Kundan Singh

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RESEARCH INTEREST

IP telephony, Internet real-time and multimedia communication systems, distributed cloud computing, next generation emergency services, peer-to-peer networks, Internet and web-based audio, video telephony and conferencing, unified messaging, scalable and reliable systems and computer networks.

EDUCATION

- **Columbia University**, Department of Computer Science, New York, NY, USA
  Ph.D., Oct 2006, Computer Science
  Thesis: Reliable, scalable and interoperable Internet telephony, Advisor: Prof. Henning Schulzrinne
- **Columbia University**, Department of Computer Science, New York, NY, USA
  Master of Science, Feb 2001, Computer Science
  Cumulative GPA: 4.066 on 4.0, Advisor: Prof. Henning Schulzrinne
- **Birla Institute of Technology and Science** (BITS), Pilani, India
  Bachelor of Engineering (Honors), July 1997, Computer Science
  Cumulative GPA: 10.0 on 10.0, *University Gold Medalist*

EXPERIENCE (approx 15 years)

- Senior Research Scientist, Avaya Labs, Santa Clara, CA, Mar 2012-till date
- Software Engineering and Consulting with startups (Twilio, Bittorrent, Emergent), Jan 2011-Mar 2012
- Lead Software Engineer, 6Connex, Campbell, CA, Sep 2009-Sep 2010
- Senior Software Engineer, TokBox, Inc., San Francisco, CA, Jan 2008-Jul 2009
- Senior Computer Scientist, Adobe Systems, San Francisco, CA, Aug 2006-Sep 2007
- Member of Technical Staff, Bell Labs/Lucent Technology, Holmdel, NJ, May 2006-Aug 2006
- Research Assistant, Columbia University, Computer Science, New York, NY, Sep 1999-May 2006
- Summer Intern, Bell Labs/Lucent Technology, Holmdel, NJ, Jun 2002-Aug 2002
- Senior Software Engineer, Motorola India, Bangalore, India, Jun 1997-Jul 1999
- Summer Intern, Bhabha Atomic Research Center, Mumbai (Bombay), India, Jun 1995-Jul 1995

AWARDS AND HONORS

- Three US patents granted #7,453,852, #7,257,201, #7,266,091
- Extraordinary Teaching Assistant Award, Fall 2001, Columbia University, New York, NY
- Research assistant for M.S. and Ph.D., Columbia University, New York, NY
- Grade of A or A+ in all subjects throughout my Bachelors, Masters and PhD study
- University Gold Medalist, 1997, Birla Institute of Technology and Science, Pilani, India
- Second rank among lakhs of students in board exam of both class 10 and 12, India
- Scored 100% marks in math in class 12, and science in class 10 board exam, India

OPEN SOURCE AND OTHER ACTIVITIES

- Open source video conference service, http://code.google.com/p/vvowproject
- Online book on "Implementing SIP telephony in Python", 2007-2008
- Software research project site for students, http://myprojectguide.org
- Open source P2P-SIP software in Python, http://39peers.net
- Open source web-based video communication software, http://code.google.com/p/videocity
- Open source REST server tools in Python, http://code.google.com/p/restlite
- Open source SIP-RTMP gateway in Python, http://code.google.com/p/siprtmp
- Open source Flash-based audio and video communication, http://code.google.com/p/flash-videoio
- Open source SIP in JavaScript with Flash as well as WebSocket/WebRTC, http://code.google.com/p/sip-js
- Ph.D. student representative, 2001, Computer Science Department, Columbia University
- Coordinator, Department of Hindi Press, APOGEE 1996, BITS, Pilani, India

ACADEMIC PUBLICATIONS

Journals

1. Kundan Singh and Henning Schulzrinne, "Failover, Load Sharing and Server Architecture in SIP Telephony", Computer Communications (Journal), Elsevier, Volume 30, Issue 5, pp.927-942, Mar 2007.
2. Milind Buddhikot, Adiseshu Hari, Kundan Singh and Scott Miller, "MobileNAT: A new Technique for Mobility across Heterogeneous Address Spaces", ACM MONET Journal, March 2005.

