# A Look Inside a SIP Testing Lab

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# TMC SIP MAGAZINE™

The Authority On Session Initiation Protocol

**SIP Breaks Cost and Complexity Barriers**

**Prof. Carol Davids on SIP Testing**

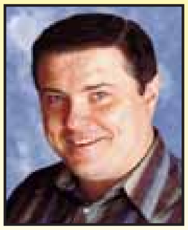
**Cisco and Orative Bring Mobility to Unified Communications**

# SIP Goes Mobile & Global


UNDER NEW MANAGEMENT (AGAIN)!

# SIP: The S is for Sizzle

EDITOR'S NOTE



by
Richard "Zippy" Grigonis

**As he wrote last month, Greg Galitzine, our long-time Executive Editor, is moving onward and upward to devote more time and energy to our fabulous web resource, tmcnet.com.**

As for myself, Greg's replacement, some of you may remember me as the fellow who founded *VON Magazine* for Jeff Pulver. Others of you with longer memories may recall that I was Chief Technical Editor of *Computer Telephony* magazine (later known as *Communications Convergence*), where I worked with the inimitable Harry Newton and editor Rick Luhmann for many years.

In any case, I now find myself at the helm of three of TMC's most illustrious magazines: *Internet Telephony*, *IMS* and of course *SIP*.

SIP, that text-based signaling protocol for creating, modifying and terminating sessions involving Internet phone calls, multimedia transmissions and multimedia conferences, has come along way since its original formulation by Columbia University's Henning Schulzrinne. This magazine is in a sense a celebration of both SIP's triumph over competitors such as H.323, and its evolution from a lightweight transport-independent protocol living in the shadow of its brethren, HTTP and SMTP, to its many present-day roles serving as the basis of all IP Communications.

SIP appears both in the voice world and in the world of Instant Messaging as the open standard, SIMPLE (Session Initiation Protocol for Instant Messaging and Presence Leveraging Extensions), an IM and presence protocol suite. Moreover, thanks to products from companies such as Ubiquity Software and dynamicsoft, SIP-based applications of arbitrary complexity can now run in the network.

And that leads us to IMS (IP Multimedia Subsystem), which takes SIP to it highest conceptual level, making it a part of a common service architecture for both wireless and wireline services. Sonus (http://www.sonusnetworks.com) and Cantata (http://www.cantata.com) recently announced an agreement to work together to deliver IMS-ready applications and services to network operators. Cantata's SnowShore IP Media Server will be integrated with Sonus' IMS-ready ASX Feature Server and IM Media Application Platform, facilitating the delivery of SIP and VoiceXML-based network applications. (For articles devoted to IMS, check out TMC's magazine on the subject.)

Of course, somebody has to verify that the new incarnations of SIP are doing their job. The rise of SIP took some testing equipment vendors and labs by surprise, but that's rapidly changing, as we reveal in the article on SIP testing in this issue and our Q&A with Prof. Carol Davids, Director of the VoIP Laboratory at the Illinois Institute of Technology (IIT) in Chicago. Indeed, just as this issue was about to go to press, it was announced that a new $5 million 3G/IMS networking lab at Georgia Tech would be dedicated by Siemens Networks LLC. The lab's co-sponsors include Cingular Wireless, the Georgia Electronic Design Center (GEDC) and the Georgia Tech Research Network Operations Center (GT_RNOC). Initially, it will be used by those students participating in the IMS Research Competition, which will award winning students with a total of $100,000 in cash prizes. (Makes me wish I was a student again!)

SIP steadily becomes ever-more ubiquitous, and its continuing success reflects the success of IP Communications in general. We at TMC hope that you'll follow SIP-based products and services, both in this magazine and on our amazingly high-ranked web resource, tmcnet.com.

*Richard Grigonis is Executive Editor of TMC's IP Communications Group.*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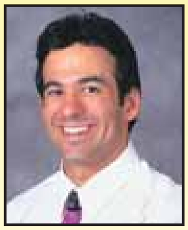
PUBLISHER'S OUTLOOK

# Unified Communications Meets Presence and Mobility



by
Rich Tehrani

**For years I have been writing about communications, constantly searching the world for exciting news to share with my readers. Often this job of mine allows me to connect with people who I know are destined to do great things. For example, in the spring of 2003 I ran into Andre Nakaso, one of the cofounders of Orative, and was very impressed with their technology that allows cellphones to have very rich presence and availability.**

I even wrote an article entitled, "Presence Meets the Cell Phone Thanks to Orative" (http://www.tmcnet.com/tmcnet/columns/2004/072304rt.htm) where I detailed how the company's products enable a corporation's workers to stay better connected than ever.

As I spoke with Andre I saw in his company's technology something that could make real my dream of "Just in Time Communications" — or the ability to more efficiently communicate. Using Orative you could send a note to a coworker telling them you want to speak to with them and even transmit the subject of the conversation. The person you are trying to contact could respond when the time is right for them — thus putting them in control over their communications.

In short, I was very impressed with what I saw and I knew the company was onto something very big — Andre had a winner. But at the time, in 2003, things were pretty lousy in the telecom space. Subsequently, I didn't know what sort of future the company would have as things were pretty rocky.

Still, the company had just received their funding a few months before my visit and I was the first person in the media to discover Orative.

Over time I kept in touch with Andre as I had good feeling about what the company was up to; moreover, I was excited to hear how the company was progressing.

In my last meeting with Andre in San Francisco, he hinted to me that some big things were coming regarding his company but couldn't elaborate.

Now it can be revealed: The big news is that Cisco just acquired the company and is using Orative's technology to bring mobility to unified communications.

I recently asked Andre for some of his thoughts on the merger and he told me there has been a huge shift in the telecom space, with wireless minutes surpassing wireline. Andre went on to say that in many companies, decision makers were evaluating top-of-the-line wireless devices as primary devices of choice for their mobile enterprise workers, since the price of these wireless devices have been steadily dropping, while the features are increasing in number and capability.

Some of the benefits of the Orative solution include Rich Caller ID/call screening, a single voicemail store with voicemail notification on the handset. In addition, users can see the messages and decide which ones they want to listen to. This is a much better scenario than having to scroll through messages serially.

It is no secret that Cisco (quote - news - alert) has been focusing heavily on unified communications and of course if you are going to unify communications you need to bring it to mobile devices as well. This is why the company decided to purchase Orative Corp. (http://www.orative.com) for $31 million. Essentially, Orative offers a client application that "sits on top of a cell phone and communicates to the server, which, in turn, communicates with Cisco's Unified Communications products and provides Unified Communications access to the cell phone," explained Cisco's director of Mobile Unified Communications Alex Hadden-Boyd.

With Orative software, users will be able to access their corporate directories, use five-digit dialing, use least cost routing, view information about voice messages and click to hear individual messages, and receive notification about MeetingPlace conference calls on mobile devices — essentially, it extends soft client capabilities to the cell phone.

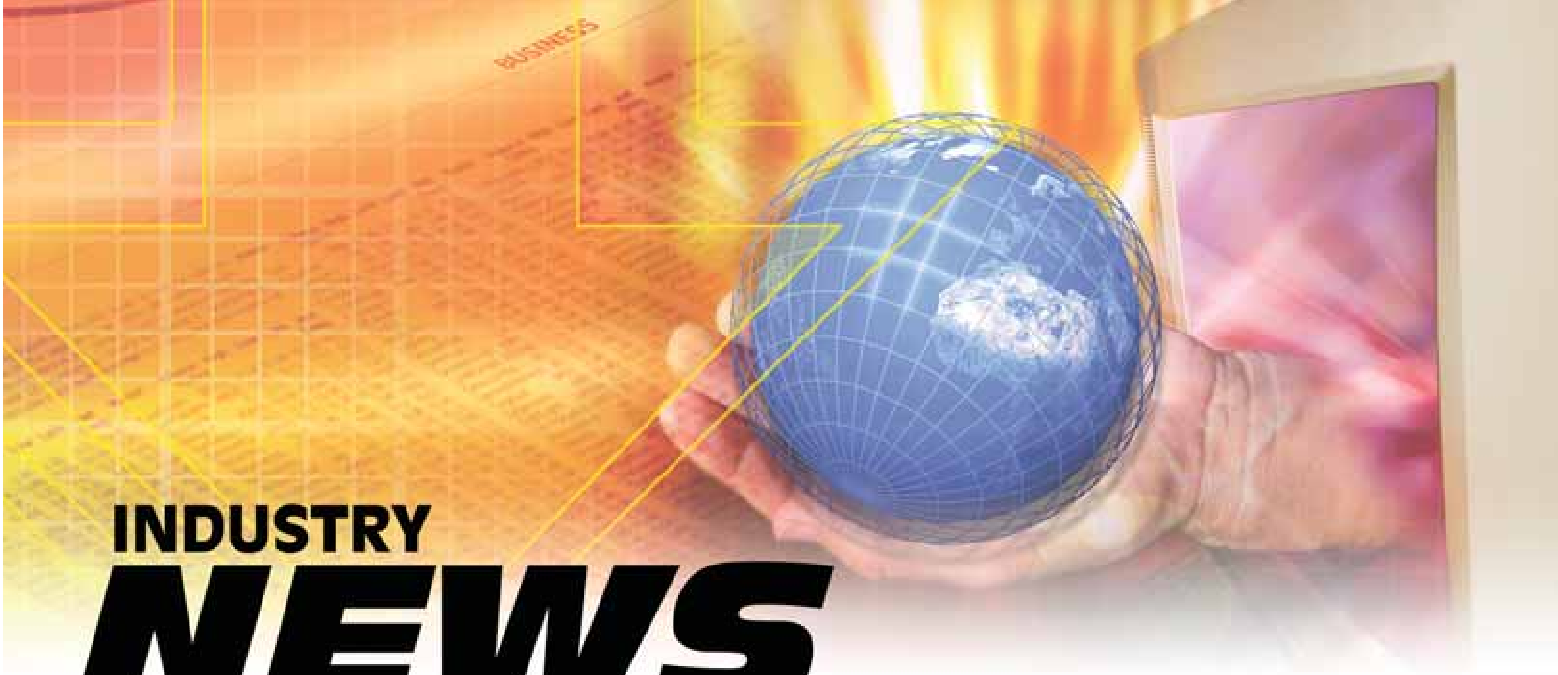
"Cisco wants to do for unified communications what Blackberry has done for push mail," added Hadden-Boyd.

