session information***) results in ***setting up a*** second session between the media delivery device and gateway (*i.e.*, a ***second media stream***). Specifically, after the media delivery device transmits a SIP OK acceptance message to the

gateway, the gateway sends a "confirmation message" (which "may be a SIP OK message") to the "mobile device 12 indicat[ing] that the subscriber has accepted the session invitation." EX1004, [0034]. The "mobile device" then "communicates the stream 18" to the gateway, which then "forwards the stream of data during a session with the media delivery device 16," *i.e.* ***second media stream***. EX1004, [0044]. In other words, as provided in limitation [1a], the stream from the mobile device to the gateway is the ***first media stream***. The ***second media stream*** is the stream between the media delivery device (*i.e.* ***second terminal***) and the gateway as annotated in Fig. 5 (EX1004):

FIG. 5



105. As provided for in Alston, in my opinion a POSA would understand that the second terminal participates in the second media stream at least by receiving the stream and by having the ability to unilaterally end the streams, *e.g.,* when "the media delivery device decides to end the session, [the] device provides a disconnect message to terminate the session per standard SIPs." Ex.1004, [0034]; [0044].

5. [1d]: "the second terminal receiving the second media stream, wherein the second media stream is associated with the first media stream"

106. In my opinion, Alston discloses or renders obvious this limitation. Alston discloses that the media "forwards the stream of data during a session with the media delivery device 16"—*i.e.*, ***the second terminal receives the second media stream***. Ex.1004, [0044].

107. Based on the express teachings of Alston (Ex.1004, [0026], [0032]), including that the "gateway" is a back-to-back user agent ("B2BUA"), a POSA would understand that the gateway is specifically configured to associate the first multimedia session information with the second multimedia session information, thereby forming an association between the first media stream and second media streams in a way that ensures seamless operation and logical continuity. Alston's disclosure emphasizes that the B2BUA acts as a critical intermediary that bridges and manages the two sessions, maintaining robust communication pathways by coupling these streams internally. While each session is treated as independent in

protocol terms to handle distinct signaling and media requirements, the B2BUA's internal state ensures these sessions are not isolated in function. Instead, it creates and preserves a tightly integrated mapping between them. This mapping is pivotal, as it allows the B2BUA to handle signaling efficiently and ensure media flows uninterrupted between endpoints, even if they operate on different networks or require protocol translations. By managing this association, the B2BUA effectively enables continuity between endpoints, accommodating a range of scenarios where multimedia communications rely on the gateway to provide seamless integration of signaling and media exchange. The result is a system where endpoints can communicate as if directly connected, even when the underlying complexity of protocol variations and session handling might otherwise disrupt or complicate the exchange. Alston's teachings clearly underscore this role of the B2BUA, illustrating its indispensability in achieving coherent and uninterrupted multimedia communication across diverse network environments.

108. Further, even without the explicit disclosure of a B2BUA, a POSA would find this limitation to be obvious based on the functionality and operation of the system described. Specifically, Alston discloses that once "the media delivery device 16 has accepted the session invitation... mobile device 12 then communicates the stream of data... to the [RBGW]," which "then forwards the stream of data during a session with the media delivery device." This statement reflects a system where

the stream of data—described as “packetized data, such as ‘live’ or real-time video”—received by the gateway from one terminal is the same stream of data that is “forwarded” to the second terminal during an ongoing session. In this context, the gateway functions as a conduit that facilitates the seamless flow of media streams between the two endpoints, ensuring continuity and logical alignment of the communication process. In my opinion, a POSA would understand that the act of forwarding this data stream during an ongoing session will involve an association between the first media session and the second media session. For example, the gateway, by receiving and forwarding the same stream of data, necessarily maintains a state or mapping that ties the two sessions together, even if this association is not explicitly characterized as the function of a B2BUA. This association is integral to ensuring that the transmitted data remains synchronized and contextually relevant for both endpoints, allowing for uninterrupted and cohesive communication. Further, a POSA would recognize that such a system relies on the ability to associate sessions as a natural and necessary step to achieve the described functionality. The process of session initiation, acceptance, and subsequent data forwarding requires the gateway to track and manage the relationships between the incoming and outgoing streams, effectively coupling the sessions. Therefore, in my opinion, this functionality clearly demonstrates that the second media session is associated with the first media session, fulfilling the claimed limitation, even without an explicit disclosure of a B2BUA. The technical principles and practical implementation outlined in Alston (Ex.1004, [0025]) make this conclusion both logical and evident to a POSA. As a separate reason, once "the media delivery device 16 has accepted the session invitation . . . mobile device 12 then communicates the stream of data . . . to the [RBGW]" which "then forwards the stream of data during a session with the media delivery device." In other words, the stream of data ("packetized data, such as 'live' or real time video") delivered to the gateway is the same data that is "forwarded" to the second terminal. Ex.1004, [0025]. Accordingly, in my opinion this means the second media session is associated with the first media session.

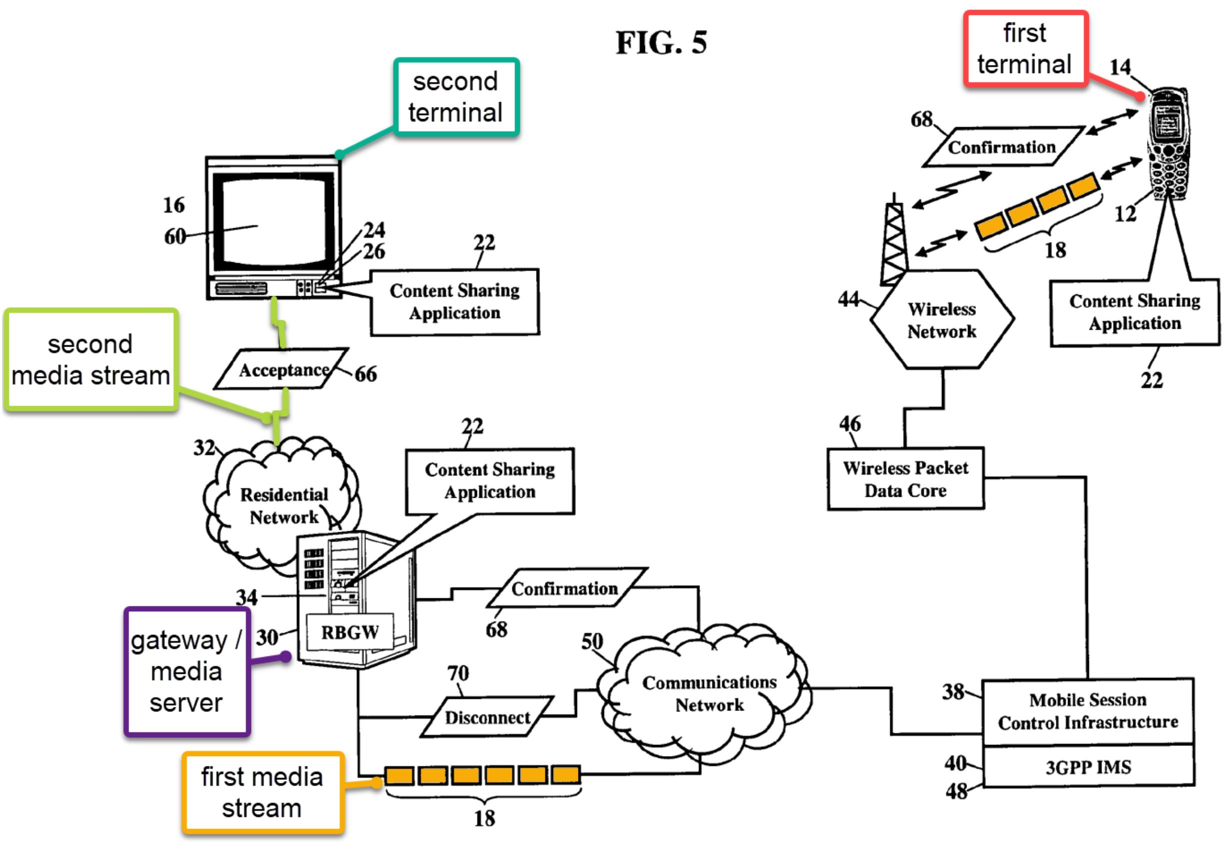
109. Moreover, the association is obvious. A POSITA would be motivated to associate the streams to give effect to Alston's teaching that streams are forwarded, by associating which incoming stream should be mapped to which outgoing stream such that the stream is forwarded to the correct terminal. A POSITA would find such association beneficial as it allows for proper forwarding of streams as taught in Alston.

6. [1e]: "wherein the first media stream is between the first terminal and a media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal."

110. In my opinion, Alston discloses or renders obvious this limitation.

111. With regard to the limitation "wherein the first media stream is between the first terminal and a media server," as I discussed above in limitations [1a] and [1c] and as annotated in Figure 5 below, the first media stream is between the first terminal and the gateway and the second media stream is between the second terminal and the gateway. The gateway is a ***media server*** because, as Alston teaches, the gateway may "locally store the stream 18 in the local memory location 34" to send to the media delivery device "at a later time." EX1004, [0046]. This is consistent with how the '151 patent describes a media server, which describes that it "allows time-based access to files on a server." EX1001, 10:49-53.

FIG. 5



112. With regard to the limitation "wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal," Alston discloses that the gateway "locally and/or remotely stores the stream 18, the shared content can be retrieved at a later time" by the media delivery device, *i.e.* ***second terminal***. EX1004, [0046]. In my opinion, a POSA would recognize that this temporary storage acts as a buffering mechanism for the media stream by temporarily storing the media stream before later transmitting it to the subscriber's "media delivery device," *i.e.*, ***buffering*** the entire ***first media stream received from the first terminal before transmitting the second media stream to the second terminal***.

113. To the extent Patent Owner argues that a POSA would not understand this disclosure as ***buffering***, in my opinion a POSA would find buffering to be an obvious addition as it would be a requirement for most any real-word implementations. For example, in my opinion, a POSA would understand that buffering is a necessary step in the context of Alston because it ensures seamless handling of data streams during storage or transmission by "server 96" because without buffering, the server may struggle to manage variability in network speeds, packet arrival times, or processing delays, which could result in data loss, corruption, or incomplete storage. Buffering allows the server to temporarily hold incoming data while it is written to storage or prepared for forwarding, ensuring that slower

downstream components or transient processing delays do not disrupt the flow of the stream. *See infra* Ground II, Claim [1e] (further discussing the obviousness of buffering systems to media delivery systems). As also taught by Alston, a POSA would understand the benefits of buffering, including so that one could store the stream when "the media delivery device 16 is unavailable." Ex.1004, [0046].

114. I also note that Fajardo expressly teaches when a "mobile device us[ing] the Session Initiation Protocol (SIP)" and specifically "IP Multimedia Subsystem," it is beneficial to use "playout buffers … to minimize the effects of packet loss or to fix small gap in the traffic." EX1019, 4:48-59, 7:55-59. Fajardo also teaches that for "excessive packet loss" or "real-time applications … sensitive to loss and delay," a "per-mobile packet buffer at an access router or network entity" can be employed. *Id.*, 7:60-8:11. In view of these teachings, in my opinion, a POSA would find it obvious to incorporate buffering into Alston for the same reasons I discussed above, which are consistent with Fajardo's teachings of the benefits of buffering to mitigate loss of connectivity. A POSA would have had a reasonable expectation of success, because media buffering was well-known and well-within the skill of an ordinary artisan to implement. *Id.* In addition, Alston, Fajardo, and the '151 patent are directed to the same field of multi-media content delivery and IMS/3GPP systems. *Id.*; EX1001, 1:6-8, 1:16-18; EX1004, [0002], [0029]; EX1019, 4:48-59, 7:55-59.

B. Claim 2: "The method according to claim 1, wherein the exchange of the first media session information comprises: the gateway providing to the first terminal at least part of the second media session information."

115. In my opinion, Alston discloses or renders obvious this limitation.

116. Alston teaches this limitation because the SIP messages that are used to share and confirm the sessions parameters between the first terminal and gateway each contain session parameters, including a media type, as required by the SIP standard. *See* Ex.1004, [0028], [0030]-[0032], [0035]; Ex.1014, 9, 147-157; *see also* Ground I, [1a]. In my opinion, the "OK" message provided by the gateway to the first terminal contains the media-type (as provided in Ground I, [1a]) that will be used in both the first and second session. As provided in Ground I, [1e], the "data stream" (media stream) is forwarded by the gateway/B2BUA to the second terminal, thus, the data stream remains the same media type. Thus, the media type provided in the "OK message" is also part of the second media session information. The gateway must confirm the second terminal's session parameters to the first terminal so that both terminals are using the same session parameters. Ex.1004 [0030]. In my opinion, a POSA would understand that this confirmation process is for maintaining session integrity, as it ensures the endpoints are synchronized in their expectations for media handling. Without such synchronization, discrepancies in media type could result in communication errors, such as incompatible parameters or dropped sessions. Additionally, the SIP standard explicitly requires that session

parameters, including media type, be consistently negotiated and communicated between all involved entities to ensure interoperability and seamless media exchange.

C. Claim 3: "The method according to claim 1, wherein the exchange of the second media session information comprises: the gateway providing to the second terminal at least part of the first media session information."

117. In my opinion, Alston discloses or renders obvious this limitation.

118. Alston teaches this limitation because the SIP messages that are used to share and confirm the sessions parameters between the second terminal and gateway each contain session parameters, including a media type, as required by the SIP standard. Ex.1004, [0028], [0030]-[0033], [0035]; Ex.1014, 9, 147-157; *see also* Ground I, [1b]. Specifically, the "INVITE" message provided by the gateway to the second terminal contains the media-type (as provided in Ground I, [1b]) that will be used in both the first and second session. As provided in Ground I, [1e], the "data stream" (media session) is forwarded by the gateway/B2BUA to the second terminal, thus, the data stream remains the same media type. Accordingly, the media type provided in the "INVITE message" is also part of the first media session information. The gateway must confirm the first terminal's session parameters to the second terminal so that both terminals are using the same session parameters. Ex.1004 [0030]. In my opinion, a POSA would recognize that the confirmation of session parameters through the "INVITE" message is essential for maintaining consistency

and compatibility between the two sessions, as it ensures that both terminals interpret and process the media stream correctly. The SIP standard requires that such acknowledgments explicitly communicate the agreed media type to prevent any discrepancies that could disrupt communication, such as mismatched parameters or unsupported media formats. Furthermore, this step reflects the functionality of a B2BUA, which ensures that both terminals are synchronized in their understanding of the media stream while maintaining the logical continuity of the session.

D. Claim 4: "The method according to claim 1, wherein the exchange of the first media session information comprises: the gateway receiving, from the first terminal, an identifier of the second terminal."

119. In my opinion, Alston discloses or renders obvious this limitation.

120. Alston teaches this limitation because a POSA would understand that the "INVITE" message provided to the gateway by the first terminal during the exchange of first media session information must identify to the gateway the final destination of its message, thus ***the gateway*** must ***receive, from the first terminal, an identifier of the second terminal*** because the gateway must know where to forward the invite. *See* Ground I, [1c]-1[e], [3]; Ex.1004, [0030]-[0034]. This was known in the art, including because the standard discloses "SIP URLs are used within SIP messages to indicate the originator (From), current destination (Request-URI) and ***final recipient*** (To) of a SIP request. Ex.1014, 18; Ex.1009, 3. In my opinion, a POSA would recognize that this fundamental requirement of SIP ensures

that the signaling flows correctly through intermediate nodes, such as gateways, to reach the intended recipient. Without an explicit identifier of the second terminal, the gateway would be unable to establish or forward the session, breaking the communication flow. Additionally, the use of SIP URLs as identifiers ensures that session routing remains consistent and reliable across varied network environments, further underscoring the necessity of including the second terminal's information in the "INVITE" message to achieve proper functionality in SIP-based communications.

E. Claim 7: "The method of claim 1, wherein the third protocol is different from the second protocol."

121. In my opinion, Alston discloses or renders obvious this limitation.

122. Alston discloses that the "second invite message may use any messaging protocol" and, similarly, the "confirmation message 68 may be any message or information that acknowledges acceptance of the session invitation." Ex.1004, [0032], [0034]; See also Limitation 1[b] and 1[c]. In my opinion, a POSA would understand that in many real-world uses the third protocol (which Alston discloses is SIP) would be different from the second protocol (which Alston discloses could be "any . . . protocol"), particularly given the complexities and evolving demands of IMS systems before 2006. The ability to accommodate different protocols was not merely a theoretical concept but a practical necessity driven by multiple challenges to the telecommunications landscape at the time.