Magazines, RFCs

1. H. Sinnreich, A. Johnston, E. Shim, K. Singh, "Simple SIP Usage Scenario for Applications in the Endpoints". RFC 5638. IETF. Sep 2009.
2. Wenyu Jiang, Jonathan Lennox, Sankaran Narayanan, Henning Schulzrinne, Kundan Singh and Xiaotao Wu (alphabetical order), "Integrating Internet Telephony Services", IEEE Internet Computing (magazine), May/June 2002 (Vol. 6, No. 3).

Conferences

1. Carol Davids, Alan Johnston, Kundan Singh, Henry Sinnreich, Wilhelm Wimmreuter, "SIP APIs for Voice and Video Communications on the Web", IPTcomm, Chicago, IL, Aug 2011.
2. Kundan Singh and Henning Schulzrinne, "Failover and Load Sharing in SIP Telephony", International Symposium on Performance Evaluation of Computer and Telecommunication Systems (SPECTS), Philadelphia, PA, July 2005.
3. Kundan Singh and Henning Schulzrinne, "Peer-to-peer Internet Telephony using SIP", NOSSDAV, Skamania, Washington, June 2005.
4. Kundan Singh, Xiaotao Wu, Jonathan Lennox and Henning Schulzrinne, "Comprehensive Multi-platform Collaboration", MMCN 2004 - SPIE Conference on Multimedia Computing and Networking, Santa Clara, CA, Jan 2004.
5. Kundan Singh, Ajay Nambi and Henning Schulzrinne, "Integrating VoiceXML with SIP services", ICC 2003 - Global Services and Infrastructure for Next Generation Networks, Anchorage, Alaska, May 2003.
6. Wenyu Jiang, Jonathan Lennox, Henning Schulzrinne and Kundan Singh (alphabetical order), "Towards Junking the PBX: Deploying IP Telephony", NOSSDAV 2001.

Workshops

1. Kundan Singh and Henning Schulzrinne, "Peer-to-peer Internet Telephony using SIP", New York Metro Area Networking Workshop, City University of New York, New York, NY, Sep 2004.
2. Milind Buddhikot, Adiseshu Hari, Kundan Singh and Scott Miller, "MobileNAT: A new Technique for Mobility across Heterogeneous Address Spaces", WMASH 2003 - ACM International Workshop on Wireless Mobile Applications and Services on WLAN Hotspots, San Diego, CA, Sep 2003.
3. Kundan Singh, Ajay Nambi and Henning Schulzrinne, "Integrating VoiceXML with SIP services", Second New York Metro Area Networking Workshop, Columbia University, New York, NY, Sep 2002.
4. Kundan Singh, Gautam Nair and Henning Schulzrinne, "Centralized Conferencing using SIP", Proceedings of the 2nd IP-Telephony Workshop (IPTel'2001), April 2001.
5. W. Jiang, J. Lennox, H. Schulzrinne and K. Singh (alphabetical order), “Towards Junking the PBX: Deploying IP Telephony”, first New York Metro Area Networking Workshop, Hawthorne, NY, Mar 2001.
6. Kundan Singh and Henning Schulzrinne, "Unified Messaging using SIP and RTSP", IP Telecom Services Workshop 2000, Atlanta, Georgia, U.S.A, Sept 2000.
7. K. Singh, H.Schulzrinne, "Interworking Between SIP/SDP and H.323", Proceedings of the 1st IP-Telephony Workshop (IPTel'2000), April 2000.

Technical reports, Internet drafts

1. Kundan Singh and Carol Davids, "Flash-based Audio and Video Communications in the Cloud", Implementation Report, IIT VoIP conference and expo, Chicago, IL, Oct 2011.
2. K. Singh and H. Schulzrinne, Data format and interface to an external peer-to-peer network for SIP location service. Internet-draft (work-in-progress). IETF. May 31, 2006.
3. Kundan Singh and Henning Schulzrinne, "Using an External DHT as a SIP Location Service", Columbia University Technical Report CUCS-007-06, New York, NY, Feb 2006, Contributed to RFC 4123 on SIP-H.323 Interworking Requirements. IETF. July 2005.