In its quest to extend its popular Call Manager solution to mobile devices, Cisco looked at several potential companies, but opted for Orative (which is also a Cisco Technology Development Partner), because "Orative had best technology and the best architecture for future enhancements," according to Hadden-Boyd. Future enhancements will include being able to accommodate instant messaging between mobile devices and other IM systems.

Also key is that the Orative solution currently supports four major mobile operating systems — Symbian, Blackberry, BREW, and J2ME — with support for Windows Mobile on the horizon. This makes its deployment considerably easier in the United States, where mobile operators want to certify anything that goes on their networks, because Orative already holds those certifications.

Once the deal is finalized, Cisco will focus on scaling the product to meet the demands of even Cisco's largest Call Manager customers, of which it has 20 with more than 10,000 users. The company will also introduce a look and feel to the product that mirrors its Unified Personal Communicator product. Cisco expects to bring its new mobile solution to market in early 2007, shortly after closing the transaction.

So, hats off to Orative (news - alert) for seeing this opportunity so far ahead of the market. Although a small company's acquisition by a large company does not always mean the technology will thrive, I feel the synergy between Cisco and Orative will yield positive results for the combined company and Cisco's massive sales force will likely do well selling Orative products. Interestingly, rumor has it that a number of other big players were interested in acquiring Orative as well. All in all, I expect the mobility market to continue to be a hot space in 2007.




# INDUSTRY NEWS

## Esnatech Adds SIP Integration to Telephony Office-LinX Platform

Esna Technologies (news - alert) announced it has added new SIP integrations to its Telephony Office-LinX unified communications platform. As part of its ongoing goal for interoperability and to offer the most integrations in the market, Telephony office-LinX will support on its first release SIP integration to the Iwatsu ECS, Teltronics 20-20, 3com NBX, Dialogic PMG gateway, and Asterisk-based VoIP solutions.

Esnatech plans to expand its complete product portfolio compatibility with advanced IP communications systems by focusing on SIP (Session Initiation Protocol), a leading IETF signaling protocol for Voice over IP (VoIP) solutions. Following this initial release of its SIP interoperability strategy, Esnatech will expand its validation to offer integrations to the Nortel CS1000 and Avaya Communications Manager products, with plans to expand even further, including compatibility with Microsoft's LCS.

SIP integration reduces the cost of hardware, provides flexibility in terms of deployment location and remote site support, as well as support for high availability and resiliency initiatives. "SIP is a very important protocol for IP telephony, video, instant messaging, and presence. It is the ideal protocol for Esnatech to leverage its future development for unified communications solutions like Telephony Office-LinX" said Esnatech's CEO/CTO, Mohammad Nezarati. "SIP will allow us to provide seamless integration into different business applications and deliver real time availability of content from any access point to our customers."

http://www.esnatech.com

## Octasic Enhances Media Gateway Modules with Stacks from RADVISION

*By Mae Kowalke*

Semiconductor manufacturer Octasic (news - alert) announced it has added media gateway controller (MEGACO) and Session Initiation Protocol (SIP) stacks from RADVISION (news - alert) to its OCT9320 and OCT9360 module families.

The addition of the new stacks means that the company's modules now "modules perform all the voice processing and packet signaling required for media gateways in a ready-to-deploy format."

Octasic's Director of Product Management Frédéric Bourget said that the enhanced technology greatly simplifies the tasks of media gateway design engineers because hardware-software integration is already completed. That results in reduced development time and costs for original equipment manufacturers (OEMs).

Octasic and RADVISION say they have performed rigorous testing of their integrated products, validating the solution as interoperable with a variety of gateways.

http://www.octasic.com
http://www.radvision.com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## Reef Point Unveils Universal FMC Gateway

Reef Point Systems (news - alert) unveiled its Universal Convergence Gateway (UCG), which gives telecommunications service providers a single platform for delivering VoIP and multimedia services to their customers over all major fixed/mobile convergence (FMC) network technologies and standards: mobile, fixed-line and wireless.

"Based on consumer behavior and the way industry standards are evolving, it is clear that high value telecomm services from now on will be FMC-based: mobile, IP, and personalized for each subscriber," said Woody Ritchey, ReefPoint's CEO. "The service providers who will win market share are those who can deliver any application, to any user's device, over any network. [This is] a practical solution that enables service providers to capitalize on these trends now."

The Universal Convergence Gateway is a control point that consolidates all major networks for accessing a service provider's core infrastructure; adheres to all major carrier IP infrastructure standards; has the intelligence and policy enforcement to assure that the network delivers personalized quality-assured IP services that subscribers have purchased; and protects service provider infrastructure from Internet threats and revenue from theft.

Situated between a service provider's core infrastructure and the networks that subscribers with mobile phones, laptops, and PDAs use to access services - VoIP, Web, IM, music and video, IPTV - Reef Point's UCG is purpose-built to address the full set of session-aware FMC convergence control functions for the full range of access networks and industry standard FMC architectures.

"Reef Point offers deep inspection and custom handling of packets that's appropriate to everything from security to QoS, from voice to legacy data services. This is the kind of offering that is needed to address the issues that every provider with an FMC mission will surely face," said Tom Nolle, President of CIMI Corporation, a leading independent telecommunications industry consulting firm.

http://www.reefpoint.com

## IVR Technologies, Inc. Releases Talking SIP 3.1

IVR Technologies, (news - alert) a leading software developer for IP enhanced services and real-time billing solutions, announced the availability of Talking SIP 3.1. The release is the end result of a tremendous investment by IVR Technologies in the development of its third generation architecture and revenue-generating applications with real-time billing control and flexibility, but without sacrificing ease-of-use nor low administrative overhead.

"Talking SIP's signaling, media processing, and application scripting components have been completely redesigned to reach new milestones in scalability, performance, and control for the next-generation network. We are extremely excited about this release as it provides an amazing foundation upon which to build voice 2.0 applications that highlight the empowerment, flexibility and control that broadband access and Voice over IP have to offer," said Randall O. Walrond, Product Development Manager, IVR Technologies.

Talking SIP 3.1 is built around a completely redesigned application, media and signaling architecture to serve as the foundation for rich, in-demand applications that capitalize on the ubiquity, access and richness of broadband connectivity.

Release 3.1 includes the following features and functionality: International and Web callback, reminder/wake-up service, improved performance, improved tones, asynchronous voice prompts, multi-leg billing for callback, supports any codec, open XML interface for Credit Card Processing.

http://www.ivr.com

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## Siemens and Cingular Sponsor Multimillion Dollar Networking Lab

Siemens Networks (news - alert) has dedicated a new 3G/IMS laboratory at the Georgia Institute of Technology. The new lab, co-sponsored by Cingular Wireless, the Georgia Electronic Design Center (GEDC), and the Georgia Tech Research Network Operations Center (GT-RNOC), will serve students and researchers as a test bed for the creation of 3G IP Multimedia Subsystem (IMS) convergence applications and services that bring together audio, video and data over computer networks.

Initially, the 3G/IMS laboratory is to be used by students participating in an IMS Research Competition co-sponsored by Siemens, Cingular and Georgia Tech. Qualified students who participate in the year-long competition will be eligible to receive a portion of $100,000 in cash prizes.

As part of its sponsorship, Siemens Networks is equipping the new 3G/IMS lab with the latest technology from its IMS portfolio, including IMS@vantage, a 3GPP-compliant implementation of IMS. The IMS@vantage product offering includes the Multimedia Session Controller CFX-5000 and the Home Subscriber Server CMS-8200, as well as the Media Gateway Controller CFX-5200 and the Media Gateway CMG-3000. The IMS@vantage suite gives network operators a strategic platform to deploy multimedia services, such as Voice over IP (VoIP), for fixed or wireless infrastructures.