These challenges spanned issues of interoperability, device diversity, media adaptation, and global variations in protocol implementation. In my opinion, the ability of gateways to support multiple protocols ensured that communication between heterogeneous systems could occur seamlessly, meeting user expectations for consistent connectivity regardless of network or device differences. This adaptability was crucial for the practical deployment of IMS systems, which relied on gateways to mediate between diverse and evolving technologies to deliver reliable multimedia services. Further, as recognized in Whitehead and other prior art references, a POSA would understand that many dual-stack solutions existed by 2006, wherein a terminal could switch between protocols when setting up a session. These solutions were developed to address the practical challenges of transitioning from legacy systems to IMS-based networks, ensuring compatibility between diverse signaling and media protocols like SIP, H.323, and proprietary standards. Dual-stack systems enabled terminals to dynamically select the appropriate protocol depending on the capabilities of the network or the destination endpoint, a critical feature in mixed environments where full protocol standardization had not yet been achieved. For instance, a terminal might initiate a session using SIP but fall back to a legacy protocol like H.323 when communicating with older systems, ensuring seamless interoperability. This flexibility was essential to the deployment of IMS systems already in place at the time, where gateways and terminals needed to bridge the gap between emerging and existing technologies to support the growing demand for multimedia services.

F. Claim 8: "The method of claim 1, wherein the first terminal transmits the first multimedia stream and the second terminal receives the second multimedia stream."

123. In my opinion, Alston discloses or renders obvious this limitation.

124. Alston discloses that "[t]he mobile device 12 then communicates the stream of data (shown as reference numeral 18 in FIG. 5) to the residential broadband gateway 30," *i.e.* ***first terminal transmits the first multimedia stream***, and "[t]he residential broadband gateway 30 then forwards the stream of data during a session with the media delivery device 16," *i.e.* ***second terminal receives the second multimedia stream***. EX1004, [0044

G. Claim 9: "The method of claim 1, wherein the trigger serves to inform the second terminal of availability of a new multimedia stream."

125. In my opinion, Alston discloses or renders obvious this limitation.

126. Alston discloses an embodiment wherein the invite message is used to "provide[] an audible and/or visual indication or alert of the mobile user's invitation to share video"—*i.e.*, ***inform the second terminal of the availability of a new multimedia stream***." Ex.1004, [0039]. A POSA would recognize that this functionality is needed for ensuring that the second terminal is effectively informed of the availability of a multimedia stream, enabling timely user action to accept or

retrieve the content. The invite message serves not only as a technical signaling mechanism but also as a notification to the end user, bridging the gap between system-level communication and user engagement. Without such a mechanism, the second terminal would lack the necessary context to initiate its session with the server or gateway, resulting in missed opportunities for session establishment and media sharing.

H. Claim 10: "The method of claim 1, wherein the first protocol is a peer-to-peer protocol."

127. In my opinion, Alston discloses or renders obvious this limitation.

128. Alston teaches that ***the first protocol is***, SIP, ***a peer-to-peer protocol***. The '151 patent explains that SIP is a peer to peer protocol. Ex.1001, 2:16-17 ("for example a peer-to-peer type of protocol, such as the SIP"); Ex.1004, [0030]-[0034].

I. Claim 12: "The method of claim 1, wherein the trigger comprises a network address of the gateway."

129. In my opinion, Alston discloses or renders obvious this limitation.

130. Alston discloses or renders obvious that the second INVITE message (trigger) ***contains a network address of the gateway***. In my opinion, a POSA would understand that SIP messages include URLs, *i.e.,* network addresses "that are used within SIP messages to indicate the originator" of the message. Ex.1016, 36-37 ("the From header field indicates the logical identity of the initiator of the request, possibly the user's address-of-record"); Ex.1014, 20. In the case of the second INVITE

message, the originator is the gateway, therefore the second INVITE message contains a network address of the gateway. In my opinion, this process is fundamental to SIP's role in establishing VoIP, video calls, and other real-time communication sessions. In my opinion, a POSA would recognize that the inclusion of the gateway's network address in the second INVITE message is needed for proper routing and session establishment, as it ensures that all subsequent signaling and media flows are directed correctly through the gateway. This is a core requirement in SIP-based systems, where the logical identity of each participant, including intermediaries like gateways, must be explicitly defined to maintain session continuity and enable reliable communication. Without the gateway's network address embedded in the second INVITE message, the downstream terminal would lack the necessary information to interact with the gateway, disrupting the overall session setup process and compromising the communication pathway.

J. Claim 13: "The method of claim 12, wherein the network address is a URI or a URI reference."

131. In my opinion, Alston discloses or renders obvious this limitation.

132. Alston renders obvious that the second INVITE message includes the network address, a URL, which is a type of URI as set out by the standards at the time. *See* Ground I, [12]; Ex.1009, 3 ("The term "Uniform Resource Locator" (URL) refers to the subset of URI that identify resources via a representation of their

primary access mechanism”). In my opinion, as set out in the standard, a POSA would understand that URLs, as a subset of URIs, were a standard mechanism for identifying and locating resources within SIP messages to enable proper routing and communication. In the context of the second INVITE message, including the network address in the form of a URL ensures that the recipient terminal or server can correctly identify and communicate with the gateway, as required for establishing or continuing the multimedia session.

K. Claim 14: “14. The method of claim 1, wherein the first protocol and the second protocol are identical.”

133. In my opinion, Alston discloses or renders obvious this limitation.

134. Alston discloses that both the first protocol and second protocol are SIP, therefore, ***the first protocol and the second protocol are identical***. Ex.1004, [0030]-[0035]; *See* Ground I, [1a-b].

L. Claim 15: “The method of claim 14, wherein the first protocol and the second protocol are Session Initiation Protocol (SIP).”

135. In my opinion, Alston discloses or renders obvious this limitation.

136. Alston teaches that both the first protocol and second protocol are Session Initiation Protocol (SIP). Ex.1004, X; *See* Ground I, [1a-b], [14].

M. Claim 16: “The method of claim 1, wherein the gateway is configured to associate the first multimedia session information with the second multimedia session information.”

137. In my opinion, Alston discloses or renders obvious this limitation. Based on the teachings of Alston that the “gateway” is a B2BUA, a POSA would

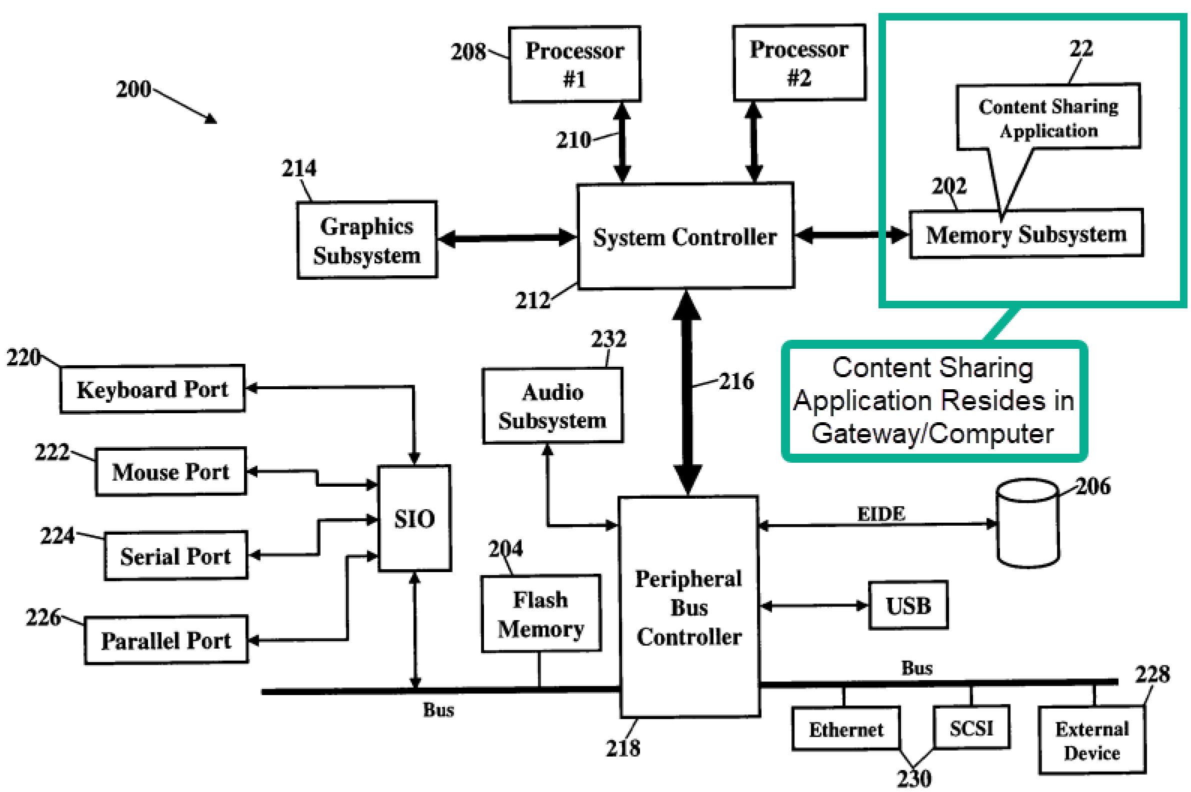
understand that the gateway would be ***configured to associate the first multimedia session information with the second multimedia session information***. Ex.1004, [0026], [0032]; *see* claim [1d]. Specifically, a B2BUAs, like those described in Alston, must associate or map the session information for each side of the stream it handles to manage and maintain logical continuity and ensure seamless communication between endpoints. While each session is treated independently in protocol terms, the B2BUA's internal state and mapping ensure the two sessions are tightly coupled, enabling effective signaling and media handling. In my opinion, a POSA would recognize that this mapping is fundamental to the operation of a B2BUA, as it ensures that signaling and media streams remain synchronized and correctly routed. This association is needed for handling features such as call transfers, conferencing, or media transcoding, where the seamless coordination of multiple streams across independent sessions is required. Furthermore, this functionality allows the gateway to enforce consistent session parameters, such as media types, ensuring compatibility and uninterrupted communication between the endpoints.

N. Claim 17:

1. [17pre]: "An article of manufacture including a non-transitory computer-readable medium, having stored thereon software instructions that, if executed by a computing device, cause the computing device to perform operations comprising:"

138. In my opinion, Alston discloses or renders obvious this limitation. Alston discloses "operating environment" for its teachings, including a "content sharing application 22 resid[es] in a computer system" such as the "gateway" can perform the functions taught by Alston. Ex.1004, [0048]-[0049]. Alston's Fig. 14 illustrates, the "content sharing application 22 operates within a system memory device" (Ex.1004, [0049]):

FIG. 14



Id., Fig. 14.

139. Alston further teaches that the "content sharing application 22 may be stored in any memory location" or "flash memory." Ex.1004, [0048]-[0049]. A POSA would understand from this teaching that Alston's methods can be carried out by a gateway (*i.e.*, ***article of manufacture***) that ***includes*** a "flash memory" or "memory subsystem" (*i.e.*, ***a non-transitory computer-readable medium***") ***having stored thereon*** a "content sharing application" (*i.e.*, ***software instructions that, if executed by a computing device, cause the computing device to perform operations***). Ex.1004, [0048]-[0049].

2. [17a]: "using a first protocol to initiate an exchange of first media session information between a first terminal and a gateway, wherein the exchange of the first media session information sets up a first media stream from the first terminal"

140. In my opinion, Alston discloses or renders obvious this limitation. This limitation includes substantially the same language as Limitation [1a], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that limitation. *See* Limitation [1a], which explains how Alston discloses the use of SIP (*i.e.*, a first protocol) to initiate an exchange first media session information between the first terminal and the gateway to set up a media stream from the first terminal. To the extent there are any differences in the language of the two limitations, I

considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

3. [17b]: "using a second protocol to provide a trigger to a second terminal, wherein the trigger is to initiate an exchange of second media session information between the gateway and the second terminal"

141. In my opinion, Alston discloses or renders obvious this limitation. This limitation includes substantially the same language as Limitation [1b], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that limitation. *See* Limitation [1b], which explains how Alston discloses the use of SIP (*i.e.,* a second protocol) to provide a trigger to a second terminal, wherein the trigger is to initiate an exchange of second media session information between the gateway and the second terminal. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

4. [17c]: "using the second protocol to conduct the exchange of the second media session information between the gateway and the second terminal, wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates"

142. In my opinion, Alston discloses or renders obvious this limitation. This limitation includes substantially the same language as Limitation [1c], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that

limitation. *See* Limitation [1c], which explains how Alston discloses the use of SIP (*i.e.*, ***the second protocol***)[1] to conduct the exchange of the second media session information between the gateway and the second terminal, wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

5. [17d]: "using the first protocol to transmit a response to the first terminal, wherein the response causes the first terminal to begin transmission of the first media stream"

143. In my opinion, Alston discloses or renders obvious this limitation. This limitation includes substantially the same language as Limitation [1a], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that limitation. *See* Ground I, Limitation [1a], showing that SIP, ***the first protocol is used for*** an INVITE/200 OK exchange that is used to set-up a first media session from the first terminal, wherein the 200 OK message (*i.e.*, ***a response***) "routes to the mobile device and indicates that the subscriber has accepted the session invitation. The mobile device . . . then communicates the stream of data containing

[1] For clarity, the second protocol of claim 17 is the third protocol of claim 1. In this case, both are SIP.

any packetized data, such as "live" or real time video images, video clip(s), or image(s), and/or VoIP data." Ex.1004, [0034]. In other words, the responses ***cause the first terminal to begin transmission of the first media stream***. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

6. [17e]: "wherein the first media stream is between the first terminal and a media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal."

144. In my opinion, Alston discloses or renders obvious this limitation. This limitation is identical to Limitation [1e], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that limitation. *See* Ground I, Limitation [1e], showing that the first media stream is between the first terminal and a media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal.

O. Claim 18

1. [18pre]: "A gateway comprising"

145. In my opinion, Alston discloses this limitation. This limitation is identical to language in Limitations [1pre] and [1a], and is disclosed or rendered obvious by Alston for the reasons I discuss above for those limitations. *See* Limitations [1pre]-[1a].

2. [18a]: "a communication interface"

146. In my opinion, Alston discloses this limitation. In my opinion, a POSA would understand that Alston's gateway is a communication interface at least because it is able to communicate with the mobile device and media delivery device. *See* Limitations [1pre]-[1a] (discussion of "gateway").

3. [18b]: "memory that stores software instructions that, when executed by the gateway, cause the gateway to"

147. In my opinion, Alston discloses this limitation. In my opinion, Alston discloses this limitation because Alston discloses: "*the content sharing application* 22 resid[es] *in a computer system* 200 (*such as the residential broadband gateway 30*, the storage device 90, the network memory location 94, or the server 96). Ex.1004, [0049]. Alston further teaches the "content sharing application 22 . . . resid[es] in a memory subsystem 202" of the gateway. A POSA would understand from this disclosure that when the gateway (or its processor)

executes the software instructions it will cause the gateway to perform the steps in claims [18c]-[18f]. *See* Limitations [1pre]-[1a] (discussion of "gateway").

4. [18c]: "(i) exchange first multimedia session information with a first terminal using a first protocol, wherein the exchange of the first media session information sets up a first media stream from the first terminal,"

148. In my opinion, Alston discloses this limitation. This limitation includes substantially the same language as Limitation [1a], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that limitation. *See* Limitation [1a]. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

5. [18d]: "(ii) provide a trigger to a second terminal, wherein the trigger is provided using a second protocol, in response to which the second terminal is able to initiate the exchange of second multimedia session information with the gateway, and"

149. In my opinion, Alston discloses this limitation. This limitation includes substantially the same language as Limitation [1b], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that limitation. *See* Limitation [1b]. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

6. [18e]: "(iii) exchange second multimedia session information with the second terminal, using a third protocol, wherein the third protocol is different from the first protocol, wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates"

150. In my opinion, Alston discloses this limitation. This limitation includes substantially the same language as Limitation [1c], and is disclosed or rendered obvious by Alston for the reasons I discuss above for that limitation. *See* Limitation [1c]. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

7. [18f]: "wherein the gateway further comprises a media server, wherein the first media stream is between the first terminal and said media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal."

151. In my opinion, Alston alone discloses or in view of Fajardo renders obvious this limitation. This limitation includes substantially the same language as Limitations [1d] and [1e], and is disclosed or rendered obvious by Alston for the reasons I discuss above for those limitations. *See* Limitations [1d]-[1e]. To the extent there are any differences in the language of these limitations, I considered

these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

P. Claim 22: The gateway of claim 18, wherein the gateway is configured to associate the first multimedia session information and the second multimedia session information with each other.