4. Kundan Singh and Henning Schulzrinne, "Peer-to-peer Internet Telephony using SIP", Columbia University Technical Report CUCS-044-04, New York, NY, Oct 2004.
5. Henning Schulzrinne, Kundan Singh and Xiaotao Wu (alphabetical order), "Programmable Conference Server", Columbia University Technical Report CUCS-040-04, New York, NY, Oct 2004.
6. Kundan Singh and Henning Schulzrinne, "Failover and Load Sharing in SIP Telephony", Columbia University Technical Report CUCS-011-04, New York, NY, May 2004.
7. Kundan Singh and Henning Schulzrinne, "SIPpeer: A Session Initiation Protocol (SIP)-based peer-to-peer Internet telephony client adaptor", Implementation report, Columbia University, New York, NY, 2004.
8. Kundan Singh, Xiaotao Wu, Jonathan Lennox and Henning Schulzrinne, "Comprehensive Multi-platform Collaboration", Columbia University Technical Report CUCS-027-03, New York, NY, Nov 2003.
9. Kundan Singh, Wenyu Jiang, Jonathan Lennox, Sankaran Narayanan and Henning Schulzrinne, "CINEMA: Columbia InterNet Extensible Multimedia Architecture", Columbia University Technical Report CUCS-011-02, New York, NY, May 2002.
10. Agrawal, Roy, Palawat, Johnston, Agboh, Wang, Schulzrinne, Singh and Maeng, SIP-H.323 Interworking. Internet-draft (work-in-progress). IETF. July, 2001.
11. Kundan Singh and Henning Schulzrinne, "Unified Messaging using SIP and RTSP", Columbia University Technical Report CUCS-020-00, New York, NY, Oct 2000.
12. K. Singh and H. Schulzrinne, Interworking Between SIP/SDP and H.323. Internet-draft (work-in-progress). IETF. May 12, 2000.
13. Kundan Singh and Henning Schulzrinne, "Interworking Between SIP/SDP and H.323", Columbia University Technical Report CUCS-015-00, New York, NY, May 2000.

Patents

1. Granted: United States Patent 7,453,852, Method and system for mobility across heterogeneous address spaces, Buddhikot; Milind M., Hari; Adiseshu, Miller; Scott C., Singh; Kundan Narendra, Lucent Technologies Inc., Filed: July 14, 2003, Awarded: Nov 18, 2008. Also has international applications.
2. Granted: United States Patent 7,257,201, System and method for unified messaging in inter/intranet telephony, Singh; Kundan, Schulzrinne; Henning, The Trustees of Columbia University in the City Of New York, Filed: Aug 13, 2001, Awarded: Aug 14, 2007. Also has international applications.
3. Granted: United States Patent 7,266,091, System and method for conferencing in inter/intranet telephony, Singh; Kundan, Nair; Gautam, Schulzrinne; Henning, The Trustees of Columbia University in the City Of New York, Filed: Feb 28, 2002, Awarded: Sep 4, 2007. Also has international applications.
4. Pending US Patent Application 20070124813, May 2007, System and Method for Testing Network Firewall using Fine Granularity Measurements, Inventors: Gaston S., Henning Schulzrinne, Eilon Yardeni, Kundan Singh.
5. Pending US Patent Application 20050097222, May 2005, System and method for call routing in an IP telephony network, Inventors: Wenyu Jiang, Jonathan Lennox, Henning Schulzrinne, and Kundan Singh.

INVITED TALKS

1. Flash-based audio and video communications in the Cloud, IIT VoIP conference and expo, Chicago, IL, Oct 2010.
2. Peer-to-Peer Internet Telephony using SIP, Panasonic Digital Networking Lab., Princeton, NJ, Apr 2005.
3. Media Services in CINEMA, Intel/Dialogic facility, Morristown, NJ, Apr 2003.
4. Introduction to the Session Initiation Protocol, NYSERtech, Albany, NY,Oct 2002
5. Interworking between SIP and H.323, VON developers conference, Jan 2001 and Jul 2000
6. Overview of IP-H.323 gateway, Engineering group at Sylantro, Dec 1999

   Complete list of academic talks and demos at http://kundansingh.com/#talks

TEACHING AND STUDENT MENTORING

- **Teaching Assistant**: Advanced Internet Services (COMS E6181-1), Columbia University, Fall 2001, with 48 students enrolled in the class, and primary responsibility of evaluating assignments and programming projects, and interacting with the students regarding the course material. I received excellent TA award. I continued as the TA and coordinator of this course for the subsequent offering of this course over Columbia Video Network (a distance learning program) for Summer 2002, Fall 2002, Spring 2003 and Summer 2003, with main responsibility being designing and grading evaluation assignments, programming projects and final exams for the enrolled students.