Siemens is also including a suite of enabling servers to allow student applications to use key IMS building blocks. These enablers include Siemens Mobile Presence Manager (Presence status), Community Service Data Manager (Group list management), Instant Messaging and Chat, and Push and Talk over Cellular.

"The new 3G/IMS laboratory will give Georgia Tech's students an opportunity to work with leading-edge technology in their quest to develop new and unique applications," said Ron Hutchins, associate vice provost for research and technology and chief technology officer at Georgia Institute of Technology.

The 3G/IMS laboratory will incorporate subsystems and services from additional sponsors Ubiquity Software (SIP A/S, SOOF Architecture), BridgePort Networks (NomadicONE(TM) IMS Convergence Server (ICS)), and Azaire Networks (TTG/PDG technology, based on ATCA platform).

http://www.siemens.com

## Woize Challenges Skype and Announces Mac Software Plans

Woize International, (news - alert) the digital communications provider, is taking on challenging the Skype and Skype-like providers by announcing plans to launch a Woize BETA client for Mac OS X in Q4 2006. A Woize client for Mac OS will challenge Skype and other similar Peer-to-Peer providers from a technical standpoint, since it will be fully SIP-based, thereby giving users the opportunity to interconnect with other SIP and P2P users worldwide. As are all other Woize clients, the Mac product will be free of charge, allowing users to call outside the Woize community with no hidden start-up fees for incoming or outgoing calls. A free phone number will be included in the Woize package and becomes available when signing up.

Some of the main features of the soon-to-be-released SIP-based Woize BETA client for Mac OS X are:

- Free calls amongst Woize® users
- Free telephone number usable worldwide
- Free calls amongst SIP users
- Free voice mail
- Cheap calls to landlines and mobiles worldwide
- Instant messaging
- Contact list
- Account handling

http://www.woize.com


### Eutectics IPP200 and IPP200T USB Phones Certified for Alcatel OmniPCX

Eutectics Incorporated announced today that its IPP200 and IPP200T USB phones have passed certification testing conducted by NextiraOne Communications (news - alert) for use with the Alcatel VoIP WebSoftphone, VoIP 4908 Softphone R4.0, IP PIMphony Multimedia R5.1 & 6.0, OmniPCX Office R5.0, and OmniPCX Enterprise R7.0 softphones.

Testing was carried out by NextiraOne Deutschland GmbH and was completed earlier this month. Alcatel provided the integration software to support Eutectics devices using Eutectics API and Toolkit. The IPP200 and IPP200T are packaged in professional, ergonomic handset designs that provide users of PC telephony desktops with a high quality, low cost telephony experience.

"We're extremely pleased to be certified by NextiraOne's test efforts for use with Alcatel's line of Enterprise Softphones," said Don Fowler, VP of Sales for Eutectics. "The convergence of PC based telecommunications at the desktop is rapidly gaining momentum in 2006. This combination of a professional soft platform with an elegant equipment solution will fuel this change."

The Eutectics (news - alert) IPP200T list price is less than $60. Eutectics USB phones, when combined with the complete line of enterprise class Alcatel (news - alert) softphones, provide a robust solution for users that desire a professional integrated telecommunications desktop.

www.alcatel.com
www.nextiraone.com
www.eutecticsinc.com

### Covergence Delivers Secure SIP Trunking to Avaya Customers

Covergence (news - alert) announced the availability of its secure SIP trunking solution for customers using Avaya's (quote - news - alert) IP telephony products. Delivered through its Eclipse family of SIP security solutions, Covergence's secure SIP trunking solution enables enterprises using Avaya's SIP Enablement Services (SES) to securely connect their Avaya Communication Manager (CM) platforms to an IP network. With the Covergence solution in place, enterprises can quickly and easily implement secure VoIP throughout their organizations, yielding significant cost savings, while protecting all user information through comprehensive application-level security to defend their VoIP service from attacks and intrusions, even over untrusted networks.

With Covergence's SIP trunking solution, customers can immediately move their telephony traffic onto an IP network to eliminate toll charges and redundant management and administration costs, while securely deploying real-time services between the service provider and enterprise or between geographically disperse sites within the enterprise, and secure connections to remote Avaya handsets located in branch offices. Covergence's Eclipse provides Avaya customers with fully authenticated, validated and encrypted connections to protect all user information, plus comprehensive application-level security to defend the service from attacks and intrusions.

Additionally, Eclipse's clustering capabilities provide load balancing for consistent performance, and redundancy for high availability through automatic IP rerouting that maintains the availability of VoIP trunks through most wide-area network failures. If the central site becomes unreachable or VoIP service fails, Eclipse provides local call completion within remote sites and diverts external calls to the PSTN. The Eclipse also offers central quality of service (QoS) monitoring, service-level agreement (SLA) verification and other management tools that give network operators the information they need to maintain availability and performance integrity.

http://www.covergence.com
http://www.avaya.com


## Polycom and Digium Offer SIP-Based Telephony Solution for Small/Medium Business Market

Collaboration platform provider Polycom and Digium, the Asterisk company, announced a multiyear agreement to develop and market integrated, SIP-based telephony solutions for SMBs that will give customers a tightly integrated, standards-based solution with simplified provisioning, broad support for Asterisk telephony features on the Polycom phones, and the delivery of new capabilities, such as Polycom's breakthrough HD Voice quality.

Polycom's (news - alert) SIP desktop and conference phones will be combined with Digium's Asterisk Business Edition, the professional-grade version of Asterisk, the industry's first open source PBX. Through the combined Digium and Polycom offering, SMB customers will have access to advanced telephony solutions that will be more affordable than proprietary systems. Additionally, the new offering will give customers the control, rapid feature development and deployment, and rich feature base that the Asterisk open source community and its partners provide. Per the agreement, Polycom will also be the preferred VoIP phone provider for Digium solutions.

As part of the agreement, Digium (news - alert) will modify the Asterisk graphic user interface (GUI), Asterisk Business Edition application and Asterisk OS to tightly integrate with Polycom's line of SIP-based desktop and conference phones. The co-development work will enable simplified provisioning and support for advanced telephony features like shared line appearance, XML microbrowser plugins and Polycom HD Voice. Polycom and Digium will also work together on joint marketing initiatives and joint selling through common channels. Polycom phones will also be the exclusive VoIP phone in Digium's Asterisk Appliance Developer Kit (AADK).

http://www.polycom.com
http://www.digium.com

## Auvi Adds Dual Mode Cordless Phone to VoIP Family

Auvi (news - alert) has added the PHIP65 dual mode cordless phone to its family of VoIP endpoints. The sleek PHIP65 is the next generation of communication hardware that connects to a traditional landline and the computer, via a USB port. This dual functionality gives consumers the freedom to make and receive both standard and Internet calls from a single cordless phone.

Designed to work with Skype, (news - alert) the PHIP65 enables individuals to use their Skype contact list at a push of a button to talk to other Skype users globally for free or make low-cost calls through other Skype services. In addition, users can use their regular phone service for local calls.

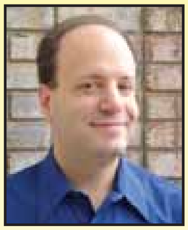
"The VoIP industry is changing the way individuals communicate, yet not everyone is willing to make the full leap into this technology and still rely on their traditional phone service," said Santosh Patel, CEO and President of Auvi Technologies.

Expandable up to four handsets, the PHIP65 is based on the new DECT technology to enhance sound quality while reducing interference with 802.11a/b/g wireless networks, Bluetooth devices, and other home appliances. The unit features an integrated speakerphone, caller ID, three-way calling, call transfer between handsets, and large LCD display on handset. The PHIP65 handset also offers different ring tones to distinguish between landline and Skype incoming calls.

http://www.auviworld.com
http://www.skype.com

SPEAKING SIP

# Peer-to-Peer SIP: More and Less than You Think



by

JD Rosenberg

**"Peer-to-peer." The term seems to evoke an emotional response of one sort or another from almost everyone in the technical community. Within the music and movie industries, it conjures images of piracy, copyright violations and teenage evildoers who think stealing music isn't stealing at all. Among ISPs, it stirs up concerns over network congestion and overload due to the volume of traffic and induces fears about liability. Among the general populace, some see it as a tool for getting free stuff, and others see it as a vehicle for getting into trouble.**

Given the excitement and controversy, peer-to-peer VoIP and specifically peer-to-peer SIP is, unsurprisingly, a contentious topic.

Certainly, the idea of using SIP (define - news - alert) in a peer-to-peer configuration is not new. Indeed, SIP was designed with many of the same concepts that P2P is built on — the placement of intelligence in the endpoints rather than centralization within the network, for example. However, P2P and VoIP were only really brought together when Skype came into view and became a smashing success. Given how close SIP already was to being P2P, interest surged in developing standards and technologies around true P2P SIP. Numerous documents proposing mechanisms and requirements for P2P SIP have been submitted to the Internet Engineering Task Force (IETF), and a pre-working group is being held during the November IETF meeting in San Diego.