152. In my opinion, Alston discloses this limitation. This limitation includes substantially the same language as Claim 16, and is disclosed or rendered obvious by Alston for the reasons I discuss above for that claim. *See* Claim 16. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

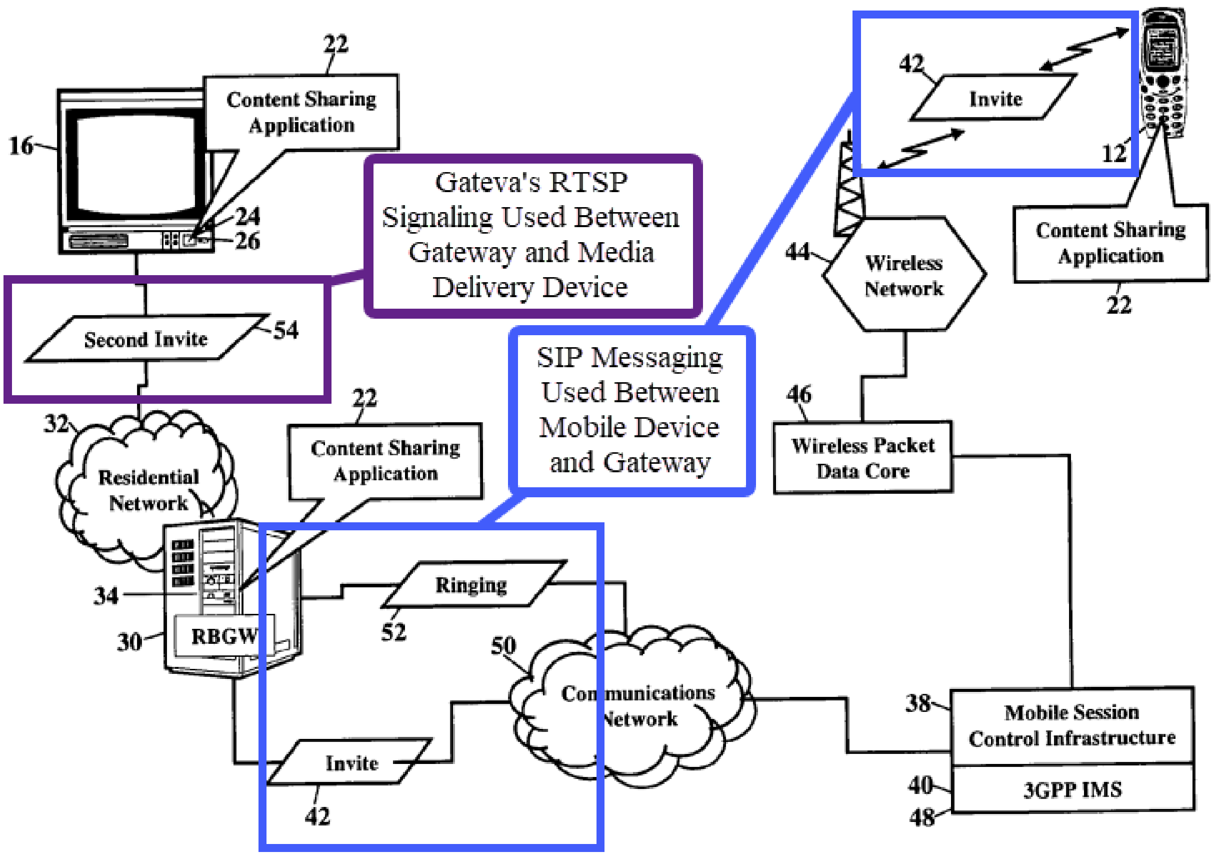
XI. GROUND II: CLAIMS 1-18, 22 ARE RENDERED OBVIOUS BY ALSTON AND FAJARDO IN VIEW OF GATEVA

153. I provide a summary of the grounds as follows: Alston teaches a gateway capable of establishing media sessions between two separate terminals in an IMS-adjacent architecture. Alston discloses that a variety of different protocols known to a POSA can be used to establish and maintain the media sessions. Alston discloses that the gateway is compatible with a variety of media delivery devices.

154. Gateva teaches a method of establishing a media session with an RTSP client wherein the media session is set-up with SIP and RTSP is used to maintain the session.

155. For the reasons described below, a POSA would have implemented Gateva's SIP-RTSP messaging scheme with Alston's teachings to Alston's communication system between the "media delivery device 16," "RBGW 32," and the "communications network 50," as annotated in Alston's Fig. 3:

FIG. 3

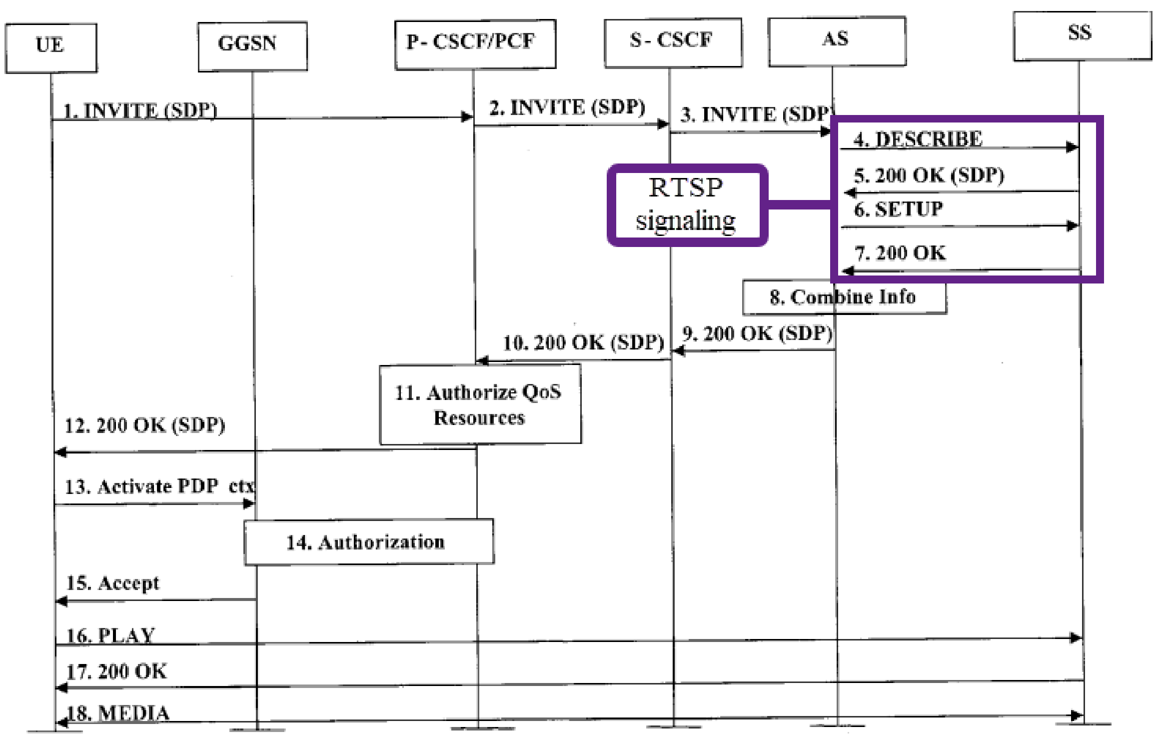


Ex.1004, Fig. 3 (highlighting added).

156. Specifically, Alston already discloses that "SIP" is a protocol that can be used between the gateway and communications networks. Further, both Gateva and Alston employ the same INVITE/200OK messaging scheme for gateway to communications network (*i.e.*, first terminal) communication. *Compare* Ex.1004

("Invite 42") *with* Ex.1007 ("3.INVITE"); *also compare* Ex.1004 ("confirmation message 68 [which] may be a SIP OK message") *with* Ex.1007 ("9.200 OK"). Alston further discloses that the protocol between the media delivery device (*i.e.*, second terminal) and the gateway can be "any messaging protocol." Ex.1004, [0028].

157. For the communication between the media delivery device and gateway, a POSA would be motivated to use RTSP as Gateva teaches, and as highlighted in Fig. 2:



Ex.1005.

158. As set forth below, it would have been obvious and simple for a POSA to apply Gateva's teachings to Alston.

A. Motivations to Combine

1. Alston Supports Any Messaging Protocol

159. As provided for in Alston, Alston's gateway can be used with any messaging protocol, preferably SIP (Ex.1004):

> [0028] The exemplary embodiments may utilize ***any messaging protocol***. The registration message 36, for example, may use the Session Initiation Protocol (SIP) or the H.323 protocol. ***As those of ordinary skill in the art understand, Session Initiation Protocol (SIP) is the Internet Engineering Task Force's standard for multimedia conferencing over IP.*** The H.323 protocol originated as an International Telecommunications Union (ITU) multimedia Standard and incorporates multiple protocols, including Q.931 for signaling, H.245 for negotiation, and Registration Admission and Status (RAS) for session control. Because SIP has gained greater industry acceptance, SIP will be shown and discussed, although ***those of ordinary skill in the art understand any messaging protocol may be used***.

160. Indeed, Alston repeatedly states ***any messaging protocol*** can be used with its teachings. Ex.1004, [0025], [0028], [0030], [0032]-[0033]. Based on the disclosures in Alston, I agree that any known messaging protocol at the time could be easily adapted for use with Alston's teaching.

161. As provided above, Alston teaches that its system could be implemented with any messaging protocol known to POSAs. Similar to SIP, at that time, RTSP was the Internet Engineering Task Force's standard for *controlling* multimedia streaming sessions over IP networks. Ex.1008, 1, 3-5. And like SIP, in

my opinion, RTSP and its corresponding RFC would be known to POSAs. As further described above and recognized by the IETF, RTSP is particularly suited when there is streaming media.

162. Alston further describes session setup, again in the preferred embodiment using SIP as an example:

> [0030] FIG. 3 is a schematic further illustrating the exemplary embodiments. The mobile user, using the mobile device 12, desires to share content with the subscriber's media delivery device 16. The mobile device 12 sends an invite message 42 to the subscriber. The invite message 42 is addressed to the subscriber, such as to the subscriber's home telephone number, home email address, or any other communications address associated with the subscriber. The invite 42, as earlier explained, may use any messaging protocol. ***Because SIP has gained greater industry acceptance, the invite message 42 will be shown as discussed as SIP invite message 42.*** The SIP invite message 42 first routes to the mobile user's home IMS network 40 via a wireless network 44 and via a wireless packet data core 46. As those of ordinary skill in the art understand, the wireless network 44 is a wireless access network of a communications service provider. The wireless network 44 includes all the network elements and equipment for the delivery of wireless services (e.g. receivers, base stations, base station controllers, transmission facilities, MSCs, HLRs, VLRs, etc.) (not shown for simplicity). The wireless network 44 may utilize any portion of the electromagnetic spectrum and any signaling standard (such as the I.E.E.E. 802 family of standards, GSM/CDMA/TDMA/WCDMA or any other cellular standard, and/or the ISM band). As those of ordinary skill in the art understand, the wireless packet data core 46 includes elements required to support wireless packet data services (such as 2.5G and/or 3G services, BLUETOOTH services, and/or the family of I.E.E.E. 802 services). These elements provide the interface between the wireless network and

> internal or external IP networks. The packet core may or may not support QoS mechanisms. Because both the wireless network 44 and the wireless packet data core 46 are known to those of ordinary skill, this patent will not provide a further explanation of either.

163. Therefore, in my opinion, a POSA reading Alston would be motivated to use existing, known protocols that were suitable to the tasks identified in Alston and demonstrated in the preferred embodiment using SIP. An additional alternative to SIP, especially suitable for streaming media, would be RTSP. As described in the RTSP RFC2326, it was particularly well-suited for control of streaming media sessions. Ex.1008, 5-6. Moreover, RTSP had messages that performed the same kinds of functions identified in Alston, such as initiating (SIP INVITE) and confirming(OK)/accepting (ACK).

164. The SIP INVITE message, which is used to initiate a session by inviting another party and providing details like codecs and media streams, is similar to an RTSP DESCRIBE. Ex.1008, 31-32. The RTSP DESCRIBE message is used to retrieve information about the media resource, such as the available streams and formats. *Id.* Thus the DESCRIBE message also accomplishes the purpose of initiating and preparing a session, similar to the SIP INVITE.

165. The SIP 200 OK message, which confirms the session has been established and provides the agreed session parameters, is very similar to an RTSP 200 OK message. Ex.1008, 31-32. The RTSP 200 OK message has a similar purpose

to the SIP 200 OK message as it both are sent in response to the initiation step to confirm parameters for the session, effectively confirming that the session is ready to proceed. Ex.1008, 31-32. This RTSP 200 OK message parallels the role of SIP 200 OK in confirming and initiating the session.

166. The SIP ACK message, which acknowledges the receipt of a 200 OK response and confirms that the session is established, is similar to the RTSP PLAY message. Ex.1008, 34-35. The RTSP PLAY message instructs the server to begin streaming media, confirming the client's readiness to proceed with the session. *Id.* Like the SIP ACK, it serves as the final step to confirm that the session is active and operational.

2. Suitability of RTSP

167. The suitability of RTSP is confirmed by disclosures in the background of the 330 patent describing the state of the art prior to the 330 patent. The 330 patent explains that many solutions to SIP-RTSP integration were known to those in the art including SIP-RTSP gateways. Ex.1001, 1:16-2:31. Beyond Alston, other prior art references describe the use of SIP and RTSP. *See, e.g., also* Ex.1005; Ex.1006.

168. As Alston contemplates, SIP is an exemplary protocol, but several other existing protocols could be used in an embodiment of Alston. This approach allows for flexibility in implementing a system based on Alston in that it can use a variety

of existing standards in known, compatible ways. The ability to use different protocols would mean that Alston could support a wider range of devices in a broad range of countries. Accordingly, a POSA would have recognized the advantages of the SIP-RTSP integration technique proposed by Gateva, and would have found it obvious to apply the improved technique to Alston's multimedia gateway, to yield predictable results. For example, in the IMS 3GPP architecture, the choice of protocol for session description—whether HTTP GET or RTSP DESCRIBE—depends heavily on the requirements of the streaming environment and the underlying network constraints. While RTSP DESCRIBE is often preferred for its ability to fetch precise, session-specific metadata in formats like SDP, there are situations where it becomes impractical or unfeasible. For example, in IMS architectures designed to support a wide range of devices and access networks, HTTP GET offers better compatibility with a wider range of systems. It seamlessly retrieves metadata from web-based streaming servers, even in environments where RTSP might be blocked by firewalls or incompatible with certain user devices. In these scenarios, relying on HTTP GET ensures the system remains functional and accessible without requiring network reconfiguration or additional protocol support on the client side.

169. Specifically, Alston as well as Gateva (and similar systems like Whitehead) demonstrated that SIP was a well-known choice for initiating

multimedia sessions, particularly in IMS/3GPP architectures where SIP's robust signaling framework facilitated efficient session setup. At the same time, RTSP was widely recognized as for managing dynamic multimedia control, offering advanced playback features like PLAY, PAUSE, and TEARDOWN. Ex.1008, 12-13. Gateva explicitly described these advantages, presenting "an improved method for controlling streaming services," and showed how the integration of SIP and RTSP addressed the well-known challenge of bridging session initiation with fine-grained multimedia control. Ex.1005, [0002]-[0010].

B. Reasonable Expectation of Success

170. A POSA would have had a reasonable expectation of success in making such modifications because they merely entail implementing one prior art solution with well-known techniques from prior art in the same field. Accordingly, a POSA would have understood that these references, as well as the POSA's knowledge, disclose interrelated teachings based on routine technologies and would have been amenable to various well-understood and predictable combinations. Indeed, Alston's explicitly provides that its exemplary embodiment uses SIP "[b]ecause SIP has gained greater industry acceptance, SIP will be shown and discussed, although those of ordinary skill in the art understand any messaging protocol may be used."

171. In my opinion, a POSA would have reasonably expected this combination to succeed and be operable in Alston's residential broadband gateway

based on the established functionality and flexibility of both SIP and RTSP, as well as the known capabilities of gateways to manage protocol translations in residential networks. Alston's residential broadband gateway was explicitly designed to handle diverse communication protocols for multimedia delivery, making it well-suited to incorporate a SIP-RTSP gateway. Such a modification would have been straightforward for a POSA, leveraging the existing architecture of Alston's gateway to bridge SIP's signaling strengths with RTSP's advanced multimedia management features. Furthermore, SIP was well-known in the art as able to set-up media streams in either direction.