- **Project Mentoring:** In the more than five years as a PhD student in the Internet Real Time Lab., I supervised many student projects such as active badges, event notification and scheduling system, screen sharing, floor control, file sharing, interworking between instant messaging and voice calls, phone announcement service, application level gateway for NAT and firewall traversal, email by phone, audio quality measurement for conferencing, location service for 911 calls in SIP proxy server, and integrating MPEG support in our media server. I also launched a software research project web site for students at http://myprojectguide.org to help project students and build community. My past and current student project can be found on that site. Furthermore, I manage and mentor the voice and video communications on the web project at Illinois Institute of Technology's ITM program.

OPEN SOURCE PROJECTS

You can get links to all my open source projects from my home page at http://kundansingh.com

- **VVoW** (Since Sep 2010): This web-based multiparty video conferencing and presentation application allows you to do real-time video conferencing, text chat and slide share. It uses resource oriented API to access the conferencing data model for communication using a generic backend MySQL/PHP server over websocket. More details are on the project page. http://code.google.com/p/vvowproject
- **39peers.net** (Since Sep 2007): This is my open source project that implements peer-to-peer Internet telephony software using the Session Initiation Protocol (P2P-SIP) in the Python programming language. P2P systems inherently have high scalability, fault tolerance and robustness against catastrophic failures. Internet telephony can be an application of P2P architecture where participants locate and communicate with each other without relying on expensive and managed service provider infrastructure. The project implements several specifications and IETF RFCs such as SIP, RTP, SDP, XMPP, NAT traversal, DHT, peers and servers. The project is developed for student developers and researchers to experiment with new ideas. http://39peers.net
- **SIP in JavaScript** (Jan-Mar 2012): This is my open source project that implements a full SIP and SDP stack in JavaScript running in the browser. It supports two modes – in Flash mode it uses Flash Player and a supporting AIR application to facilitate device access, media path and signaling transport; in WebRTC mode it uses WebSocket for signaling transport in conjunction with a SIP proxy that supports SIP over WebSocket, and uses WebRTC for the device access and media path. http://code.google.com/p/sip-js
- **The Internet Videocity** (Jul 2009-aug 2009): This is my open source Flash and web-based video telephony and conference application. The video communication is abstracted out as a city. Once you signup, you own a home, where you can have several rooms. You can decorate your rooms with your favorite photos and videos, invite your friends and family to visit a room by handing out Internet visiting card or softcard (TM), or visit other people's rooms to video chat with them or to leave a video message if they are not in their home. You can keep a room open for public or make it private. http://code.google.com/p/videocity
- **SIP-RTMP gateway**: The goal of this project is to allow Flash to SIP calls and vice versa. In particular it allows multimedia calls from Flash Player to SIP network and SIP network to Flash Player. The gateway implements translation of signaling as well as media between Flash Player's RTMP and standard SIP, SDP, RTP/RTCP. The client side API allows you or any third-party to build user interface of web based audio and video phone that uses SIP in the back end. The user applications can be built using ActionScript for web browser as well as standalone AIR. http://code.google.com/p/siprtmp
- **Flash-VideoIO** (2010): Flash-VideoIO is a reusable generic Flash application to record and play live audio and video content. It can be used for variety of use cases in audio and video communication, e.g., live camera view, recording of multimedia messages, playing video files from web server or via streaming, live video call and conferencing using client-server as well as peer-to-peer technology. http://code.google.com/p/flash-videoio
- **Restlite** (Nov 2009): Restlite is a light-weight Python implementation of server tools for quick prototyping of your RESTful web service. Instead of building a complex framework, it aims at providing functions and classes that allows your to build your own application. It combines REST, Python, JSON, XML, SQLite, and authentication. The goal is to provide simple tools instead of intrusive framework. http://code.google.com/p/restlite

INDUSTRY PROJECTS

- **Avaya Labs** (Mar 2012-till date): In a research setting, I am comparing the various approaches to integrate the emerging web-based real-time communication (WebRTC) with legacy VoIP systems. I am exploring various enterprise use cases of the emerging HTML5 standards. I create ideas and prototypes to propose future directions.