What does P2P SIP mean for the industry? Is it just a tool for building standards-based clones of the Skype experience? This common perception represents perhaps the least useful and least likely benefit of the technology. Indeed, PSP SIP applications are quite broad and far-reaching:

**Small Enterprise Systems:** The cost of PBX systems, both in terms of capital and operational expenses, is a big impediment to moving them down market into small enterprises and even SOHO markets. One idea for reducing that cost is to eliminate the actual PBX itself and instead provide phones that self-organize into a low-overhead P2P network.

**Emergency Response:** In national emergencies, such as a hurricane or terrorist attack, the telecom infrastructure might be out of service. However, the need for communications among the fire, policy, and government officials that are onsite is more critical than at any other time. One way to solve this is for an *ad hoc* wireless network to be put in place, and then layer on top of that a P2P communications system. Because there is no centralized infrastructure at all, the network is robust against failure and easily set up in time of need.

**Privacy:** Achieving true privacy and anonymity on the Internet is actually very challenging. Anyone who knows your IP address can readily find information about where you are. Indeed, even knowing who your SIP service provider is can reveal information about you that may be as good as revealing your identity. Consider, for example, a small enterprise with 10 employees. A call from someone in that enterprise narrows the set of callers down to 10 — not much privacy at all. Consequently, a true system for privacy requires a large worldwide interconnected mesh of loosely organized servers through which calls can be routed. Such a mesh would make it virtually impossible to ascertain the identity of a caller. This is exactly the kind of topology that P2P SIP can provide.

**Low Cost, Highly Reliable VoIP:** There is some belief that P2P SIP would reduce the cost of providing consumer VoIP services, since it has no centralized infrastructure. Furthermore, the lack of such infrastructure makes it highly robust, so that 'five 9s' of reliability can readily be achieved. Debate rages about whether this is really true. I believe that it is not. In practice, these P2P VoIP systems end up needing centralized servers for bootstrapping purposes and for firewall and NAT traversal. They are also required for the storage of sensitive data (such as voicemail or buddy lists). Furthermore, it's not clear that the costs of customer support are really any lower with P2P VoIP than any other kind of telephony system. Finally, a traditional SIP system, with traditional proxies, can be built with relatively little server infrastructure. Indeed, traditional SIP proxies are stateless and systems built on them are highly resilient as a consequence. Is a stateless SIP proxy more expensive to operate than a bootstrapping P2P server? It's far from clear that it is.

Finally, there are those who believe that the secret of Skype's success is its use of P2P technology. However, I believe this is not true at all. The secret of its success is that the product features a great user interface, works reliably, sounds great, and has a fabulous distribution channel. None of that has anything to do with P2P. PC-to-PC clients based on traditional SIP can be built with similar properties: the Gizmo Project is one such example.

All of that said, there is no doubt that P2P SIP has many important applications in the VoIP marketplace. If you take something as controversial as P2P and combine it with something as hot as VoIP, how can the results be anything but exciting?

*Jonathan Rosenberg is co-author of the original SIP specification (RFC 3261). He is currently a Cisco Fellow and Director of VoIP Service Provider Architecture for the Broadband Subscriber Applications Business Unit in the Voice Technology Group at* Cisco Systems. (define - news - alert)

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FEATURE ARTICLES

# SIP in Mobile Carrier Networks

By Richard "Zippy" Grigonis

The most distinguishing feature of modern 3G mobile wireless services over the 2G and 2.5G services of the past has been the emergence of multimedia. The original Short Messaging Service (SMS) could send mere 160-byte text messages from one mobile phone to another. In 2002, however, a combination of new handset hardware and a software extension to SMS gave us the Multimedia Messaging Service (MMS), which enables the transmission by users not just of rich text (complete with selected fonts and colors), but also of images (JPEGs, GIFs, etc.), audio (MP3, MIDI) and video (MPEG) from WAP sites or cameras built into the mobile phones.

The upsurge in multimedia's popularity and increased demand for sophisticated multimedia-based services in both the wireless and broadband wireline worlds led to the search for a common service architecture. While the 3G wireless networks were on the drawing boards, wireline carriers and equipment vendors had discovered and were beginning to adopt the Internet's IP packets and Voice-over-IP's SIP-based signaling and call control schemes. Therefore, it made sense that the new cellular networks could simply reuse these existing Internet Engineering Task Force (IETF) protocols, and by doing so, IP and SIP could now serve as the core of a superlative distributed common service architecture for both wireless and wireline networks, making possible quick and easy service creation, integration and usage in any and all environments (cellular, WiFi, DSL, cable, etc.). This idea is currently being perfected in the form of IMS (IP Multimedia Subsystem, the first full specification of which also appeared in 2002) that will bring SIP-based multimedia communications to any IP-based wireless or wireline network, and will lead to the future, unified world of Fixed-Mobile Convergence (FMC).

The upshot of all of this is that SIP becomes the standard signaling protocol/call control mechanism to support multimedia sessions in 3G and 4G networks and (presumably) beyond. Since SIP wasn't originally designed for the stringent bandwidth restrictions of mobile networks, various developments have appeared to help facilitate the deployment of SIP in wireless environments, such as SigComp ("Signaling Compression") which enables lossless compression and decompression of SIP messages ranging in size up to several thousand bytes.

SIP (define - news - alert) has actually been a part of some wireless networks for several years, appearing early on as the fundamental part of push-to-talk (PTT) services. In 2003, for example, Sprint PCS deployed their SIP-based Sprint Readylink PTT service. Sprint was soon followed by Unefon, a Mexican carrier. However, the question now becomes: can a more sophisticated "Mobile SIP" be deployed on a large scale, allowing users to roam with their varied services to any device or environment?

## SIP on the Move

Bill Lesley, Founder and CTO of Longboard (news - alert) (http://www.longboard.com), says: "I've been working with SIP in one form or another close to ten years now. Even in its early days it wasn't exactly a peer-to-peer protocol; there was always envisaged a server component, but the server was originally just going to find the person in the network you were trying to contact, and then everything else at that point would pretty much revert to peer-to-peer. The view of the academic community was that everything on the Internet was going to be free and everything would center on smart endpoints. I must admit I was recently attending a meeting where I opined how complex SIP has become. Quite a few things have been bolted onto the protocol over the years, for a variety of reasons: to make it work inside of specific networks, or just to correct its initial flaws."

"Longboard focused on building a SIP applications server," says Lesley. "We got into that business because, when we founded the company, we were looking at softswitching technology and in talking to a variety of people they said, 'Yes, Voice-over-IP is clearly where the industry is heading.' Others kept asking, 'Where are the features going to come from?' So, we started building an applications server that would bring some of the basic, required features into the network. We looked at the different protocols available for VoIP and we selected SIP as the one we were going to work with. We got off to a reasonably good start, built a platform and made available some basic services."

"SIP is at an interesting crossroads at the moment," says Lesley, "because SIP has been heavily embraced by the guys in the 3GPP for building IMS. This spurred the creation of many extensions to SIP so it could be used in a mobile environment. The IMS network architecture is quite complex, and there's still quite a few things that need to be understood in how these networks all fit together in an end-to-end perspective. It's really difficult, as the IMS architecture has grown along with the number of its components and their functions. Fewer and fewer people in the world actually have a complete understanding of how it all actually works."

"Even so, we should remember that there are many large networks out there that run using SIP as their basic protocol," says Lesley. "We shouldn't conclude that SIP itself is not mature enough to run in sizable networks. At Longboard, we've had considerable experience in the Japanese market, for example, where there are many large SIP networks. They use SIP not just for the interconnection of the long distance calls with media gateways linking to the backbone network, but in actually providing end user services over broadband connections. They're quite sizable networks, and SIP has been successfully used as the base protocol inside of them. It's worked very well in a multi-vendor environment. It's proven itself to be a very flexible, scalable protocol. So we've got some sizable, established networks in place where SIP is running today."

"I don't think there's anything in SIP that makes it onerous to work in the IMS environment," says Lesley. "Our application server runs in a regular SIP network, and one of the things that we looked at over a year ago is how we could also act as an application server inside of an IMS network; that required us to be compliant with the recommendations for what's called the ISC interface, which specifies the interface between the CSCF [Call State Control Function] and the application server in the IMS architecture. That interface is referenced as ISC, but it's actually a very SIP-based interface. When we analyzed that interface and its requirements we were pleasantly surprised to discover that there was basically very little difference between that and what we already supported inside of our platform, just because of the maturity of SIP and how it had evolved over time. With the migration from what had occurred in the original RFC for SIP [RFC 2543] to the newer one [RFC 3261] we saw that, by taking a rigorous approach in implementation, the number of new additions necessary to support an ISC interface were relatively few and minor."