172. In my opinion, residential broadband gateways, like Alston's, were built to facilitate a wide range of multimedia applications over IP networks, including Video on Demand, conferencing, and live streaming. These use cases required seamless communication between session signaling protocols like SIP and multimedia control protocols like RTSP. A POSA would have understood that Alston's gateway, with its flexible protocol handling capabilities, could readily accommodate a SIP-RTSP gateway by leveraging known techniques for translating SDP parameters, URIs, and multimedia session information between the two protocols. Given the established nature of SIP-RTSP gateways and the adaptability of Alston's residential broadband gateway, a POSA would have expected the

implementation of such a solution to be not only feasible but also predictable and efficient.

173. In addition, even though Alston's gateway system and Gateva's messaging scheme involve different directional implementations, with Gateva being downstream-focused, this distinction would not deter a POSA from combining their teachings. A POSA would recognize that the directionality of the protocols does not change the fundamental benefits of integration. Instead, combining Gateva's downstream technologies with Alston's gateway system would enable an architecture that incorporates the advantages of both references. Thus, a POSA would have been motivated to combine these Gateva and Alston to leverage the strengths of both approaches and achieve the predictable benefits described in the prior art.

174. Finally, Alston, Gateva, and the '330 patent are analogous art because all three are directed to the same field of multi-multimedia content delivery and IMS/3GPP systems. Ex.1001, 1:6-8 ("The invention relates to a method and a system for transmitting a multimedia stream from a first terminal to a second terminal"), 1:16-18("IP Multi-Multimedia Subsystem (IMS) defined by the 3GPP and 3GPP2 standards open the door to a whole range of new multimedia services"); Ex.1004, [0002] ("This application generally relates to communications and, more particularly, to sharing content in communications"), [0029] ("FIG. 2 illustrates the

3GPP (Third Generation Partnership Projects) IMS (EP Multimedia Subsystem) network 40. As those of ordinary skill in the art understand, the 3GPP has standardized the Universal Mobile Telecommunication System ("UMTS") to offer the next generation of wireless services. The IMS network 40 supports traditional telephony services as well as multimedia services."); Ex.1005, 8 ("Rationale for using SIP . . . in some networks like the Internet Multimedia Subsystem (IMS) it is now mandated to use SIP for session setup.").

C. Claim 1

1. [1pre]

175. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [1pre]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

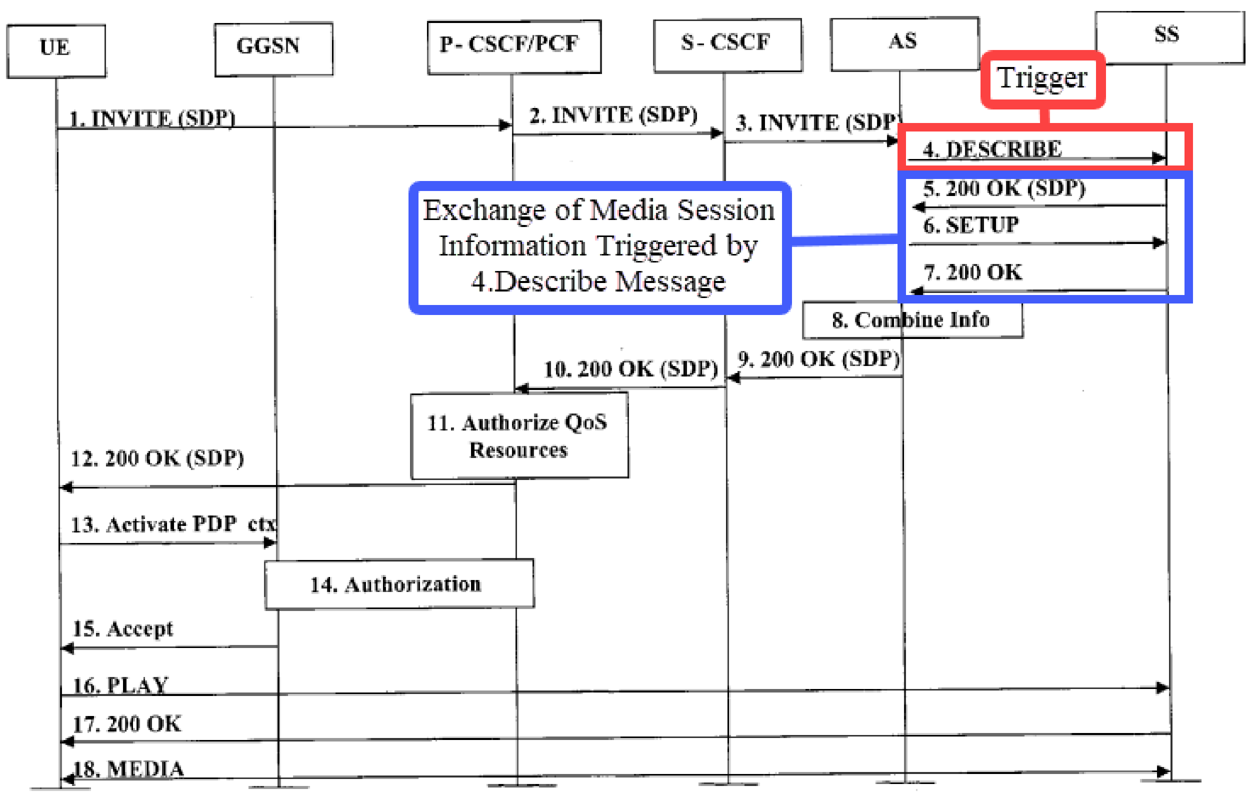
2. [1a]

176. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [1a]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

3. [1b]

177. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [1b]. To the extent Alston does not already disclose or render obvious this limitation, in my opinion the Alston-Gateva combination discloses or renders obvious this limitation.

178. Applying the RTSP messaging scheme to Alston renders this limitation obvious. In the proposed Alston-Gateva combination, the ***gateway uses a second protocol***, "RTSP" to send a "4.DESCRIBE" message, which acts as a ***trigger***, ***to the second terminal*** "to request a description of the streaming session," *i.e.*, ***initiate an exchange of second media session information between the gateway and the second terminal***. Ex.1005, Fig. 4, [0051]. As illustrated in Gateva's Fig. 2 below, the "4.DESCRIBE" message triggers sending of "5.200 OK," "6.SETUP," and "7.200 OK" messages (Ex.1005, Fig. 2, [0046]-[0064]):



179. Each of the highlighted messages in Gateva Fig. 2 contain session parameters or SDP, *i.e.*, media session information about the stream that is being

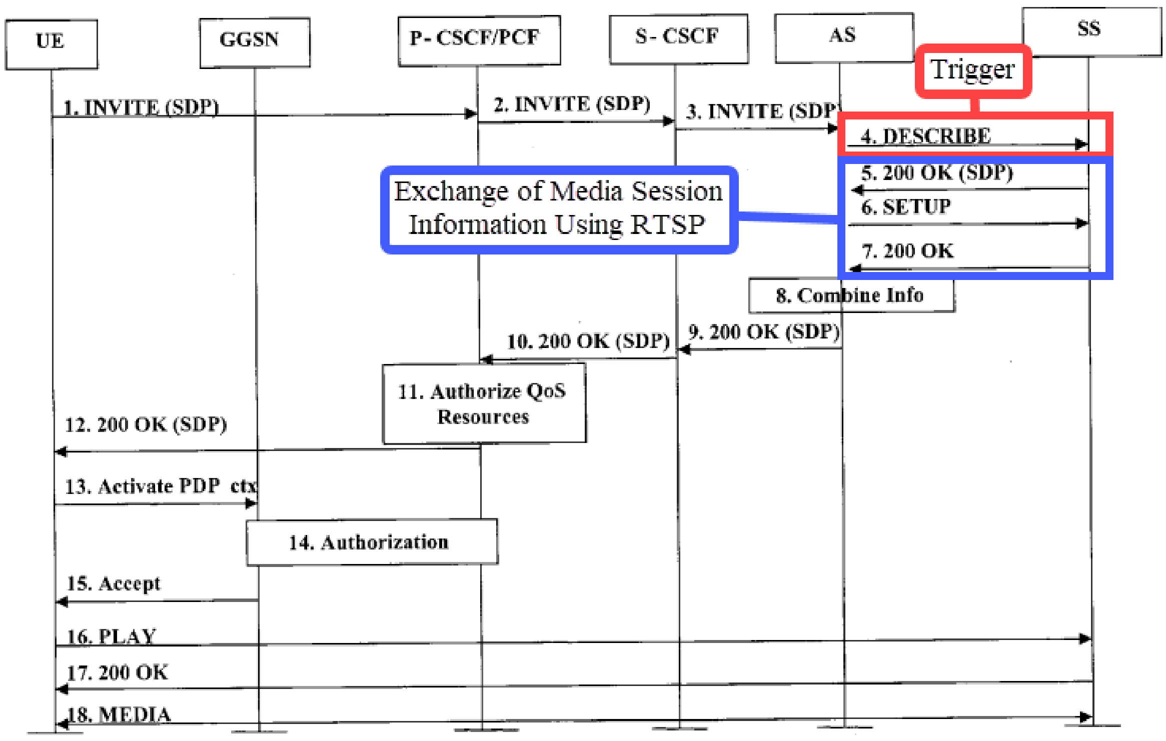
setup. *Id*. For example, a POSA would know from the standard at the time that a "SETUP message . . . specifies the transport mechanism to be used for the streamed media" and the "OK" messages sent in response to the describe and set-up messages also contain media session information such as the "content-type" and "content-length." Ex.1008, 31, 33.

4. [1c]

180. Alston alone discloses or renders obvious this limitation. *See* Ground I, [1c]. To the extent Alston does not already disclose or render obvious this limitation, in my opinion, the Alston-Gateva combination discloses or renders obvious this limitation as Gateva's RTSP messaging scheme applied to Alston's gateway renders this limitation obvious.

181. With regard to the limitation "in response to receiving the trigger, the second terminal using a third protocol to initiate the exchange of the second media session information between the gateway and the second terminal," in the proposed Alston-Gateva combination, Gateva's messaging scheme ***in response to receiving the trigger***, *i.e.*, "4.DESCRIBE" message, Gateva teaches ***using a third protocol to initiate the exchange of the second media session information between the gateway and the second terminal*** in the form of the "5.200 OK," "6.SETUP," and "7.200 OK" messages, each of which a POSA would understand to contain SDP,

i.e., media session information about the stream that is being setup as provided in Ground II, [1b]:



182. With regard to the limitation "wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates," in my opinion, Alston alone discloses or in view of Gateva renders obvious this limitation. *See* Ground I, [1c].

5. [1d]

183. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [1d]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

6. [1e]

184. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [1e]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

D. Claim 2

185. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [2]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

E. Claim 3

186. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [3]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

F. Claim 4

187. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [4]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

188. The Alston and Gateva combination also discloses or renders obvious this limitation as both references teach the same "INVITE" and "acceptance"/"OK" messaging sequence when sending messages from the first terminal to the gateway to exchange media session information. C*ompare* Ex.1005, Fig.2 ("3.Invite" and

"9.200 OK") *with* Ex.1004 Fig.3 ("Invite 42"), Fig.5 ("Confirmation 68"). Therefore, Alston alone, and also in light of Gateva, teaches this limitation.

G. Claim 5: "The method according to claim 4, further comprising: before using the second protocol to provide the trigger to the second terminal, the gateway resolving the identifier of the second terminal into a network address of the second terminal."

189. In my opinion, Alston alone discloses or in view of Gateva renders obvious this limitation.

190. The Alston-Gateva combination teaches that the "4.DESCRIBE" message serves as the trigger. *See also* Ground II, [4]. A "4.DESCRIBE" RTSP message contains a IP address (*i.e.*, ***a network address***) of both the sender and the receiver of the RTSP message. This was common knowledge to a POSA at the time because it is what the cellular standard required. Ex.1008, 30-32. Accordingly, the application server of Gateva (which is substituted for the gateway of Alston in the combination) must be able to take the identifier provided in the "Invite" message and resolve it into a network address of the second terminal when translating between protocols. Indeed, at the time, SIP messages often used domain names as an identifier rather than direct IP addresses (e.g., sip:bob@example.com), requiring proxy resolution to map the domain name to the network address of the receiver.

H. Claim 6: "The method according to claim 5, wherein providing the trigger to the second terminal comprises the gateway transmitting the trigger to the resolved network address of the second terminal."

191. In my opinion, Alston alone discloses or in view of Gateva renders obvious this limitation.

192. The Alston-Gateva combination discloses or otherwise teaches that when providing the gateway to the second terminal, the gateway transmits the trigger to the resolved network address of the second terminal. *See* Ground II, [4, 5].

I. Claim 7: "The method of claim 1, wherein the third protocol is different from the second protocol."

193. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [7]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

194. Furthermore, the Alston-Gateva combination teaches that the "4.DESCRIBE" message that serves as the trigger can alternatively be a "HTTP Get message to request a description of the streaming session." Ex.1005, [0050]. In other words, HTTP protocol can be the second protocol. Gateva further uses RTSP as the third protocol, *i.e.*, "200 OK." "6.SETUP," and "200 OK" messages used to exchange second media session information. Ex.1005, Fig. 2. Accordingly, Gateva's teachings applied to Alston's gateway render obvious ***the third protocol is different from the second protocol***. In my opinion, POSA would recognize that using different protocols, such as HTTP and RTSP, is a well-understood design

choice in IMS and streaming architectures, where specific protocols are selected to optimize different aspects of the communication flow. HTTP is often preferred for metadata retrieval due to its ability to function in environments with strict network constraints, while RTSP excels in managing real-time media streams and session control. This complementary usage ensures that the system can efficiently handle both metadata exchange and media streaming, making the combination of these protocols a natural extension of Alston's and Gateva's teachings.

195. I understand KPN may argue that Gateva or any other art teaches away from the use of HTTP protocol in conjunction with RTSP and SIP, for a POSA the choice between HTTP and RTSP for the trigger would have been a simple decision based on the RTSP system at hand. In the IMS 3GPP architecture, the choice of protocol for session description—whether HTTP GET or RTSP DESCRIBE—depends heavily on the requirements of the streaming environment and the underlying network constraints. While RTSP DESCRIBE is often preferred for its ability to fetch precise, session-specific metadata in formats like SDP, there are situations where it becomes impractical or unfeasible. For example, in IMS architectures designed to support a wide range of devices and access networks, HTTP GET offers better compatibility with a wider range of systems. It seamlessly retrieves metadata from web-based streaming servers, even in environments where RTSP might be blocked by firewalls or incompatible with certain user devices. In

these scenarios, relying on HTTP GET ensures the system remains functional and accessible without requiring network reconfiguration or additional protocol support on the client side. In my opinion, a POSA would recognize that this adaptability is critical in IMS systems designed to handle heterogeneous environments. The ability to alternate between HTTP GET and RTSP DESCRIBE based on network and device constraints reflects a practical and well-understood approach to session management, ensuring that the choice of protocol aligns with the system's requirements without fundamentally altering its functionality or purpose.

J. Claim 8

196. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [8]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

K. Claim 9

197. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [9]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

L. Claim 10

198. In my opinion, Alston alone discloses orrenders obvious this claim. *See* Ground I, [10]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

M. Claim 11: "The method of claim 1, wherein the third protocol is a client-server protocol, and wherein the second receiving terminal has client capabilities of the third protocol and not server capabilities of the third protocol."

199. In my opinion, Alston alone discloses or in view of Gateva renders obvious this limitation.

200. The Alston-Gateva combination discloses or otherwise teaches that ***the third protocol is RTSP which is a client-server protocol***. *See* Ground I, [1]. RTSP, as defined in the standards at the time, is explicitly designed to operate in a client-server model, enabling a client device to control streaming media sessions by sending commands like PLAY, PAUSE, DESCRIBE, and SETUP to a server. This client-server interaction is a fundamental characteristic of RTSP, ensuring precise control over the media stream from the client side while the server manages media delivery. Further, a POSA would understand that the receiving terminal could be a number of different types of terminals known in the art, many of which do have client capabilities of the third protocol and not server capabilities of the third protocol. For example, any terminal used in the combination would have ***the client capabilities of the third protocol*** because it must be able to receive messages in that language to consume and control the stream. While a POSA would understand that the second terminal *could* have the server capabilities of the third protocol, it would not need to. And further, a POSA would realize, in many applications, it would be advantageous to be able to stream media to terminals that don't have server

capabilities of the third protocol, RTSP, *e.g.*, set-top boxes. In my opinion, a POSA would recognize that RTSP is designed to facilitate client-server interactions, where the client only requires the capability to send commands (e.g., PLAY, PAUSE, SETUP) and interpret responses from the server to consume and control the media stream. This client-focused design is particularly suited for devices like set-top boxes, which are intended to receive and play media without the added complexity or resource requirements of serving content to other devices. Furthermore, enabling the use of terminals with only client capabilities ensures broader compatibility and simplifies device design, making it feasible to deploy streaming systems across a wide range of consumer electronics without requiring server functionality on every receiving terminal. This approach reduces cost and technical barriers, allowing for widespread adoption in practical, real-world applications.