- **Emergent Communications Inc** (Apr 2011-Feb 2012): Emergent communications provides software pieces for the next generation emergency services by productizing the innovations from Columbia University where the core pieces such as SIP and location-to-service-translation (LoST) were invented. Some of these software pieces were written by me during my time at the university. I was responsible for the entire software engineering and development at Emergent. The job involves productizing the software using advanced cloud-based services and modern web based applications for the call taker terminals. The main challenge is in creating integrated software solution out of the multitude of technologies used and developed at the university. http://emergentcomm.com

- **Twilio Inc** (Jan 2011-Feb 2012): Twilio is a cloud telephony application provider that allows web developers to take full advantage of the legacy telephone system by using a simple, elegant and robust API hosted on the amazon cloud. As a part time engineer, I was responsible for architecture, design and prototype implementation of mobile applications and gateway for web browsers to connect to the service. I also did performance measurement of voice quality under server load, and built prototype of a video pipe from the browser client. I served as an in-house technology expert for Flash-based communication (RTMP) and SIP/RTP standards. http://twilio.com

- **6Connex Inc** (Sep 2009-Sep 2010): It provides a virtual experience platform where organizations can host virtual events and participants attend sessions, interact in virtual rooms, and build their social network. I lead the architecture and implementation of audio and video communication, conferencing, messaging and social interaction among the participants for the platform. http://6connex.com

- **Tokbox Inc** (Jan 2008-Jul 2009): It allows web-based video telephony for Internet users using easy to use Flash technology. As a senior software engineer, my role at Tokbox was to enhance the system by using standard protocols such as SIP and XMPP for signaling of video communication, to interact with telephone network to allow PC to Phone and Phone to PC communication, and to build scalable and reliable backend infrastructure for the Internet scale. I have also helped in building more robust front end using the new Adobe Flex technology. http://tokbox.com

- **Adobe Systems** (Aug 2006-Sep 2007): As a senior computer scientist at Adobe, I designed and implemented Flash-based Internet telephony and peer-to-peer systems. I implemented a SIP stack and a P2P library in ActionScript and built several prototype Flash-based applications such as integrated SIP and XMPP communicator, click-to-call Flash component, browser extensions for Firefox and IE for PC to phone calling, and a P2P-SIP user agent. My P2P implementation was based on Bamboo DHT and incorporated authenticated data storage, secure transport and reliability as discussed in my thesis.

- **Bell Labs** (May 2006-Aug 2006): As a member of technical staff, I did design of a scalable and robust server-less infrastructure for mobile carriers to support gaming and other services in a distributed peer-to-peer manner. I also did design and implementation of Java-based user interface of an attack detection software for mobile carriers.

- **MobileNAT** (Jun 2002-Aug 2002): As a Bell Labs intern, I did research, design and implementation of MobileNAT that provides IP mobility for devices in private address spaces. I wrote the client application that implements DHCP client and server, and the driver that traps and alters the IP packets on Windows XP. I also wrote the server application that runs on the Linux router, implements DHCP server and alters the NAT mapping. The project was one part of the bigger project on integration of 802.11 and 3G technologies.

- **Motorola India** (Jun 1997-Jul 1999): In a group of two engineers, we did design and implementation of a complete H.323 video conferencing client for Windows using external components for initial signaling and media codecs. I also helped in various other ongoing projects such as VoIP gateway, embedded systems for H.320 video conferencing, and mentored an intern for H.323-H.324 gateway. I also did internship from Jan 1997 to June 1997 for six months as part of my B.E. curriculum.

ACADEMIC PROJECTS

- **Columbia University** (Sep 1999-May 2006): My research focus was on scalable and robust Internet telephony and multimedia internetworking: IP telephony, SIP-PSTN interworking, SIP-H.323 signaling gateway, SIP-RTSP based unified messaging system, comprehensive multimedia collaboration, VoiceXML-based interactive voice response system, SIP/RTP based scalable and robust, conference, and SIP protocol stack. Focus of my thesis is on scalability and reliability of IP telephony systems in peer-to-peer as well as server-based architectures using existing standards. My software pieces were part of CINEMA (Columbia INternet Extensible Multimedia Architecture) test bed in Prof. Schulzrinne's lab. My software pieces were productized and sold by startup companies, SIPquest and FirstHandTech, spun out of Columbia University. http://www.cs.columbia.edu/irt/cinema. During the initial years, I wrote an object-oriented SIP user agent library in C++, using our underlying SIP transaction and parsing library. I developed other components such as unified messaging, voice mail and answering machine server, multimedia conference server, interactive voice response server and SIP-H.323 signaling gateway. I wrote reusable object oriented modules for conference library and media-streaming library. Later, I built scalability and reliability mechanism for SIP servers that provide PSTN-grade availability (five nines) and scalability (ten million BHCA), albeit at much lower cost. I also developed techniques and built systems for robust and scalable peer-to-peer Internet telephony without incurring any server maintenance cost.