"At the moment Longboard is focused on the Fixed-Mobile Convergence [FMC] space. We have solutions where you can take a voice call and seamlessly roam with the active call from a GSM network to a WiFi network, and *vice versa.* We use SIP as one of the call protocols for doing that. We're actually seeing many other companies in this arena using SIP as the protocol of choice. That work is going through standardization as part of 3GPP for IMS. In talking with other vendors in this space, I believe that the way we all use SIP is probably not too dissimilar. So it's reasonable to expect that with a little bit of cooperation between vendors, companies building SIP-based FMC solutions should be able to enjoy a reasonably high degree of interoperability since many of the SIP components already work today."

"When we first got started in this industry it was always a challenge making a new SIP endpoint work with a new SIP server," says Lesley, "but the usual case now when we receive SIP endpoint devices from a variety of vendors is that we just plug them in and they work out-of-the-box. There might be one or two minor things that need adjustment from the endpoint vendor's perspective, but we find that there's a much wider range of interoperability between endpoints and servers than ever before. Now we're looking at how we can ensure that there's a wider range of interoperability by focusing on the ways people build SIP-based solutions for FMC. It's all about how you use the messages within SIP to form particular applications. Generally, in the past there have been sets of 'best common practices' wherein you define the preferred mechanism of how you use SIP to build a particular service or feature. From what I've seen we've got to put those in place regarding FMC."

"There's a lot of activity now in what's being called Mobile VoIP, (define - news - alert) where you've got a wireless SIP client that can make calls," says Lesley. "Now, if you take a regular wireless VoIP client, there's no difference with a SIP server in the network, since the client can be plugged into the Ethernet socket in the wall in the way that we would make a normal call. So, there's an opportunity for hammering out an agreement on the interoperability mechanism for clients and servers in this space, to create an environment so that so that when you start a call using, say, SIP over WiFi, as the device moves out of range of the WiFi access points, you can seamlessly reestablish that call using the GSM or CDMA network. Also, the reverse, or how you can figure out an agreement on what the messaging should look like to both the mobile network and SIP messaging in order to reestablish the call as it roams back to a WiFi environment from the mobile network."

Lesley muses: "We're chatting with several other companies in this space, but two years ago these conversations would never have occurred, because we all thought that we had discovered the ideal 'secret recipe' and we weren't about to go around advertising to our competitors how we could make calls move from one network to another using SIP. I think we've all finally reached the level of maturity in the FMC industry where we realize that most of the companies that have been working in this space — especially the SIP-based space — have actually devised fairly similar solutions. So, it's much more interesting to look at how you can grow an industry by harmonizing the way that these SIP-based solutions work so that you can have growth through interoperability amongst solutions, as opposed to having what would mistakenly be labeled 'proprietary solutions'. I think it's much better to harmonize the mechanisms that you use across the industry so that the players in this space can actually rely on a good degree of interoperability amongst their solutions just as you would with things such as IP phones talking to SIP servers."

"My discussions with the vendors in this space boils down to, 'Yes, this is something we need to do, otherwise we can put ourselves in a position as an industry in FMC by saying that the standards-based activities going on in 3GPP are really the only 'real' standards out there, and because those standards aren't going to be completed until sometime in 2007, there won't be any standardized solutions, when in fact I think we can actually take what we've got to date — without requiring IMS — and actually get industry agreement on a common set of practices, so that we can establish interoperability across a greater number of vendors."

### Proof of Concept

With all this talk about Mobile SIP in a wireless network, the public hasn't actually seen some of the more interesting full-blown tests and demos until recently.

Take the demo recently presented (September 2006 at VON) by Reef Point Systems (http://www.reefpoint.com) makers of carrier-class Security Gateways for IMS/FMC networks (and which recently debuted their Universal Convergence Gateway at ITEXPO West in San Diego), and Spirent Communications (http://www.spirentcom.com) a provider of integrated performance analysis and service assurance systems for next-gen networking technology.

In the demo, Reef Point's iQ8000 and iQ4000 Security Gateways were deployed as IMS Security Gateways to provide protection for IMS voice and video calls, and were used in conjunction with Spirent's Landslide Fixed-Mobile Convergence network tester. The security gateway features showcased in the demo included dynamic stateful-session bandwidth, SIP-aware security and encrypted tunnels for media transfer.

Scott Poretsky, Director of System Quality Assurance at Reef Point Systems, says, "We demonstrated mobile IMS scaling and performance validation using Spirent's Landslide testing product — they had just introduced IMS testing functionality. As recently as two months ago, there was nothing on the market for actually doing IMS testing like this. Nothing could do IPsec or media, because the tests until now have been pure SIP-only. IMS takes SIP to the next level, and it has additional simulator requirements and additional protocols. So it's not just SIP through an IPsec VPN tunnel; it also has to do with the sequencing of the SIP signal."

"Because of the absolute absence of test tools for IMS," says Poretsky, "we looked around and decided that Spirent, one of the leading test equipment vendors, has the best potential for doing top-notch IMS testing, extending functionality to actually test SIP in an IMS network. We worked closely with their engineering team to help them develop an IMS tester which is absolutely fabulous. It emulates all of the components of a complete IMS topology, and then you can take any networking equipment as a device-under-test and insert it into this emulated topology. The Spirent box with two interfaces will test a device-under-test and actually figure out both its conformance and performance for IMS. Sprient is clearly the leader now in IMS testing."

"If you looked at our demo with the Reef Point (news - alert) security gateways in the middle, the Spirent (news - alert) Landslide connected to the mobile handsets on one side, and the Internet on the other," says Poretsky. "And the whole demo simulated and provided VoIP and video security for 50,000 mobile subscriber handsets. To do this we had to set up 100,000 IPsec tunnels. In the 3GPP IMS standard you need two IPsec tunnels per handset: one tunnel is for calls originated from the mobile handset, and the other is for calls originated from the public network to the mobile handset. So, the Landslide emulated 50,000 mobile handsets. The SIP signals are encrypted through the IPsec tunnels and out our security gateway that's also terminating the IPsec tunnels, then they're forwarded to what would be the other, public side of the IMS network, where the traffic can get onto the service provider's network. That's also emulated by the Landslide test equipment."

Poretsky beams: "We managed to prove with two boxes — the Sprient Landslide and the Reef Point Security Gateway — that the IMS concepts in the mobile 3GPP standard work."

(Editor's Note: For more information about this test, see the SIP Testing article elsewhere in this issue.)

Certainly the SIP that supports mobility must be a great deal more clever than its Internet-based cousin. The application layer must be prepared to suddenly encounter new networks (i.e., movement detection) and it always must maintain knowledge of the current (active) interface's address. But tests and demos such as those by Reef Point and Spirent indicate that SIP is prepared to bring forth a new generation of exciting mobile services.

*Richard Grigonis is Executive Editor of TMC's IP Communications Group.*


# SIP Anywhere: A Look at Enterprise Mobility, à la SIP

By Hal Clark

The combination of increased availability of a wide range of mobile devices incorporating SIP User Agents and a focus on SIP evolution to meet inter-networking requirements yields significant opportunities for developers of mobile enterprise applications. An enterprise is a community of resources directed towards positive business objectives. This article examines how SIP enables a member to actively participate in that community without sitting at a fixed location or being tied to a specific network so that transaction times can be reduced by action where the member is working.

## Workflow Pauses Impair Business Efficiency

Programs to improve financial performance in the current business cycle emphasize revenue production per employee — a measure of efficiency — in managing the cost side of the equation for profitability. Failing to complete transactions within the duration window implied by transaction rate targets yields both lower revenue and higher costs per transaction — a double reduction of profitability.

Many factors effect the time it takes to complete individual transactions successfully. The focus of investigation here is the time it takes to communicate with community members who execute component tasks within a transaction. Traditionally, delays in reaching the next person in the string of tasks to complete a transaction were minimized by having a sufficient number of specialized staff on hand to handle the workload and having long completion windows for each transaction. Today's business environment cannot afford narrowly-specialized staffing and demands increasingly greater skills or knowledge to complete transactions. This has led to a workforce with blended skill-sets performing multiple roles. Both business demands for high transaction rates and customer demands for immediate gratification drive the enterprise community to short transaction duration. While the community as a whole has sufficient capacity to achieve transaction volume goals, an individual transaction may pause, due to unavailability of the appropriate skill or role resource.

The blocking factor of overlapping demands for specific skills and roles is compounded by the unreachability of seemingly available community members. Today's telephone systems tie individuals to a fixed spot so any physical movement away from the phone set makes the individual unreachable, a condition only detectable by the lack of answered phone calls. Workarounds such as voicemail, email, call forwarding, Find-Me services, cell phones, instant messaging and WiFi phones have evolved and seek to advance transaction processing as quickly as possible. However, they do not offer an integrated and coordinated approach to reducing transaction times.

## SIP Enables Workflow Mobility

SIP is fundamentally about connecting communications sessions among users wherever they are available on the IP network using methods available to the user interface device. Deployment of SIP-based systems removes the constraints of physical location for establishing communications sessions and provides the basis for a coordinated approach to connecting individuals under varying reachability conditions.