N. Claims 12-16

201. In my opinion, Alston alone discloses or renders obvious these claims. *See* Ground I, [12]-[16]. Therefore, Alston in view of Gateva discloses or renders obvious these claims.

O. Claim 17

1. [17pre]

202. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [17pre]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

2. [17a]

203. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [17a]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

3. [17b]

204. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [17b]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

4. [17c]

205. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [17c]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

5. [17d]

206. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [17d]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

6. [17e]

207. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [17e]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

P. Claim 18

1. [18pre]/[18a]: "A gateway comprising: / a communication interface"

208. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [18pre]/[18a]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

2. [18b]: "memory that stores software instructions that, when executed by the gateway, cause the gateway to"

209. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [18b]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

3. [18c]: "(i) exchange first multimedia session information with a first terminal using a first protocol, wherein the exchange of the first media session information sets up a first media stream from the first terminal,"

210. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [18c]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

4. [18d]: "(ii) provide a trigger to a second terminal, wherein the trigger is provided using a second protocol, in response to which the second terminal is able to initiate the exchange of second multimedia session information with the gateway, and"

211. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [18d]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

5. [18e]: "(iii) exchange second multimedia session information with the second terminal, using a third protocol, wherein the third protocol is different from the first protocol, wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates"

212. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [18e]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

6. [18f]: "wherein the gateway further comprises a media server, wherein the first media stream is between the first terminal and said media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal."

213. In my opinion, Alston alone discloses or renders obvious this limitation. *See* Ground I, [18f]. Therefore, Alston in view of Gateva discloses or renders obvious this limitation.

Q. Claim 22: The gateway of claim 18, wherein the gateway is configured to associate the first multimedia session information and the second multimedia session information with each other.

214. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [22]. Therefore, Alston in view of Gateva discloses or renders obvious this claim.

XII. GROUND III: CLAIMS 1-5, 7-10, 12-20, AND 22-23 ARE RENDERED OBVIOUS BY MELA ALONE, OR IN LIGHT OF ALSTON AND/OR FAJARDO

215. I provide a summary of the ground as follows:

216. Ground III relies on the teachings of Mela in combination with the common knowledge of a POSA regarding commonly used and well-known protocols and communication systems, as alternatively evidenced by Alston and/or Fajardo. As set out below, Mela describes a messaging scheme that works through a gateway. Mela specifically describes the content of each RTSP and SIP message that it sends. The contents of these messages lay out the session descriptions that are exchanged for setting up a media stream. To the extent Mela is silent on buffering, modifying Mela to include buffering, a well-known technique in the art, was obvious as the benefits of buffering were wildly known for, *inter alia*, controlling the quality of a media stream and reducing jitteriness. Indeed, as further described below, most practical application of Mela would demand buffering.

A. Motivation to Combine Mela and Alston and/or Fajardo

217. In my opinion, a POSA would have been motivated to incorporate a media server with buffering functionality into a system using an RTSP client, such as Alston's, as this approach addressed well-recognized challenges in multimedia content delivery, such as jitter, latency, and packet loss. Adding a media server with buffering capabilities to such systems would enhance the reliability and quality of media delivery. Media servers could buffer incoming streams by temporarily storing portions of the media in memory before transmitting them to the client, thereby compensating for network variability and ensuring uninterrupted playback. This architecture not only optimized end-user experience but also reduced the computational load on RTSP clients, making it particularly attractive for systems requiring high scalability or supporting multiple simultaneous users.

218. In my opinion, the integration of a media server for buffering aligns with the broader goals of improving multimedia streaming quality in RTSP-based systems. Media servers with buffering capabilities were a natural extension of the RTSP architecture, allowing the RTSP client to focus on stream control while offloading buffering and stream stabilization tasks to a centralized server. This design reduced redundancy, improved system efficiency, and enabled advanced features such as trick-play functionality (pause, seek, fast-forward), which required partial storage of media streams during delivery. A POSA would have recognized

this enhancement as a predictable solution to known challenges in multimedia systems. Alston's teachings on leveraging RTSP for streaming session management, combined with Mela's acknowledgment of the importance of robust media control, provide clear motivation to introduce a media server to buffer media streams, yielding a reliable, scalable, and efficient architecture for delivering high-quality multimedia content. Ex.1004, [0025], [0028].

219. In my opinion, a POSA would have understood that adding a media server with buffering to systems employing RTSP clients would improve the system's performance in predictable ways. The use of a media server to buffer incoming streams was a well-known technique in the art, and Alston's emphasis on improving streaming session quality through buffering further supports this modification. This addition would ensure smooth streaming quality while simplifying the system architecture by leveraging routine and well-established technologies. Moreover, a POSA would have had a reasonable expectation of success in implementing such a solution, as it involved applying a known architectural improvement—buffering media streams via a centralized server—to an existing RTSP-based system.

220. In my opinion, Mela and Alston reflect a shared understanding of the challenges and solutions in multimedia systems, particularly in environments requiring robust media streaming where buffering media at a centralized server was

both a cost-effective and technically straightforward improvement. Mela explicitly contemplates the role of RTSP clients in managing media sessions, and Alston highlights the advantages of using buffering to stabilize media streams. Adding a media server to perform buffering was thus a logical and predictable enhancement to address variability in media stream delivery, enabling a more robust and user-friendly multimedia experience. Ex.1004, [0002], [0029]. Such a combination would have been obvious to try, given the finite number of predictable solutions for ensuring stream stability, and would align with the practical goals of system efficiency, reliability, and scalability.

221. Regarding Claims [7], [14], and [15], in my opinion, a POSA would have been motivated to generalize Mela's system for use with a broader range of protocols—this includes using the same protocol to communicate between the gateway and both terminals (e.g., SIP) and using different protocols to "trigger" the second terminal and exchange media session information, in view of Alston. Alston discloses that "any messaging protocol" may be used to "establish a session" or serve as the trigger. EX1004, [0025], [0028], [0030], [0032]. A POSA would be motivated to apply this in Mela to increase flexibility and usable applications, such as for use with the many different "media delivery device[s]" described in Alston. This broader protocol applicability would expand the system's utility across diverse communication scenarios, improving adaptability for various network architectures,

allowing for Mela's integration with existing technologies. A POSA would have reasonably expected to succeed because the combination merely requires applying and implementing well-known protocols in a conventional communication system. In my opinion, such implementation leverages existing engineering practices and does not introduce significant technical challenges, as the necessary protocols and techniques are already well-documented and widely understood in the field.

222. Regarding Claim [9], in my opinion, a POSA would have been motivated to modify Mela's system to use its trigger for the additional purpose of informing the second terminal of the availability of a new media stream, in view of Alston. Alston discloses that its trigger causes the second terminal to "provide an [] indication or alert of the … invitation to share video." EX1004, [0039]. A POSA would be motivated to apply this in Mela to enhance user experience. By incorporating this functionality, the system would provide users with notifications before setting up the stream, allowing for responses to or terminations of incoming media-sharing requests. This would be beneficial in scenarios where timely acknowledgment and interaction are critical, such as video conferencing, live streaming, or collaborative work environments. A POSA would have reasonably expected to succeed because the combination only requires the routine modification of the protocols already in use in Mela (by adding additional session description information to the SIP and RTSP messages) to provide additional information

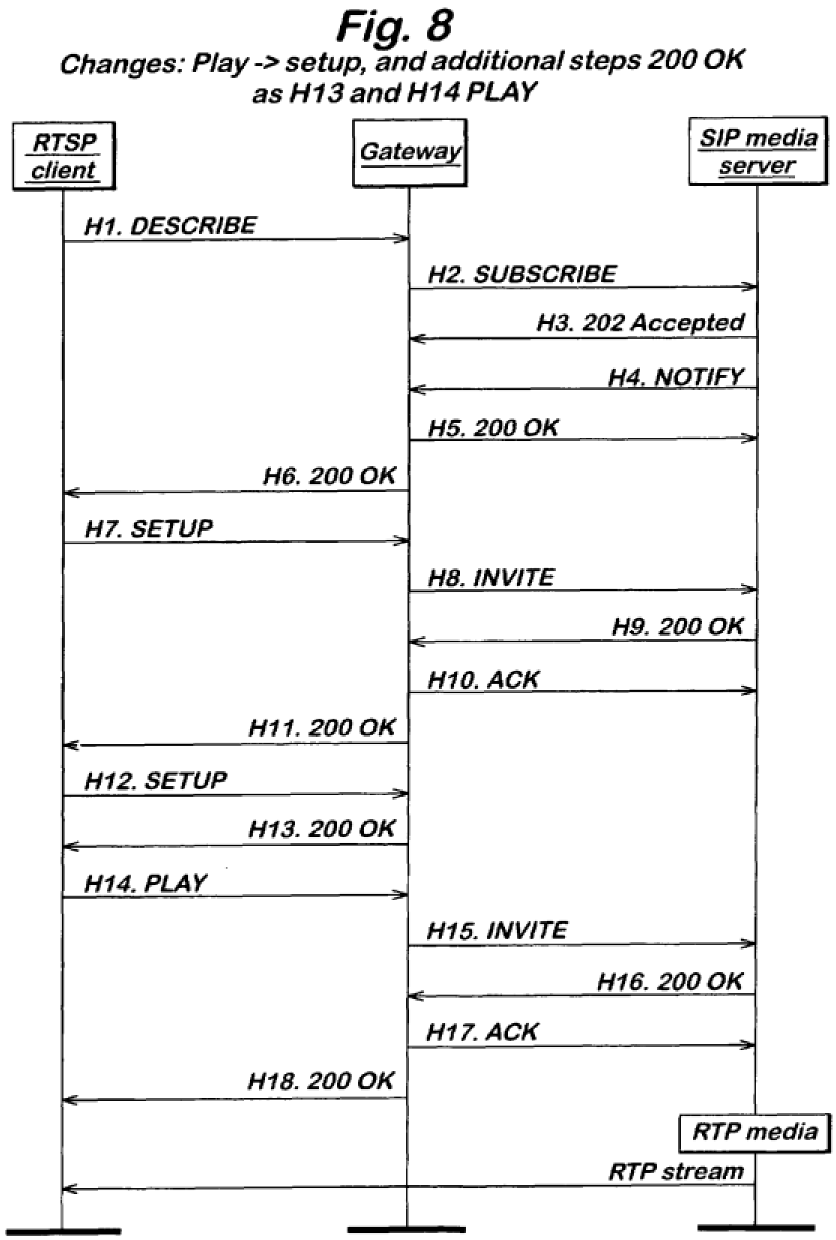
(availability of a stream) that can be used by the end terminal to provide information on the availability of a new media stream, as was well known in the art.

B. Claim 1

1. [1pre]

223. In my opinion, Mela alone discloses or renders obvious this limitation.

224. To the extent the preamble is limiting, in my opinion, Mela teaches "***a gateway for use in a communications system***, wherein said gateway is arranged to establish a SIP session [*i.e.*, ***communicate***] with one entity [*i.e.*, ***a first terminal***] and a RTSP session [*i.e.*, ***communicate***] with a further entity [*i.e.*, ***a second terminal***]." Ex.1007, 2:63-3:2. The ***gateway*** (also referred to by Mela as the "second entity") is further described as "being arranged to map between SIP and RSTP information to allow data to pass [*i.e.*, ***communicate***] from one of said first and third entities [*i.e.*, the ***first terminal*** and ***second terminal***] to the other of said first and third entities." Ex.1007, 2:46-53. Fig. 8 is an exemplary embodiment of the teachings of Mela wherein the "<u>SIP media server</u>" is the first terminal and the "<u>RTSP client</u>" is the second terminal:



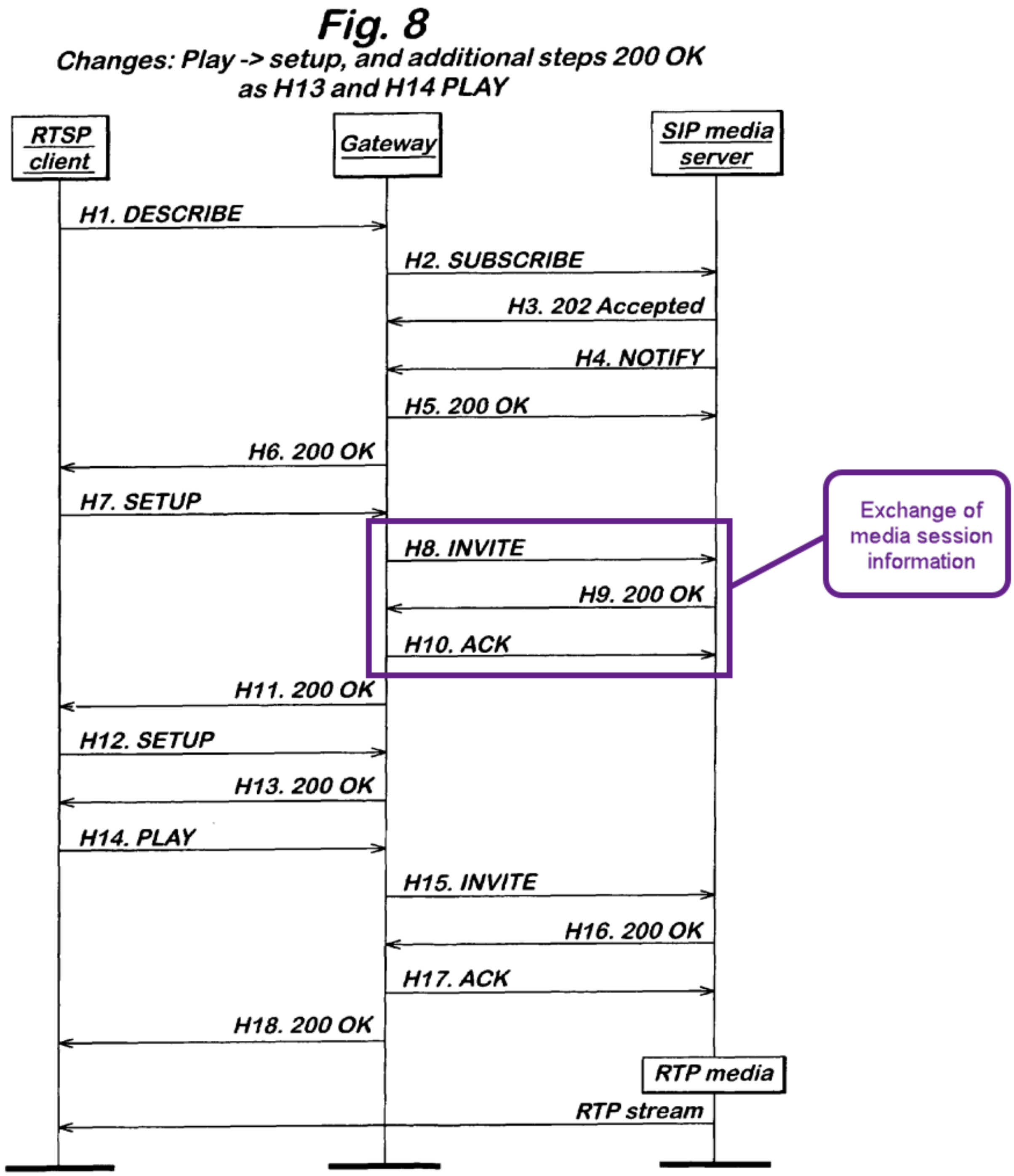
Ex.1007, Fig.8.