TECHNICAL SKILLS

- I have extensive programming experience in C, C++, Python, ActionScript (Flex), Java, Tcl and Perl. I have worked on both Unix and Windows platforms, as well as on real time OS. I am familiar with various tools such as MySQL, Apache, TomCat, gcc/make, VC++, CGI, servlet, Flex Builder, Eclipse, LAMP/WAMP, git, cvs and svn. I have worked with various hardware and software tools such as Cisco router 2600 series, Cisco IP phone, Nortel MCS 5100 system, Intel/Dialogic IP telephony, MySQL replication, Vovida's SIP and TRIP stacks, DNS SRV and NAPTR, DHCP server and client, FMS and Red5 media servers, SER/OpenSER servers, Google App Engine, Facebook application, RESTful architecture. I have working knowledge of software process including CMM quality levels and software design models. I have also worked briefly with Linux kernel module programming, Windows driver programming and MacOS audio module programming.

- I have extensive experience with various Internet protocols such as Session Initiation Protocol (SIP), Real-time Transport Protocol (RTP), Real Time Streaming Protocol (RTSP), Session Description Protocol (SDP), Extensible Messaging and Presence Protocol (XMPP), Real-time messaging protocol (RTMP), VoiceXML, Simple Object Access Protocol (SOAP), ITU-T recommendations H.323, H.225.0, cryptography, security protocols, wireless/mobility protocols such as Mobile IP and some intra-domain mobility protocols for fast handoff, IP-PSTN interworking for telephony and related protocols. I have worked extensively on server scalability and reliability, and peer-to-peer systems and algorithms.

PHD DISSERTATION

Title: **Reliable, Scalable and Interoperable Internet Telephony,** Defense Date: June 21, 2006.
Committee: Prof.Henning Schulzrinne, Prof.Gail Kaiser, Prof.Vishal Misra, Prof. Dan Rubenstein, Dr. Milind Buddhikot

Abstract: The public switched telephone network (PSTN) provides ubiquitous availability and very high scalability of more than a million busy hour call attempts per switch. If large carriers are to adopt Internet telephony, then Internet telephony servers should offer at least similar quantifiable guarantees for scalability and reliability using metrics such as call setup latency, server call handling capacity, busy hour call arrivals, mean-time between failures and mean-time to recover. This thesis presents a reliable, scalable and interoperable Internet telephony architecture for user registration, call routing, conferencing and unified messaging using commodity hardware. The results extend beyond Internet telephony to encompass multimedia communication in general.

The architecture presented in this thesis deals with two aspects: at least PSTN-grade reliability and scalability of the Internet telephony servers, and interoperable Internet telephony services such as conferencing and voice mail using existing protocols. We describe the architecture and implementation of our Session Initiation Protocol (SIP)-based enterprise Internet telephony architecture known as Columbia InterNet Extensible Multimedia Architecture (CINEMA). It consists of a SIP registration and proxy server, a multi-party conferencing server, a gateway for interworking SIP with ITU's H.323, an interactive voice response system and a multimedia mail server. CINEMA provides a distributed interoperable architecture for collaboration using synchronous communications like multimedia conferencing, instant messaging, shared web-browsing, and asynchronous communications like discussion forum, shared files, voice and video mails. It allows seamless integration with various communication means like telephone, IP phone, web and electronic mail.

We present two techniques for providing scalability and reliability in SIP: server redundancy and a novel peer-to-peer architecture. For the former, we use DNS-based load sharing among multiple distributed servers that use backend SQL databases to maintain user records. Our two-stage architecture scales linearly with the number of servers. For the latter, we propose a peer-to-peer Internet telephony architecture that supports basic user registration and call setup as well as advanced services such as offline message delivery, voice mail and multi-party conferencing using SIP. It interworks with server-based SIP infrastructures.