The focus of SIP is the communications user and not the physical line. System software can track the user's availability status as well as determine how the user is reachable at any given moment. Many SIP-based systems merely reimplement historical concepts of availability from legacy phone systems — registration of a SIP phone is equivalent to user availability to answer calls, and Find Me services are replicated as workarounds to reach individuals at different locations. Even at this rudimentary level, SIP-based systems can offer applications finer-grained information on user availability as well as a broader reach to users on a wide range of generally available devices and software.

Advanced implementations of SIP-based systems can be capable of far greater precision of availability indication and reachability. Interfaced to applications via a variety of methods ranging from SIP stack interworking to Web services APIs, SIP infrastructure both consumes and exposes additional information regarding user activities that reflect on availability for defined tasks and roles. For instance, the SIP-based system can use availability information to modify call treatments based on calendar appointments. The automated exchange of information promotes more accurate determination of a user's ability to respond so the system can return context-appropriate messages, advance to an alternate resource, or escalate the transaction as appropriate. Escalations may be hierarchical in a management chain, or may use alternate communications methods to reach the targeted resource. For example, an IM or SMS alert may also be sent through the SIP system with response options to accept the call, escalate, send reply message or divert to voicemail. These alternative actions are dependent only on the software capabilities or reachability of the users' networked device.

Call Center ACDs (Automated Call Distributors) have long had the ability to change the availability of a line by changing the phone operating mode via a key press or application log-in to an agent role with assigned skills. The applicability of this operating scenario is generally limited to the Call Center application stack as well as proprietary phone instruments associated with the legacy environment. SIP provides a platform to extend communications to community members based on skills and roles across the entire network. Again, integration with applications offers flexible workflow integration not envisioned for currently deployed communication environments.

SIP-based systems are easily implemented as software services in data center operations similar to data applications. SIP also partners well in Web services-style integration with workflow applications, leading to easily implemented and modified applications that keep up with business changes. Service Oriented Architectures (SOAs) leverage this level of integration to orchestrate multiple applications into comprehensive business processes. Interestingly, this flexible application interface enables creative ways of determining user availability and automates policy and preference decisions that maximize productivity of the enterprise community, regardless of user location.

Mobility in this environment varies from "hallway warriors" within a single building to "nomads" traveling around the globe.

Software SIP clients extend the conceptual model to include the user in managing their own availability and reachability preferences. While users may actually use existing PBX phones during migration from a legacy telephony infrastructure, improved availability management promotes convergence to a minimum number of devices with greater productivity and availability capabilities.

## Business Efficiency Everywhere

Mobility in this environment varies from "hallway warriors" within a single building to "nomads" traveling around the globe. While SIP supports dynamic movement from point-to-point, office-to-office, office-to-conference room, and site-to-site, wireless technologies fill-in connectivity while in motion between points. WiFi access to local networks maintains connectivity within buildings where wide-area radio coverage is inconsistent. Wide area wireless provides reachability when out of WiFi range. Untethering a PC via wireless carries all application resources along with it, but is rather inconvenient to use while in motion. SIP wireless phones are easier to use on-the-go within limited areas, but are not universal instruments, except in specific voice applications.

Converged mobile devices coming to market combine a wide area switched mobile phone (i.e., GSM or CDMA) with a PDA and WiFi that can support SIP communications concurrently. These devices offer equipment options to enterprises for a universal communications device that can be used for community activities wherever the member happens to be. The combined communications capabilities can replace separate desktop phones, cell phones, mobile email devices and PDAs with a single device for many community members. Leveraging enterprise SIP infrastructure, the device participates in the enterprise phone network and can manage the cellular phone calls as extensions to the SIP infrastructure providing the advantages of fixed-mobile convergence today. The SIP-based system deployed to enable this wide-ranging mobility for an enterprise community will support the migration of 3GPP and PSTN networks to IMS over the coming years.

The form factor of converged devices ranges from phone-like to full-blown PDA to fit individual application requirements. The fact that most of these devices support at least messaging, enterprise calendaring and email as well as web browsing means the devices can be used in many SOA-orchestrated business processes. The data network interface to both application portals and SIP infrastructure frees business analysts and application process designers to gain precious response time improvements from community members without tying them down to specific locations. The ability of a SIP system to determine availability of mobile users and to incorporate multiple methods to reach these users with essential communications improves the customer relationship, whether the transaction is a sale approval or a critical response to a medical condition. In these cases, simple transactions do not have to delay revenue recognition and critical responses are not dependent on key personnel being at a specific location.

SIP is a key technology for business communications platforms, offering unified communications and deployed to serve increasingly mobile enterprise communities in their efforts to reduce transaction delays that limit business productivity. Higher revenue, increased profit, and even better lifestyles can result from fully leveraging the capabilities of SIP with IT applications.

*Hall Clark is a Senior Product Manager at BlueNote Networks.* (news - alert) *For more information, please visit the company online at http://www.bluenote.com.*

## ADVERTISING INDEX


# SIP & Mobility: Bringing Radio into the 21st Century

By Doug Hall

The adoption of SIP in the mainstream telephony world has numerous and proven benefits: open standards, a rich feature set, and readily available equipment from multiple vendors. Now SIP brings all of these benefits to land mobile radio (LMR) users plus entirely new interfacing and interoperability options. This article explores how to interface radio equipment with SIP infrastructure and explains the features radio users can expect to enjoy.

Analog telephone users have been a part of the SIP world for a while now, thanks to the Analog Telephone Adapter (ATA) which interfaces their analog telephone equipment to a SIP network. Intrinsic within POTS phones are a number of characteristics that don't correlate with the IP network realm, such as loop current, ring voltage, and in-band signaling. The ATA resolves these peculiarities, allowing analog telephone users to make and receive SIP calls unaware of the activity transpiring under the ATA's hood. Likewise, radio brings its own set of idiosyncrasies to the SIP world.

Most radios, such as those used by public safety agencies, are half-duplex devices, meaning that they can't simultaneously transmit and receive. These radios require user intervention to change from receive to transmit, usually via a Push-to-Talk (PTT) switch. Telephones, being full-duplex by nature, have no such requirement. Furthermore, with radios, any signaling via DTMF or other tones is usually done in-band. What's needed is an ATA-like device that gracefully handles all the nuances of radio communications.

An example of such a device is the ARA-1, part of a family of ARA™ (Analog Radio Adapter) devices from Raytheon's JPS Communications. The ARA-1 Radio-to-SIP interface allows radios to become SIP endpoints. Like an ATA, the ARA-1 includes a 10/100BaseT Ethernet interface to the SIP network. However, on the analog side, its only similarity to the ATA is that its users can communicate, blissfully unaware of how the interface does its magic.

The ARA-1's analog port provides an interface for the signals commonly found on radio equipment: transmit and receive audio and PTT and Squelch signals. (Editor's Note: Squelch involves turning off the radio's speaker when there is no signal on the tuned frequency, and turning on the speaker again when the received signal is sufficiently strong.) Adjustments are provided for the audio levels and the type of squelch indication (valid received signal present) that will be used. These adjustments handle differences in interface requirements between radio makes and models. Some radios expect mic-level input (TX) audio, while others expect line-level audio.

A radio's squelch indication output (COR) signal can be active-high, active-low, or just plain non-existent. When no signal is provided, the ARA-1 can derive one by examining the incoming receive audio with a voice-activated switching (VOX) algorithm. Knowing when a valid signal is being received by the radio allows the ARA-1 to preserve network bandwidth by not streaming audio packets when there is nothing valid to send. Or, if required, the unit can also stream silence.

The final radio interface signal is its PTT input. Let's assume that a SIP phone is connected over the network to an ARA-1 and associated radio. SIP phones don't have PTT switches, so again there's nothing for the ARA-1 to work with other than the incoming network audio. Whenever the SIP phone user talks, the radio will transmit this speech, and when the phone user stops, the radio switches back to the receive mode. The ARA-1 provides an adjustable "hangtime" which keeps the transmitter active during momentary pauses between words and syllables, as it's not good to try to move too quickly between the receive and transmit states.

If only it were that simple. . . but as any telephone user knows, there are other noises on the phone line besides speech. Background noise, static, and a variety of "pops" and "clicks" of unknown origin — all are audible and could inadvertently and momentarily activate the transmitter. The ARA-1 resolves this problem by providing a Voice Modulation Recognition (VMR) algorithm. VMR examines the frequency content of the incoming audio, seeking concentrations of energy that are characteristic of human speech. It then goes a step further by ignoring those concentrations of energy that don't occur at the same rate as syllables in speech. This allows the algorithm to reject noise and other non-speech signals, so the radio transmitter is activated only when someone is actually talking.

Like an ATA, the ARA-1 can accept an incoming call (a SIP Invite) and be programmed to answer automatically after a specified time. A much more difficult problem to address is how does a radio user initiate a call? It's easy to visualize a call being put through by a radio with a DTMF keypad, but the majority of LMR radios don't have one. The only user controls available on most of the handheld radios commonly used by public safety agencies are a PTT button, a speaker, and volume and channel knobs. Fortunately the ARA-1 can accommodate outgoing calls from radios limited to these few simple control mechanisms.