2. [1a]

225. In my opinion, Mela alone discloses or renders obvious this limitation.

226. With regard to the limitation "the gateway or the first terminal using a first protocol to exchange first media session information between the first terminal

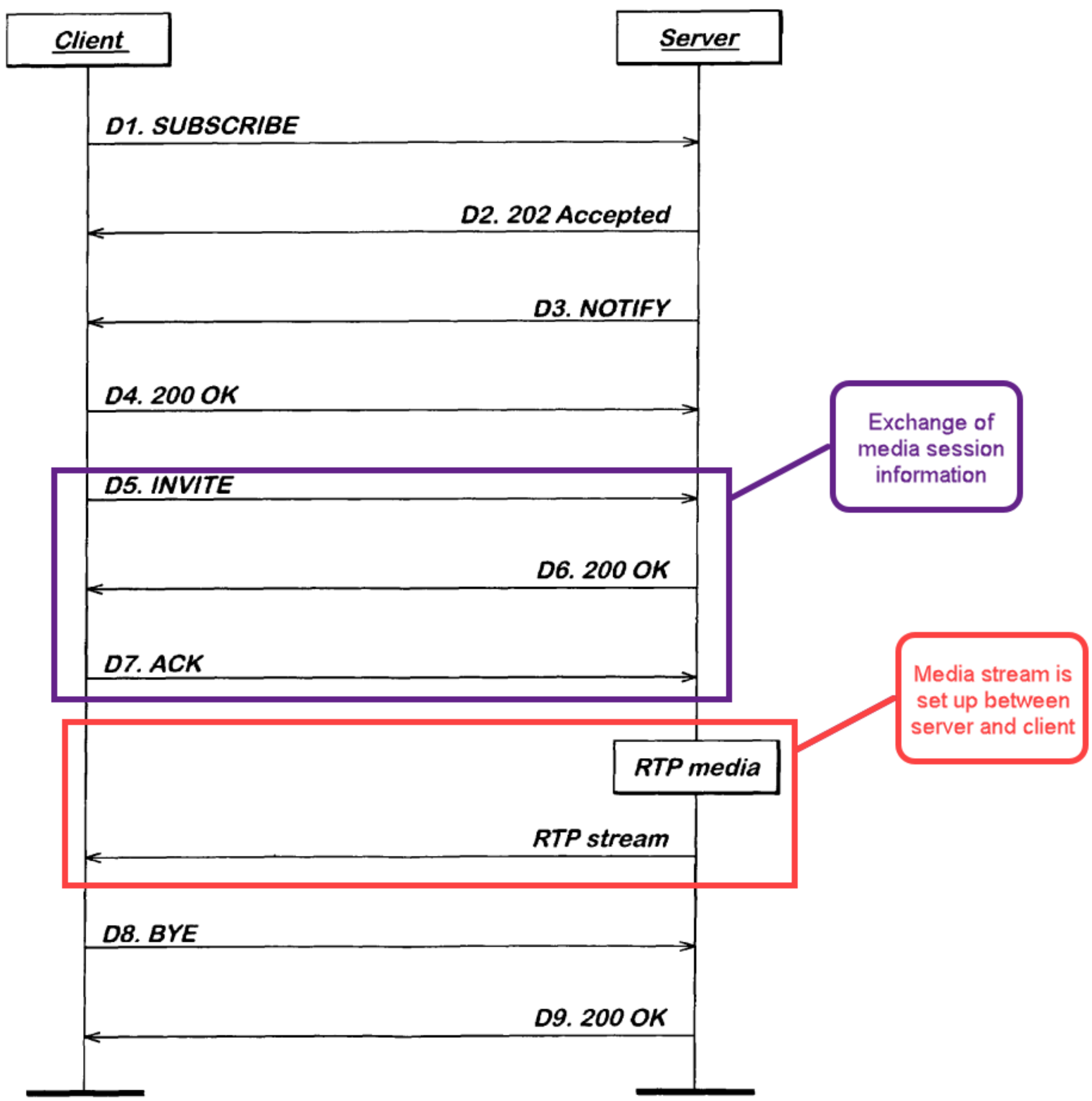
and the gateway," Fig. 9 of Mela (Ex.1007) shows the ***first terminal*** (the SIP media server) ***using a first protocol***, "SIP," ***to exchange first media session information between the first terminal and the gateway*** with the use of SIP "INVITE," "200 OK," and "ACK" messages (steps H8 to H10), annotated below:



Mela discloses that the SIP "INVITE," "200 OK," and "ACK" messages include media session information, such as "the network address of the client" and "media description" including "media type" and "transport port." EX1007, 7:54-9:45, 21:62-22:41; *see also* EX1001, 10:38-42 ("multimedia session information may include … IP addresses, port numbers for the RTP streams, the type of media (voice, audio, video, etc.) and codec information."). In my opinion, a POSA would understand from these disclosures that this is an exchange first media session information between the first terminal and the gateway. Further, a POSA would recognize that the inclusion of media session parameters, such as "content-length," is essential for describing the structure and details of the media stream, allowing the gateway to interpret and process the session correctly. This exchange establishes the necessary communication framework, ensuring both parties agree on critical aspects of the session, such as payload size, media format, and transport protocol. Additionally, Mela's explicit depiction of this exchange in Fig. 8 reinforces its role in initiating and configuring the first media session, aligning with standard practices in SIP-based multimedia systems.

227. With regard to the limitation "wherein the exchange of the first media session information sets up a first media stream from the first terminal," Mela discloses that after the SIP "INVITE," "200 OK," and "ACK" messages (*i.e.*, the ***first media session information***) are ***exchanged***, an RTP ***media stream*** (*i.e.*, ***first media stream***) is set up from the SIP media server (*i.e.*, the ***first terminal***). For example, in Fig. 4, Mela illustrates how the SIP messages are used to set up an RTP media stream between a server and a client, as annotated below.

Fig. 4

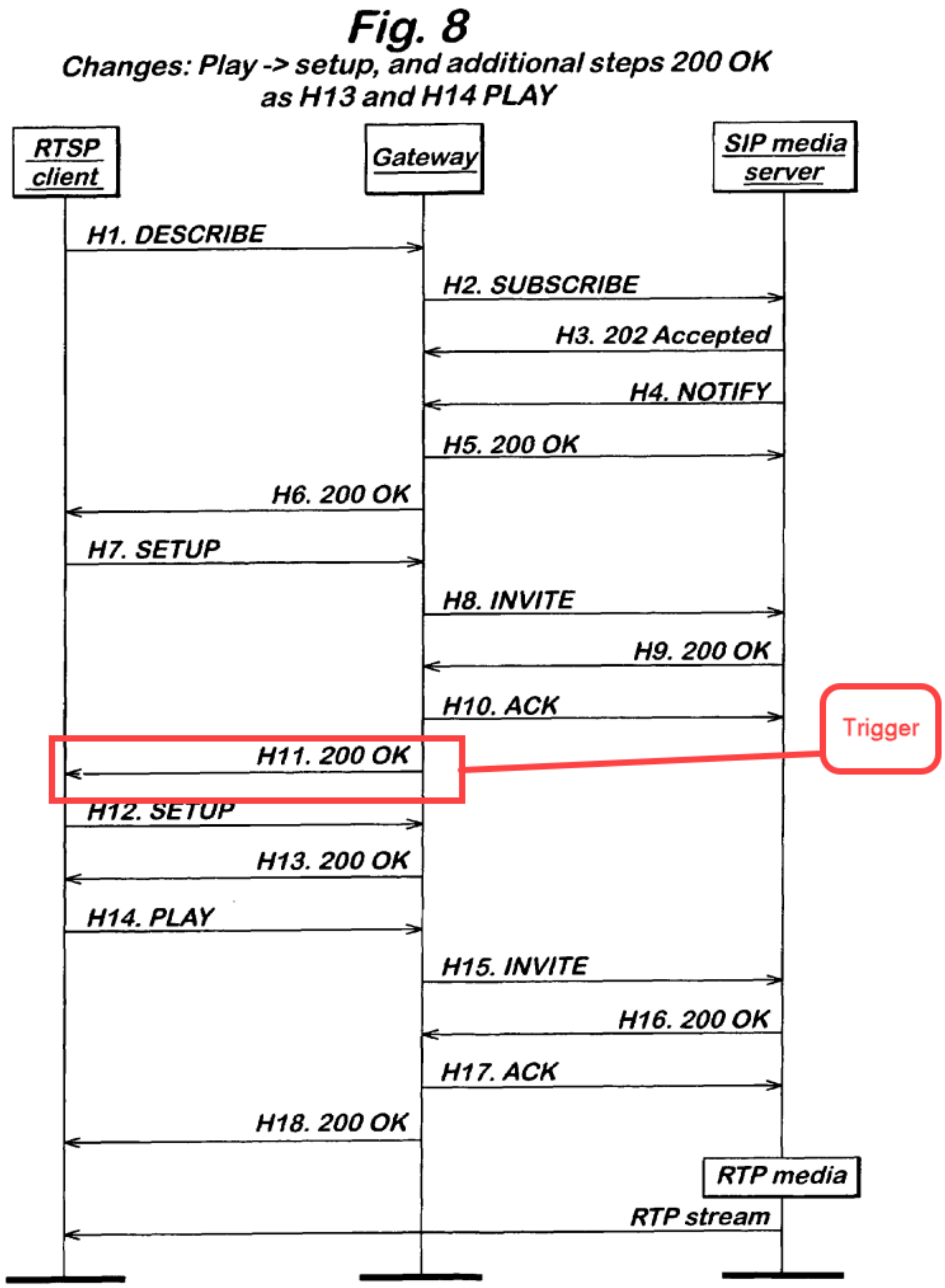


EX1007, Fig. 4, 7:50-9:46. Mela further explains that for Fig. 8, "[c]ommunication between the gateway and the SIP media server is similar to previous examples, so that steps from H8 to H10 are similar to steps D5 to D7 in FIG. 4." EX1007, 21:64-67. A POSA would thus understand that the SIP "INVITE," "200 OK," and "ACK" messages between the SIP media server and the gateway in Fig. 8 similarly set up a first media stream between the SIP media server and the gateway. Ex.1007, Fig. 8-9, 22:1-41; *see also* Ex.1007, Fig. 2-3, 3:36-63.

3. [1b]

228. In my opinion, Mela alone discloses or renders obvious this limitation.

229. With regard to the limitation "the second terminal using a second protocol to receive a trigger provided by the gateway," as shown in Fig. 8, Mela discloses that after the gateway exchanges the SIP "ACK" message, the gateway sends an RTSP "200 OK message" to the RTSP client:



230. In other words, the RTSP client (*i.e.*, ***the second terminal***) receives a "200 OK" message from the gateway (*i.e.*, ***receives a trigger provided by the gateway***), using RTSP (*i.e.*, a ***second protocol***). With regard to the limitation

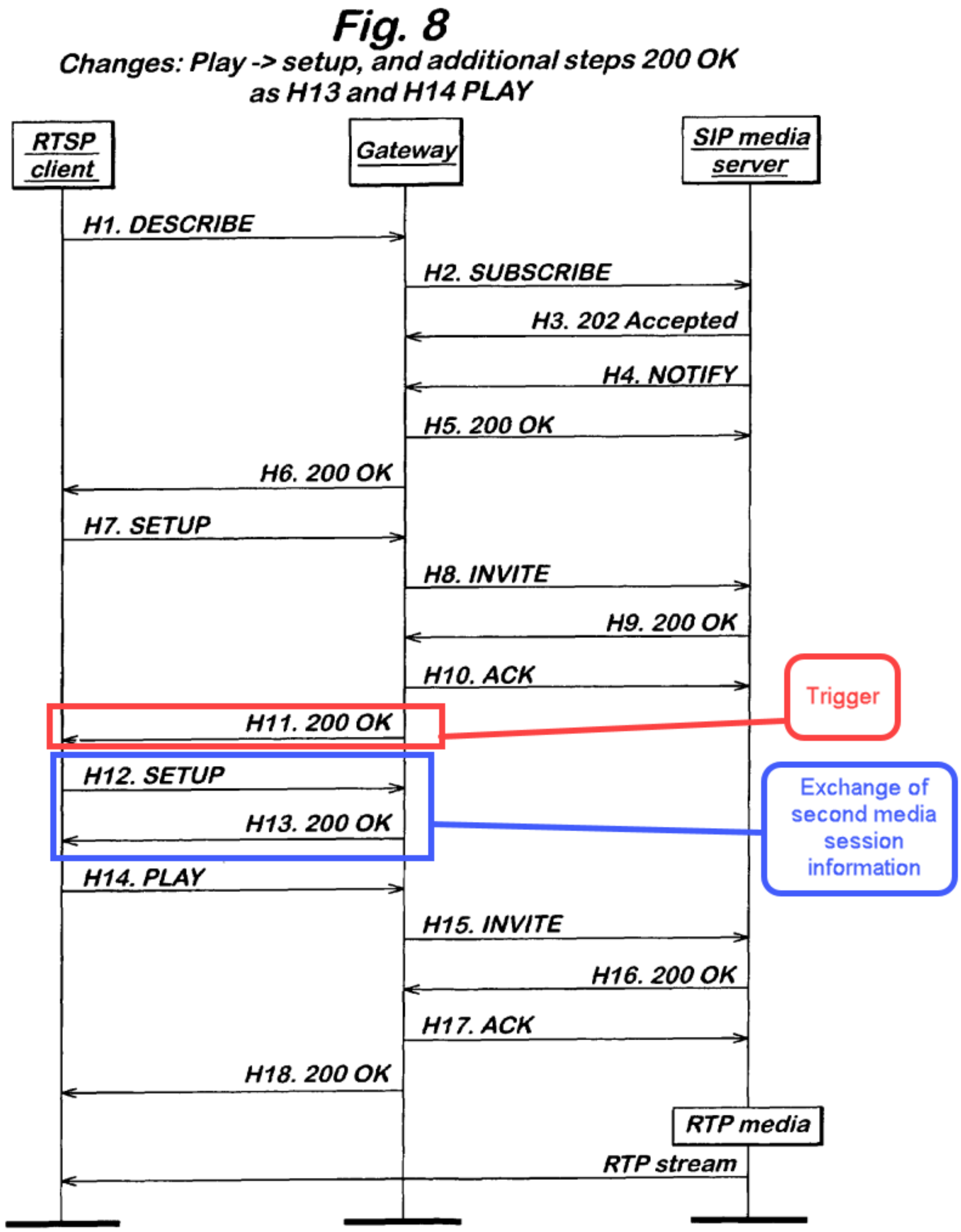
"wherein the trigger is to initiate an exchange of second media session information between the gateway and the second terminal": The RTSP "200 OK" message is a ***trigger*** because it causes the RTSP client to "set[] up" a media session. EX1007, 22:43-57. The set up is ***initiated*** by sending an RTSP "SETUP" message from the RTSP client (*i.e.*, the ***second terminal***) to the ***gateway***, as shown in step H12 of Fig. 8. *Id.*, 22:56-65. Mela discloses that ***second media session information*** is ***exchanged*** between the RTSP client and the gateway via the RTSP "SETUP" message (step H12) and the responsive RTSP "200 OK" message (step H13), which contain a source IP address (*e.g.*, message H13 "source" field) and transport port numbers (*e.g.*, message H12 and message H13 "port" fields). *Id.*, 22:60-23:13. In my opinion, a POSA would understand from these disclosures that the "200 OK" message is a trigger, as it *causes* the exchange of session information by *causing* the H.12 SETUP and H.13 200 OK messages to be delivered. Without the 200 OK message, the H.12 SETUP and H.13 200 OK messages would not be sent. Therefore, in my opinion, the "200 OK" message is a trigger.

4. [1c]

231. In my opinion, Mela alone discloses or renders obvious this limitation.

232. With regard to the limitation "in response to receiving the trigger, the second terminal using a third protocol to initiate the exchange of the second media session information between the gateway and the second terminal": As explained in

Limitation [1b], Mela teaches that in response to the RSTP "200 OK" message at step H11 of Fig. 8 (*i.e.*, the ***trigger***), the RTSP client (*i.e.*, ***second terminal***) sends a "SETUP" message to the ***gateway*** (step H12) using RTSP (*i.e.*, a ***third protocol***), which ***initiates the exchange of second media session information*** via the RTSP "SETUP" and "200 OK" messages (steps H12 and H13). *See* Limitation [1b]; Ex.1007, 22:43-23:13, Fig. 8-9, Fig. 2-3, 3:36-63.



233. With regard to the limitation "wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates," Mela discloses that the RTSP "SETUP" and "200 OK"

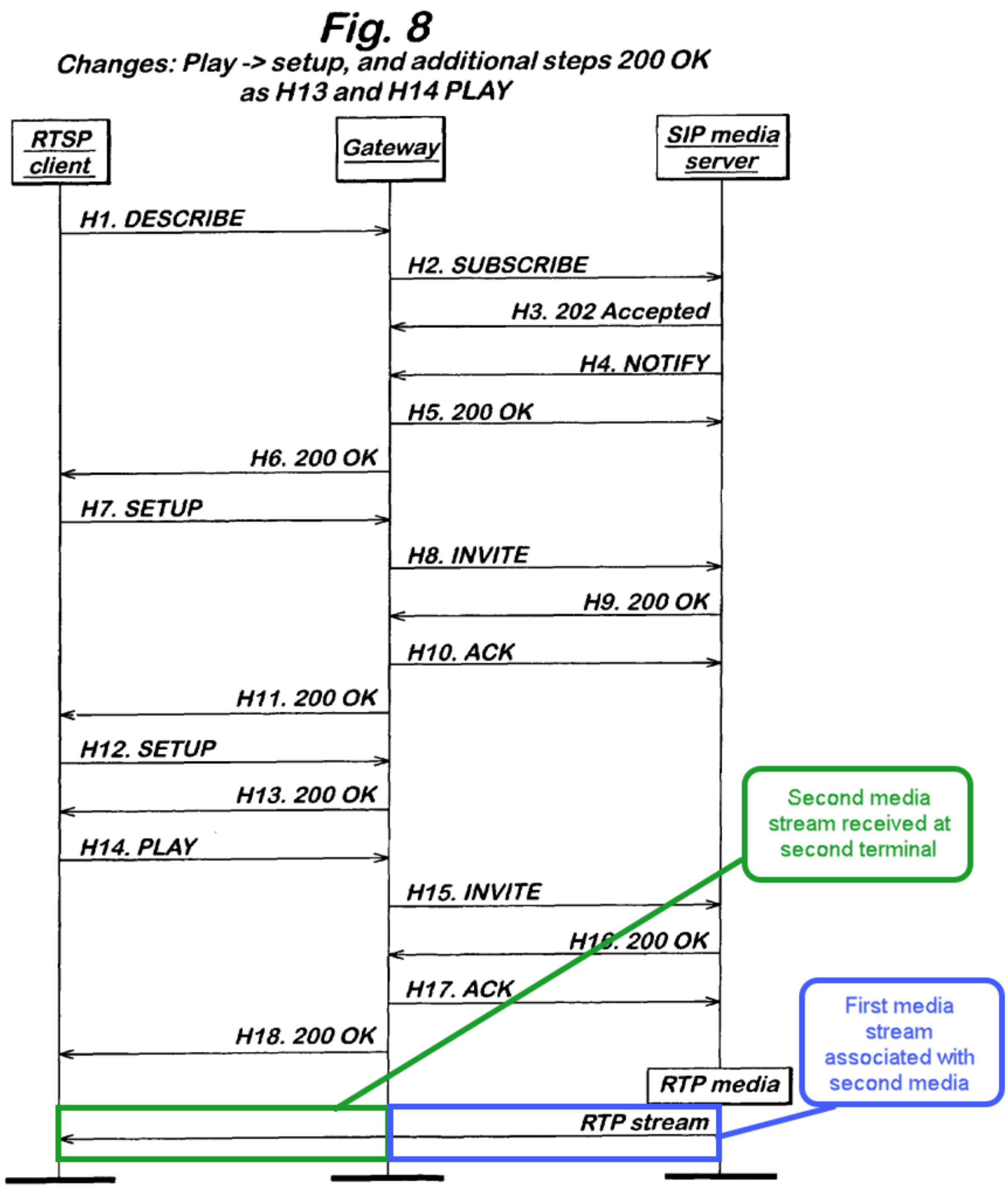
messages (***second media session information***) ***exchanged*** in steps H12-H13 of Fig. 8 are used to "set up" a media session (*e.g.*, a video stream) (***second media stream***). Ex.1007, 22:43-57. A POSA would understand that the resulting media stream is set up between the ***gateway*** and the RTSP client (***second terminal***) because the RTSP messaging occurs only between the gateway and the RTSP client. Mela teaches that the RTSP Client ***participates in the stream*** because it can control the stream, *e.g.*, by pausing it. Ex.1007, 24:25-28. In my opinion, a POSA would further recognize that such participation requires the exchange of detailed session information to establish the parameters of the stream, including timing, media format, and transport protocols. This exchange ensures the second terminal has sufficient information to request and control the stream effectively, aligning with standard RTSP functionalities. By allowing actions like pausing and fast-forwarding, the RTSP Client demonstrates its integration into the media stream. In addition, it is my opinion that based on the teachings of Mela that Mela discloses that the RTP stream illustrated in Fig. 8 travels through the gateway before reaching the RTSP client. This picture discloses this to a POSA, and even if the picture does not evidence this, a POSA would understand that such a modification would be obvious and well within the skill of the art. Indeed, Mela teaches using a gateway as a conduit for the RTP stream at multiple points in its disclosure. Ex.1007, Fig. 8-9, Fig. 2-3, 3:36-63. To be clear, in Fig. 8, the RTP stream between the gateway and the SIP media server is the first media stream and the RTP stream between the gateway and the RTSP client is the second media stream. To the extent not otherwise disclosed by Mela, Ground III, [1e], provides that there would be two separate streams on either side of the media server/gateway. In addition, it is my opinion that based on the teachings of Mela that Mela discloses that the RTP stream illustrated in Fig. 8 travels through the gateway before reaching the RTSP client. This picture discloses this to a POSA, and even if the picture does not evidence this, a POSA would understand that such a modification would be obvious and well within the skill of the art. Indeed, Mela teaches using a gateway as a conduit for the RTP stream at multiple points in its disclosure. Ex.1007, Fig. 8-9, Fig. 2-3, 3:36-63. To be clear, in Fig.8, the RTP stream between the gateway and the SIP media server is the first media stream and the RTP stream between the gateway and the RTSP client is the second media stream. To the extent not otherwise disclosed by Mela, Ground III, [1e], provides that there would be two separate streams on either side of the media server/gateway.

5. [1d]

234. In my opinion, Mela alone discloses or renders obvious this limitation.

235. Mela discloses the RTSP client (***second terminal***) ***receiving the second media stream,*** as illustrated in Fig. 8 (Ex. 1007, 24:24-28; *infra* [1e]). Further, the second media stream is ***associated with the first media stream*** because the first

media stream delivers the media from the SIP server to the gateway, as discussed in Limitation [1a] and as also illustrated below. Ex.1007, Fig. 2-3, 8-9, 3:36-63, 22:1-41.



In my opinion, a POSA would recognize that the association between the first and second media streams is established through the signaling and session parameters exchanged during the setup process. *See also* [1e]. This linkage ensures that the second media stream corresponds to the original content transmitted in the first media stream, maintaining continuity and logical coherence. In RTSP-based systems, such as the one described in Mela, the SIP Media Server acts as the intermediary, facilitating the transition and association between the two streams to enable seamless delivery of the media to the RTSP Client. This functionality is critical for applications where content integrity and synchronization across sessions are required, such as video-on-demand or live streaming scenarios.

6. [1e]

236. In my opinion, Mela alone discloses or, optionally in view of Alston and Fajardo, renders obvious this limitation.

237. Mela discloses that ***the first media stream is between*** the SIP media server (***first terminal***) and the gateway, *see* Limitation [1a], and ***the second media stream is between*** the RTSP client (***second terminal***) and the gateway, *see* Limitation [1d].

It would have been obvious to a POSA to implement Mela's gateway as or including a ***media server***. Mela discloses "a gateway for use in a communications system," wherein the communication system is capable of "communication of voice, data, multimedia and so on" and "operates in accordance with a given standard or specification." Ex.1007, 1:9-31; 2:63-3:2. A POSA would have understood that Mela's teachings may be used in well-known, standard communication systems, which, as of the date of the '151 patent, included the IP Multi-Media Subsystem (IMS) network architecture (used and made widely available by Verizon and AT&T).

238. A foundational aspect of these systems was the use of buffering techniques to manage media streams effectively, addressing challenges such as jitter, packet loss, and latency. This buffering process was universally implemented using physical memory, specifically RAM, which was well-suited for storing and processing incoming media streams in real time. RTSP clients, such as Apple's QuickTime Player and Microsoft's Windows Media Player, relied on RAM to temporarily hold portions of media data as it was received. By doing so, these clients ensured that packets arriving at irregular intervals could be reordered and processed smoothly, providing uninterrupted playback to the end user.

239. In my opinion, the mechanism was straightforward: as a media stream was received, the RTSP client allocated a portion of the system's RAM as a buffer. Incoming packets of the media stream were written into this buffer, where they were temporarily stored before being decoded and played back. This process allowed the client to absorb variations in network transmission, ensuring that any gaps or delays

in packet delivery did not disrupt playback. Similarly, Video-on-Demand (VoD) services used RAM in their media servers to buffer streams received from a first terminal before transmitting them to a second terminal. X. This was particularly critical for features like pausing, seeking, and fast-forwarding, which required the system to temporarily hold parts of the media stream in memory for on-demand access. A POSA would recognize that this use of RAM to implement buffering was not only a practical necessity but also a widely established design approach, making it clear that the limitation of buffering media streams before transmission was well-known in the art. Furthermore, this buffering necessitated an association between the first and second media streams, as the server required a mechanism to map the incoming stream to its corresponding outgoing stream. Without such an association, the system would not be able to correctly link the content from the first terminal with the playback requested by the second terminal, undermining critical functionalities like seeking or pausing in VoD systems. This association ensured seamless transmission and logical continuity between the two streams.

240. It was well known by those of skill in the art that other buffering techniques were also common at the time, for example a widely implemented technique in RTSP-based systems and IMS architectures involved the use of separate media servers to handle buffering between terminals or clients. These servers acted as intermediaries, receiving an incoming media stream, temporarily storing it, and

then transmitting a processed or reformatted version of the stream to its destination. This approach was particularly advantageous in large-scale systems where multiple streams needed to be managed concurrently. By centralizing the buffering function on a server, the system could ensure steady transmission of media to clients, even when the originating terminal or the network exhibited variability in performance. This intermediary buffering also allowed servers to perform additional processing, such as packet reordering, transcoding, or error correction, which were critical for maintaining a high-quality user experience.

241. In Video-on-Demand (VoD) services and IMS systems, the media server's buffering capabilities were often combined with advanced features to optimize delivery. For example, media servers frequently preloaded a portion of the stream before beginning transmission, creating a buffer that allowed clients to access trick-play functions like pausing, seeking, or fast-forwarding without interruption. These techniques were also integral in managing multiple concurrent streams by aligning the transmission rates of incoming and outgoing streams, ensuring no client experienced disruptions due to network variability or load balancing issues. A POSA would readily recognize this server-based buffering as a standard and effective method for addressing the challenges of real-time media delivery, particularly in environments requiring high scalability and reliability.

242. This understanding is consistent with Mela's express disclosures. Mela teaches that an SIP "client" can "buffer the stream and avoid interrupts in the playback" using the "speed" parameter in SIP messages. EX1007, 27:48-50, 28:61-67. In the embodiment of Fig. 8, the SIP "client" is the gateway in communication with the SIP media server. A POSA would thus understand that the gateway may function as a media server to buffer streams between the SIP media server and the RTSP client. *Id.*

243. To the extent KPN argues that Mela does not teach this limitation or otherwise render it obvious, a POSA would have turned to the many teachings in the art, such as Alston's media "server 96," to solve the buffering problem that would be an obvious addition to any of Mela's practical uses.

244. Additionally, as I discussed in Ground I, limitation [1e], Fajardo expressly teaches many of the above known benefits regarding media buffering, including its ability to mitigate loss of connectivity. EX1019, 4:48-59, 7:55-8:11. For all of the same reasons discussed above, it would have been obvious to incorporate such teachings in Mela.

C. Claim 2

245. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation. As discussed in Limitation [1a], Mela's gateway exchanges ***first media session information*** with the SIP media server (***first terminal***)

in steps H8-H10 of Fig. 8. The first media session information includes a network identifier of the RTSP client (the ***second terminal***), which is ***part of the second media session information*** exchanged in steps H12-H13 (*see* Limitations [1b]-[1c]).

246. Specifically, Mela discloses that the gateway "interprets" messages from the RTSP client "to a form that the SIP media server can understand." EX1007, 21:60-67. The SIP "INVITE" message ***provided from the gateway*** to the SIP media server in step H8 includes, as part of the media session information, an SIP URI/URL, *i.e.*, "sip:***twister@media.nokia.com***;user=stream." EX1007, 22:5. The same URI/URL is provided, in RTSP form, as part of the second media session information in step H12, *i.e.*, "rtsp://***media***-gw.***nokia.com***/***twister***/video." *Id.*, 22:62

D. Claim 3

247. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

248. As discussed in Limitations [1b]-[1c], the ***gateway provides to the second terminal*** a "200 OK" message in step H12 of Fig. 8 ***as part of the exchange of the second media session information***. The "200 OK" message comprises the network address of the source, *i.e.*, "source=131.228.54.140" (EX1007, 23:12), which is also exchanged as ***part of the first media session information*** at step H9, *i.e.*, "c=IN IP4 131.228.54.140" (*id.*, 22:22).

E. Claim 4

249. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

250. As discussed in Limitation [1a], Mela's gateway exchanges ***first media session information*** with the SIP media server (***first terminal***) in steps H8-H10 of Fig. 8. Specifically, in step H9, ***the gateway receives*** a "200 OK" message from the SIP media server containing media session information. EX1007, Fig. 8. The "200 OK" message includes ***an identifier of the*** RTSP client (***second terminal***), specifically, "user@nokia.com." *Id.*, 22:14; *see also* EX1007, 20:52-60 (initial "SUBSCRIBE" message at step H2 identifying RTSP client as "user@nokia.com").

F. Claim 5

251. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

252. Mela discloses "***[b]efore using the second protocol to provide the trigger***"—*i.e.,* the "200 OK" message in step H11—***to the*** RTSP client (***second terminal***), ***the gateway resolves the identifier of the second terminal into a network address of the second terminal***. Specifically, Mela teaches that "[t]he connection 'c' field indicates the network address of the client," *i.e.,* the RTSP client in Fig. 8. EX1007, 8:26-30. Mela further teaches that at step H10 (*i.e.,* before the trigger in step H11), the gateway sends an "ACK" message that includes the "c" field and

identifies the RTSP client's network address, *i.e.*, "c=IN IP4 172.21.164.126." EX1007, 22:38. A POSA would have understood that the gateway resolved the RTSP client's network address based on the identifier previously exchanged (*e.g.*, at step H9, as described in Claim 4).

G. Claim 7

253. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [7]. Therefore, Mela alone or in view of Alston discloses or renders obvious this claim.

H. Claim 8

254. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation. In my opinion, as discussed in Limitation [1d] and [1e], Mela discloses the ***first terminal transmits the first media stream*** to the gateway and the ***second terminal receives the second media stream***. *See* Limitation [1d] and [1e]

I. Claim 9

255. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [9]. Therefore, Mela alone or in view of Alston discloses or renders obvious this claim.

J. Claim 10

256. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

257. As discussed in Limitation [1a], Mela's ***first protocol*** is SIP. EX1007, 21:60-22:40. The '151 patent explains that SIP is a ***peer to peer*** protocol. Ex.1001, 2:16-17 ("for example a peer-to-peer type of protocol, such as the SIP").

K. Claim 12

258. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

259. Mela discloses that ***the trigger***, *i.e.*, the RTSP "200 OK" message at step H11, ***comprises a network address of the gateway***. Specifically, Mela provides that the contents of the "200 OK" message include "source=131.228.54.140," *i.e.*, the network address gateway. EX1007, 22:48-54.

L. Claim 13

260. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

261. In my opinion, Mela discloses this limitation because a POSA would understand that ***the network address of the gateway***, "source=131.228.54.140," is a URI. *See also* Ground I, Claim 13.

M. Claim 14

262. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [14]. Therefore, Mela alone or in view of Alston discloses or renders obvious this claim.

N. Claim 15

263. In my opinion, Alston alone discloses or renders obvious this claim. *See* Ground I, [15]. Therefore, Mela alone or in view of Alston discloses or renders obvious this claim.

O. Claim 16

264. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation. In my opinion, as discussed above in Limitation [1e], Mela discloses or render obvious this limitation because the gateway must be configured to associate the first and second media streams. Since the specific content that these streams support are based on the multimedia session information (*e.g.* content type, aspect ratio), the gateway also associates the first and second multimedia session information.

P. Claim 17

1. [17pre]

265. Mela teaches a "gateway," which is ***an article of manufacture***. Mela further renders obvious this limitation because a POSA reading Mela's teachings would know that a gateway is a computing device that would have a computer-readable medium for storing software instructions that when executed by the gateway's processor, cause the gateway to perform the action's taught in Mela. This requirement is obvious because the gateway would need to execute software

instructions to function and those software instructions would need to be stored on a memory.