Before we go any further, we need to properly visualize a radio-to-SIP call. An analog telephone user who makes a call through an ATA simply picks up the phone, dials the extension, and is directly connected. In contrast, a radio user with DTMF capability transmits DTMF "digits" over the air to another radio. This is the one that's cabled to the ARA-1 that provides the SIP network interface. Any return network audio is transmitted over the air and received by any number of radio users "in the field" who have radios of the right frequency. So an ARA-1 interfaces an entire radio system, not a single user.

The ARA-1 can be programmed to initiate calls based on pre-programmed DTMF sequences received from those radios equipped with DTMF capability. For the rest, the ARA-1 provides a feature called COR Cadence signaling. A radio's COR output signal goes to an active state whenever a signal is being received by the radio. Radios in the field without DTMF capability can activate their PTT switches at a preset rate; the resulting RF pulses are detected by the radio cabled to the ARA-1. This radio's COR output pulses in time with the incoming RF and the ARA-1 detects and compares these pulses with an internal "speed dial" that matches these COR cadence sequences with stored SIP PBX extensions or IP addresses.

For example, pressing and releasing the PTT button 5 times within a 2 second period could instruct the ARA-1 to dial a pre-programmed extension. The numbers to be dialed are programmable, and the timing of the COR cadence can be adjusted to suit specific radio requirements. DTMF and COR Cadence sequences can also be used to disconnect a call.

Installing and provisioning an ARA-1 is similar to installing an ATA, though the radio interface process is more involved than plugging in a telephone's RJ11 cable. There are a number of network, SIP, and radio settings which can be viewed and optimized with a web browser. Ready-made ARA-to-radio cables are available from JPS Communications for over 100 different radio models, or the user can construct a cable based on information in the ARA-1 manual.

The ARA-1 includes some additional user-friendly features such as the ability to "speak" its IP address and other network settings over the radio interface. Initiated by a preset DTMF or COR Cadence sequence, this feature helps determine the address when it has been assigned by DHCP. On the network side, STUN support is provided for operation behind a NAT device.

While the ability to interface individual radios as SIP endpoint devices has tremendous utility, features already existing within SIP allow a valuable bonus: Radio interoperability! Multiple radios can be interconnected through a SIP conference call just as regular telephones could be. A SIP conference call might include a VHF radio, a UHF radio, and an 800MHz trunked radio, all connected to the SIP phone system through ARA-1 devices. Any radio traffic received on the 800 MHz radio would be retransmitted to the VHF and UHF radios. The radios can join the conference through the use of DTMF or COR Cadence signaling, or they can be added to a conference by an operator. In short, they enjoy all the benefits that a SIP phone system brings to ordinary SIP phones, such as call forwarding, call logging, and any existing or yet-to-be-imagined feature that's applicable to radio communications (or can be adapted to it).

As more and more organizations build out their SIP infrastructure the applicability of this technology to radio communications will become more and more apparent. The old saying "If you can't beat 'em, join 'em," is certainly appropriate. Why try to make telephones that act like radios, when with SIP and devices such as the ARA-1 we have the best of all worlds: Since everyone already knows how to use a telephone, why not make radios act like telephones?

*Doug Hall is the Senior Scientist at JPS Communications* (news - alert) *(http://www.jps.com), a wholly-owned subsidiary of Raytheon Company.*

FEATURE ARTICLES

# SIP Breaks the Cost and Complexity Barriers

By Sanjeev Sawai

**Computer telephony integration (CTI) has been a bedrock technology in advanced customer service environments for more than 25 years. The Session Initiation Protocol (SIP) is the best thing to happen to CTI since it was invented — even if it eliminates CTI as we know it today.**

CTI is a critical function that makes automated and agent-based customer interactions efficient and productive. CTI infrastructures route calls and automate information retrieval to help customer service reps resolve issues accurately and without transferring callers or leaving them on hold. This is the "one and done" nirvana that call center managers strive for. SIP promises to change the way in which applications, agents, systems and networks work together to provide CTI capabilities.

SIP's ability to take over call setup, routing and teardown will enable companies to infuse CTI functions throughout IT infrastructures instead of concentrating them in a proprietary hardware/software layer, as they must today. Applications will use SIP (define - news - alert) commands to perform call-related tasks and non-call related functions such as presence management. When that happens, CTI will be far more widely implemented than it is today, even though it will have disappeared as a discrete function.

## The BMW of Customer Service Technologies

CTI is one of customer service technology's crown jewels. It enables contact centers to use IT to its highest potential through its two primary functions of automating customer information retrieval and coordinating voice and data traffic.

When a call comes in to a CTI-enabled environment, CTI servers use caller ID information from PBXs to retrieve customer profile information — name, address, transaction history, previous service issues, credit reports, etc. — from applications within and outside the contact center. CTI servers load the information into a screen pop, pair it with the voice call and send them as a package to a service rep. This automation eliminates all the time wasted when a sales rep has to take information over the phone, enter it into their call center application, and search outside databases and applications for additional information they need to close the call. Also, if they have to transfer the caller to another rep, the information follows the voice call so the customer does not have to repeat him/herself. These and other CTI functions are the difference between good customer service and outstanding customer service that helps bring customers in the door and keep them there. CTI functions improve customer loyalty by cutting wait times and making customer service reps more productive. They're also beyond the reach of all but the best-heeled companies.

CTI middleware infrastructures are proprietary and complex. Entry prices are high and implementation costs can be even higher. Integrating applications into a CTI environment is a series of time-consuming custom jobs because proprietary protocols prevent the knowledge gained from integrating one application from carrying over to the next. Even large companies that can afford CTI systems often use them only for select customer segments because using them for all customers is too expensive. SIP's move into the IT infrastructure will open CTI functions to almost any company, at reasonable price and complexity levels.

### SIP Expands the CTI World

The basic function of enabling a voice device to communicate with a data network has, until now, required the specialized CTI middleware layer to translate TDM (Time Division Multiplexed) voice traffic into packet traffic so the customer endpoint could communicate with back-end applications. The combination of SIP and an all-IP infrastructure eliminates CTI as a distinct layer in the IT infrastructure because SIP enables customer endpoints — cell phones, laptops, PDAs — to communicate directly with IT resources.

SIP assumes the call setup, routing and teardown functions once locked in the proprietary CTI layer. In all-IP networks, calls will enter the application infrastructure as Voice-over-IP (VoIP) traffic and travel to a SIP proxy server. The SIP proxy server will initiate sessions with the necessary applications to perform call routing and information lookups that once required a CTI server. In this way, CTI will dissolve into the larger fabric of call routing. The functionality for specific tasks such as transferring a call within the contact center will be embedded directly into applications.

The immediate benefit from SIP supplanting proprietary CTI is cost. Standards-based SIP proxy servers are inherently less expensive to buy and maintain than proprietary CTI servers.

SIP will also ensure that the current centralized architectures are decomposed into distributed system components with specific capabilities such as presence state servers, rules engines, routing systems and standard SIP switching systems. All this infrastructure will seamlessly integrate with the business applications and web infrastructure that already exists in more corporate IT departments.

The immediate benefit from SIP supplanting proprietary CTI is cost. Standards-based SIP proxy servers are inherently less expensive to buy and maintain than proprietary CTI servers. Companies can implement SIP proxy servers on standard hardware, which helps make better use of existing resources. Integration is much less costly and complex because all the input and output in the network is one standard protocol.

Standardization will also drop the CTI cost barrier for small- and medium-sized businesses (SMBs). In today's environment, it isn't profitable for CTI vendors to develop CTI solutions for SMBs. Re-writing applications in proprietary code to interoperate with the applications found in smaller companies is expensive, and there aren't enough smaller companies that can afford them. However, when all applications are written for all-IP environments, it will be cost effective for vendors to adapt their high-end products to smaller companies' needs.

SIP-enabled environments with integrated CTI functions are still two to five years away from widespread implementation. Many difficult questions remain to be resolved — security, authentication and authorization chief among them. Companies must also migrate from TDM voice traffic to VoIP.

Nevertheless, the market is definitely moving towards SIP in the contact center. The TDM-to-VoIP migration is underway as transitional platforms reach the market — PBXs that can handle both TDM and IP traffic then evolve to all-IP, for example. SIP is already making its way into the IT infrastructure. BEA Systems has added SIP support to its Java platform and Oracle has purchased the Swedish company HotSip to add SIP functionality to its middleware. Solution providers such as Cisco, Avaya, Nortel, Envox and Genesys have announced plans to develop SIP-enabled products. The first generation of servers that can perform CTI functions in a SIP environment — called either SIP proxy servers or SIP application servers — have recently come onto the market.

The companies investing in SIP-enabled product development today aren't the kind that rush down blind alleys. Delivering CTI functions through SIP may be a futuristic idea, but it's neither far-fetched nor likely to fizzle out before it gains a foothold in the business world, primarily because of its potential to raise customer service levels. Customer service is an increasingly important differentiator in competitive markets. Companies that seize the opportunity that SIP offers them to implement CTI functions will win in those markets.