2. [17a]

266. In my opinion, Mela alone or in view of Alston renders obvious this limitation. This limitation includes substantially the same language as Limitation [1a], and is disclosed or rendered obvious by Mela alone or in view of Alston for the reasons I discuss above for that limitation. *See* Ground III, [1a]. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

3. [17b]

267. In my opinion, Mela alone or in view of Alston renders obvious this limitation. This limitation includes substantially the same language as Limitation [1b], and is disclosed or rendered obvious by Mela alone or in view of Alston for the reasons I discuss above for that limitation. *See* Ground III, [1b]. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

4. [17c]

268. In my opinion, Mela alone or in view of Alston renders obvious this limitation. This limitation includes substantially the same language as Limitation

[1c], and is disclosed or rendered obvious by Mela alone or in view of Alston for the reasons I discuss above for that limitation. *See* Ground III, [1c]. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

5. [17d]

269. In my opinion, Mela alone or in view of Alston renders obvious this limitation. This limitation includes substantially the same language as Limitation [1a], and is disclosed or rendered obvious by Mela alone or in view of Alston for the reasons I discuss above for that limitation. *See* Ground III, [1d]. To the extent there are any differences in the language of the two limitations, I considered these differences in my analysis, but ultimately relied on the same evidence and reached the same conclusions with respect to both limitations.

6. [17e]

270. In my opinion, Mela alone or in view of Alston and Fajardo renders obvious this limitation. This limitation is identical to Limitation [1e], and is disclosed or rendered obvious by Mela in view of Alston and Fajardo for the reasons I discuss above for that limitation. *See* Ground III, [1e].

Q. Claim 18

1. [18pre]: "A gateway comprising"

271. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

272. Mela teaches a "gateway." Ex.1007, 1:9-11, 20:21-32; *see also* Limitation [1pre].

2. [18a]: "a communication interface"

273. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation. Mela discloses "[a] communication system is a facility that enables communication between two or more entities such as user equipment and/or other nodes associated with the system"—*i.e.*, a communication interface. Ex.1007, 1:9-11.

3. [18b]: "memory that stores software instructions that, when executed by the gateway, cause the gateway to"

274. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

275. Mela renders obvious this limitation because a POSA reading Mela's teachings would know that the gateway must have a memory for storing software instructions that when executed by the gateway, cause the gateway to perform the action's taught in Mela. This requirement is obvious because the gateway would

need to execute software instructions to function and those software instructions would need to be stored on a memory. *See also* Limitation [1pre].

4. [18c]: "(i) exchange first multimedia session information with a first terminal using a first protocol, wherein the exchange of the first media session information sets up a first media stream from the first terminal,"

276. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

277. As provided above in Limitation [1a], Mela's gateway uses "H8.INVITE" and "H9.200 OK" messages to ***exchange first multimedia session information with a first terminal using a first protocol, wherein the exchange of the first media session information sets up a first media stream from the firstterminal***. Ex.1007, Fig. 4, Fig. 8, 7:45-9:35; 21:59-22:42.

5. [18d]: "(ii) provide a trigger to a second terminal, wherein the trigger is provided using a second protocol, in response to which the second terminal is able to initiate the exchange of second multimedia session information with the gateway, and"

278. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

279. As provided above in Limitation [1b], Mela's gateway uses the "H6.200 OK" message to ***provide a trigger to a second terminal, wherein the trigger is provided using a second protocol, in response to which the second***

terminal is able to initiate the exchange of second multimedia session information with the gateway. Ex.1007, 21:30-44.

6. [18e]: "(iii) exchange second multimedia session information with the second terminal, using a third protocol, wherein the third protocol is different from the first protocol, wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates"

280. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

281. As provided above in Limitation [1c], Mela's gateway uses "H12.SETUP" and "H13.200 OK" message to ***exchange second multimedia session information with the second terminal, using a third protocol, wherein the third protocol is different from the first protocol, wherein the exchange of the second media session information sets up a second media stream in which the second terminal participates***. Ex.1007, 22:60-23:17.

7. [18f]: "wherein the gateway further comprises a media server, wherein the first media stream is between the first terminal and said media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal."

282. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

283. As provided above in Limitation [1d], a POSA would find it obvious to adapt Mela using the teachings of Alston by adding ***a media server, wherein the first media stream is between the first terminal and said media server, wherein the second media stream is between said media server and the second terminal, and wherein said media server buffers at least part of the first media stream received from the first terminal before transmitting the second media stream to the second terminal***. Ex.1007, 22:60-23:17.

R. Claim 19

1. [19a]: "The gateway of claim 18, wherein the gateway is configured to connect to a first terminal supporting a first protocol and to a second terminal supporting a second protocol"

284. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

285. Mela discloses that ***the gateway is configured to connect to*** an SIP media server (***a first terminal supporting a first protocol***) and an RTSP client (***aa second terminal supporting a second protocol***). Ex.1007, Fig. 3, Fig. 8, 3:51-63. Mela also teaches an "aspect of the present invention [is] said gateway is arranged to establish a SIP session with one entity and a RTSP session with a further entity." *Id.*, 2:63-3:2.

2. [19b]: "wherein the first protocol is Session Initiation Protocol (SIP) and the second protocol is Real Time Streaming Protocol (RTSP)"

286. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

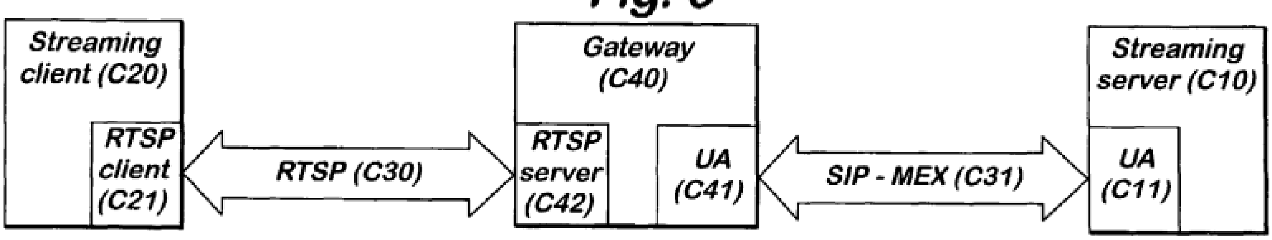
287. Mela discloses that the first protocol is "SIP" and the second protocol is "RTSP." *See* Limitation [19a]; Ex.1007, Fig. 3, Fig. 8, 3:51-63, 20:51-60; 22:4-40.

3. [19c]: "wherein the first terminal is an originating SIP client of a user at a user side"

288. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

289. Mela teaches that the "user application C11 of the streaming server (C10)" is an ***originating SIP client***. Fig. 3 illustrates "a gateway C40 between a RTSP domain C20 and a SIP media server C10, where the stream is fetched from the SIP Media Server C10 to RTSP domain C20." Ex.1007, Fig. 3, 3:51-54.

Fig. 3



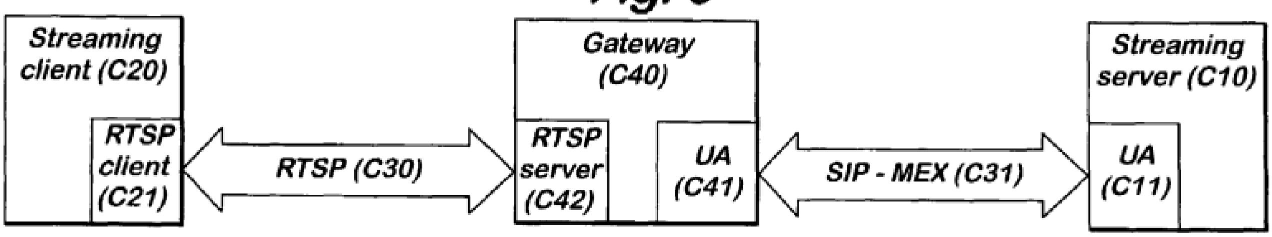
Mela's gateway "handles the mapping of the streaming media ***originating from the user application of the streaming server*** C11 to the RTSP server C42 so that it is compatible with the RTSP connection C30." Ex.1007, 3:37-50. In other words, the streaming media ***originates*** at a ***user application*** of the SIP media server (***first terminal***).

4. [19d]: "wherein the gateway acts as a terminating SIP client"

290. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

291. As provided above, the first terminal is an originating SIP client, and Fig. 3 illustrates, that the SIP connection terminates at the gateway, *i.e.*, ***the gateway acts as a terminating SIP client***:

Fig. 3



Ex.1007.

5. [19e]: "wherein the gateway is configured to confirm that the gateway is able to receive a multimedia session from the user and to provide the first terminal with multimedia session information"

292. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

293. Mela discloses that at step H5 of Fig. 8, a "200 OK" message is sent to the SIP media server. Ex.1007, Fig. 8, 21:59-22:45. The "200 OK" message is used

to confirm that "[the gateway] has received parameters for the streaming connection" and includes a "status code of ... with reason phrase 'OK' [which] indicates success," *i.e.*, ***the gateway is configured to confirm that the gateway is able to receive a multimedia session from the user***. Ex.1007, 7:32-35, 10:11-13. The "200 OK" message also includes information about the multimedia session, such as the recipient (sip:user@nokia.com) and the media server (sip:movies/twiser@media.nokia.com;user=stream) (Ex.1007, 21:5-30); thus, ***the gateway also provides the first terminal with multimedia session information*** in the same message.

6. [19f]: "wherein the second terminal is an RTSP client"

294. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

295. Mela discloses the second terminal is an "RTSP Client." Ex.1007, Figs. 3 & 8, 3:25-28, 3:51-64.

7. [19g]: "wherein the gateway acts as an RTSP server on behalf of the user"

296. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

297. Mela discloses that the Gateway includes an "RTSP server C42" that communicates with the RTSP client. Ex.1007, Fig. 3, Fig. 8, 3:25-28, 3:51-64.

8. [19h]: "wherein the second terminal uses RTSP for session control, and"

298. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

299. A POSA would understand that Mela's Fig. 3 provides that ***the second terminal uses the RTSP stream for session control*** as the purpose of the invention is to allow said second terminal to use "RTSP[,] a standard protocol for controlling multimedia streaming on Internet." Ex.1007, 1:32-45. Mela also teaches that the RTSP client can use RTSP messages to control the multimedia session, *e.g.*, by sending an RTSP "PAUSE" message to pause a stream. Ex.1007, 24:25-38.

9. [19i]: "wherein the gateway is configured to inform the second terminal that an incoming session is pending and notify the RTSP client to connect to an RTSP server function of the gateway."

300. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

301. Mela teaches the ***gateway is configured to*** send a "200 OK" message, *e.g.* at step H6 of Fig. 8, ***to inform the second terminal that an incoming session is pending and notify the RTSP client to connect to an RTSP server function of the gateway***. Ex.1007, 21:30-50. In my opinion, a POSA would have known that an RTSP "200 OK" message is used for this purpose. Indeed, Mela discloses that in response to the "200 OK" message at step H6, the RTSP client attempts to "start[]

streaming the movie" by setting up audio and video streams via the gateway's RTSP server functionality. Ex.1007, 21:49-59.

S. Claim 20: "The gateway of claim 18, wherein the gateway is configured to transmit a notification message to the second terminal in response to an invitation of the first terminal, in order to activate the second terminal to set up the second multimedia session between the second terminal and the gateway."

302. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

303. Mela teaches the ***gateway is configured to*** send the RTSP client (***second terminal***) an RTSP "200 OK" message, *e.g.* in step H6 of Fig. 8 (a ***notification message***) in response to the SIP "NOTIFY" message from the SIP media server (***first terminal***). Ex.1007, Fig. 4, Fig. 8, 7:32-47, 21:6-31. The "NOTIFY" message "encloses the stream parameters" and thereby provides an ***invitation*** to stream conent. Ex.1007, Fig.4, Fig. 8, 5:49-7:35; . In response to the "200 OK" message at step H6, the RTSP client attempts to "start[] streaming the movie" by ***setting up*** audio and video streams (*i.e.*, the ***second multimedia session***). Ex.1007, 21:49-59; *see also* Limitation [19i].

T. Claim 22: "The gateway of claim 18, wherein the gateway is configured to associate the first multimedia session information and the second multimedia session information with each other."

304. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

305. *See* Ground, III, Limitation [16], showing that the ***gateway associates the first multimedia session information and the second multimedia session information with each other***.

U. Claim 23: "The gateway of claim 18, wherein the gateway further comprises Session Initiation Protocol (SIP) client functionality and Real Time Streaming Protocol (RTSP) server functionality."

306. In my opinion, Mela alone discloses or, optionally in view of Alston, renders obvious this limitation.

307. Mela discloses that the gateway contains an SIP Client user agent and an "RTSP server" functionality because a POSA would understand from Fig. 3 and its description that the gateway has both SIP and RTSP functionality and, moreover this is a requirement to practice the invention because the gateway handles both SIP and RTSP. Ex.1007, 3:51-64, Fig. 3. For example, in the embodiment shown in Fig. 8, the gateway acts as an SIP client (to communicate with the SIP media server) and as an RTSP server (to communicate with the RTSP client). EX1007, Fig. 8.

XIII. SECONDARY CONSIDERATION

308. I am not aware of any secondary considerations, or so-called "objective indicia of non-obviousness" for the challenged claims.

309. Secondary considerations could include things like commercial success of a product due to the merits of the claimed invention; a long felt need for the solution provided by the claimed invention; unsuccessful attempts by others to find

the solution provided by the claimed invention; copying of the claimed invention by others; unexpected and superior results from the claimed invention; acceptance by others of the claimed invention as shown by praise from others in the field or from the licensing of the claimed invention; teaching away from the conventional wisdom in the art at the time of the invention; independent invention of the claimed invention by others before or at about the same time as the named inventor thought of it; and other evidence tending to show obviousness.

310. Patent Owner has not identified any such secondary considerations, and I am not independently aware of any. For example, Patent Owner has no commercial products embodying the ’151 patent that I am aware of. Additionally, as discussed throughout, the claimed features reflect predictable results of combining known elements according to their known functions, and thus, there are no unexpected and superior results from the claimed invention. Also, there are no licenses of the ’151 patent of which I am aware.

311. Further, even if Patent Owner were to assert that secondary considerations exist, given the strong reasons that the challenged claims are obvious in light, including the motivations to combine the references set forth above, I do not believe any such secondary considerations would rise to the level of overcoming the invalidity opinions I have expressed.

312. I understand that Patent Owner may address the issue of secondary considerations more fully in the future. I reserve the right to respond to any specific bases for such secondary considerations that Patent Owner may identify.

XIV. PUBLIC AVAILABILITY OF IETF DOCUMENTS

313. Exhibits 1006, 1008, 1009, 1011 and 1014-1017 are documents published by the IETF. For reasons explained below, in my opinion, these publications were publicly available by the date listed on the face of the publication, and the exhibits are true and correct copies of the publications on the date they were published.

314. Exhibit 1006 is an Internet-Draft proposal of improvements to media streams. Exhibits 1008, 1009, 1011 and 1014-1017 are Requests for Comments ("RFC's"). All of these documents were published by the Internet Engineering Task Force ("IETF") with which I have had significant involvement, as discussed above in my Background. The IETF is a well-known standards organization that publishes thousands of standards for protocols used on the Internet, including the protocols that make up the backbone of the Internet, such as TCP and IP. I am familiar with their publication practices, as would a POSITA.

315. A POSITA would have understood that since the 1990s, the IETF has published RFC's and Internet Drafts on its website as they became available. Such documents were generally drafted by a working group and then disseminated to the

public for comments via the IETF website and an email announcement with a link to the document. As a result, it is my opinion that the publication date on the face of these documents is a reliable indicia of its public availability, and that a POSITA would be able to locate IETF documents with reasonable diligence by that date. These dates are confirmed by the IETF website (https://datatracker.ietf.org) which lists the same dates of publication for each respective document.

316. I am also familiar with how IETF maintains the documents on its website. In my opinion, documents downloaded today reflect how they existed when they were first published, unless otherwise indicated. Thus, it is my opinion that the IETF exhibits are all true and correct copies of their respective publications on the date indicated on the face of each respective document.

XV. CONCLUSION

317. For the reasons set forth above, I believe claims 1-20, 22-23 of the ’151 patent are unpatentable in view of the prior art. In signing this declaration, I understand that the declaration will be filed as evidence in a contested case before the Patent Trial and Appeal Board of the United States Patent and Trademark Office. I acknowledge that I may be subject to cross-examination in this case and that cross-examination will take place within the United States. If cross-examination is required of me, I will appear for cross-examination within the United States during the time allotted for cross-examination.

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

Dated: January 20, 2025 By: Kevin C Almeroth

Kevin C. Almeroth, Ph.D.