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# SIP Testing Gets Serious

By Richard "Zippy" Grigonis

SIP testing, like VoIP testing itself, was considered something exotic. Now that SIP has pretty much supplanted H.323 as the call control signaling protocol of choice, equipment vendors are scrambling to find labs and test equipment that can do full-blown tests of IP communications devices and applications running in a SIP-enabled network environment.

One of the great names in telecom-related testing over the past 10 years has been Empirix (news - alert) (http://www.empirix.com), formerly Hammer, makers of the immensely varied Hammer product line that includes the Hammer NXT, used by next-gen network equipment makers and service providers to generate sufficient 'real-world' traffic to simulate real-world load and stress conditions so as to ensure proper operation of VoIP devices and next-gen applications for network deployment.

Recently, Speakeasy (news - alert) (http://www.speakeasy.net), one of the nation's largest independent broadband services companies, has begun monitoring its VoIP network with Empirix' Hammer XMS, a carrier-class monitoring and analysis solution incorporates signaling and media analysis and call correlation capabilities, capable of evaluating live network traffic to provide real-time network performance assessment.

Back down at the enterprise, the Hammer VoIP Test Solution for Enterprises handles pre-deployments, smoothes the way for device additions to a network, and takes care of troubleshooting. The Solution actually consists of three other Empirix products: the Hammer FX-IP that generates test IP calls and assesses voice quality; the Hammer CallMaster graphical scripting and reporting tool for creating test call flows; and the Hammer Call Analyzer, a diagnostics and troubleshooting solution VoIP networks.

One fellow who's up to his ears in Empirix test equipment is long-time testing guru Chris Bajorek, Director and Founder of CT Labs (http://www.ct-labs.com), a full-service testing and product analysis firm specializing in testing services to converged communications product manufacturers and next-gen network service providers. CT Labs is one of the few organizations that can do complete testing of SIP-related products."

"CT Labs (news - alert) is an independent operating unit of Empirix," says Bajorek. "The reason for that distinction is that we engage primarily with network equipment and VoIP infrastructure manufacturers, the nature of our engagements most often are pre-release in nature and the types of tests that we do are such that the vendors would prefer that everything we discover be held in confidence. There's some benefit to being physically apart from our headquarters in Bedford, Massachusetts."

"The nature of the types of tests we can now do involving SIP spans up to hundreds of thousands of simultaneous SIP calls," says Bajork. "This gives us the capacity for doing testing on carrier-grade components used in building next-gen VoIP networks. We've actually done a number of interesting tests in that regard, some of which have been self-published by the manufacturers with which we've engaged directly."

Bajorek elaborates: "Our SIP testing platform offers a combination of capabilities. We can test different classes of equipment. It really breaks down into three different domains: First, there's the residential environment where we test things such as analog terminal adapters [ATAs], VoIP soft phones, hard phones and residential routers. Second, there's the enterprise level, with IP-PBXs, contact center equipment, VoIP phones, firewalls, intrusion prevention devices, media servers, and things like that. Third and finally, at the carrier level we can test everything from softswitches to session border controllers, media servers, proxy servers, media gateways, and so forth."

"Obviously, the one major distinction between those three product categories relates to density and capacity," says Bajorek. "When you're testing residential products, a full-blown load test on a terminal adapter device usually involves just two lines. That's not too hard to handle. However, when you test at the top of the food chain and start dealing with things such as a high availability session border controller that's designed to handle hundreds of thousands of calls at a time, you soon discover that it takes quite a bit of equipment to do that well. Fortunately, our lab can handle it."

"So, our SIP testing facilities are a combination of testing products that we have in our lab that enables us to do tests across a wide range and down into various things within each domain," says Bajorek. "It's not just about testing for load and stress, which is the classic test we started out doing nine years ago. There's a lot more to testing now, obviously."

"Even so," says Bajork, "Things that we started testing for nine years ago that were important in the first generation of VoIP products, are still every bit as important. For example, it's still extremely important to verify voice quality in VoIP-related equipment and software. Obviously, SIP is the reigning standard for VoIP phone calls and next-gen networks. So it occupies a better part of 95 percent of the work we do in the lab right now."

"As for verifying voice quality in a SIP environment, it's interesting to see how things have changed over time," says Bajorek. "Starting Day One at our labs we did tests of VoIP gateways, since that was the way your call traffic got from TDM to some kind of an IP network. We do less gateway testing now, but we do a lot of voice quality testing across a wide range of devices — everything from SIP soft and hard phones to SIP PBXs, to SIP-aware firewalls and SIP-aware session border controllers. Anything that media either originates from or flows through, can potentially be negatively impacted by the device being used if traffic levels reach peak levels."

Bajorek continues: "So what could, say, a firewall possibly do to a media stream flowing through it? The answer is, if you're driving that firewall under high loads — any of these devices have a rated maximum number of connections or amount of actual IP traffic — then the processing power starts getting strained, and you can start imparting everything from packet loss to jitter to the media stream, which will have a direct negative effect on voice quality. We've developed tools and techniques in the lab for not only sending high quantities of SIP-based calls through these devices, but also other types of traffic too that we call 'Internet mixed' traffic. Using a special generator we can send HTTP and FTP sessions as well as SMTP and POP3 traffic — all those different kinds of traffic that might come through a normal corporate firewall when you're going out over the net and surfing, say, HTTP pages. For a firewall device, that's an example of where you'd want to do that kind of "Internet mixed" traffic generation in addition to just pure VoIP traffic."

## Let's Make a Deal

"The way we bill for our testing work is usually based on the estimated number of hours for the project," says Bajorek. "It can vary. These tests can range from quick-and-straightforward to complex-and-long. It depends on the nature of the product and whether it's going to have an early or a late release."

"We realized long ago that there was something that more and more was becoming an issue," says Bajorek. "Namely, what happens if customers want to test against different third-party devices? There are different ways you can address that need. One way is that you can go out and buy one each of every kind of third-party device. You could buy four different softswitches, for example. We've taken a different approach. Instead of obtaining all of the third-party devices that could possibly connect in some way the device undergoing testing, we've taken the approach of using a platform to emulate different devices in our lab. We've virtualized the third-party devices that you'd normally need in your lab for the tests. Think of the power of doing that. When you get past the point of needing to spend some development time creating the environments that emulate what, for example, a particular type of softswitch might do, once you have the emulation, then you can run it and gain the additional benefit of being able to see intermediate states during a test to which you might not have had access if you simply used the actual third-party product."

"Ironically, using the real product gives the test more of a 'black box' flavor, and you are stuck hoping that the device is doing what it's supposed to do," says Bajorek. "We continue to improve our device emulation platform, and it will give us more and more ability to test against many different devices that are all really staged on a single platform. This is a pretty neat, new concept. Empirix actually offers a platform that does this. We think it's a powerful model, especially for a test lab such as us, to be able to use moving forward. Again, if a customer wants to test against different types of devices, we can now do that merely by selecting the appropriate profile and running that emulation."

"One of the things we're using that platform for right now is emulating large communities of access network users for purposes of doing registrations," says Bajorek. "So let's say you're making a product that allows you as a service provider to build a VoIP network. One of the things you want to do is verify the performance of your staged network to handle hundreds of thousands of users having SIP phones that are registering on a SIP server. Perhaps some are doing calls, but the vast majority of your user community is just sitting there typically registering once an hour, sometimes more often, perhaps as often as every 15 to 20 seconds."

"It turns out that when you turn on authentication and have, say, 250,000 users that are out there basically ready to receive calls, it takes a fair amount of processing power to handle those requests," says Bajorek. "Thus, we're using the emulation platform for emulating very large numbers of users in some of these tests. It's a fairly new tool, but it's turning out to be really valuable. That's an example of something we were not able to easily do in the recent past, but it's now possible, thanks to these new tools."

"The development of all these new tools are driven by the new demands coming from greater numbers of deployed VoIP networks that are mostly SIP-based," says Bajorek. "And we're not even talking yet about what IMS is bringing to the table, but you can imagine the sophistication and complexities that will be involved."

> "These SIP attacks are really a very hot area right now," says Bajorek.

### Trends in SIP Testing

"At the moment we're getting many requests for validating platforms against all sorts of SIP-specific attacks," says Bajorek. "We have a proprietary platform in our lab that allows us to simulate very high rates of attack. For example, let's say you have a session border controller [SBC] and, among other things, you want to validate the SBC's ability to resist attacks at very high rates — gigabit wire rate — and protect networks from those types of external attacks. We have a tool that allows us to do that. There's quite a creative variety of different types of attacks that 'denial of service-intending' attackers will attempt to launch. It's interesting to see how different devices respond to these attacks. Some manufacturers are quite surprised and others not. It's not something that's been done a great deal, and our industry has been in a bit of a 'grace period' in terms of the frequency of real-world attacks it has experienced. But such attacks are definitely on the rise. It hasn't reached a crescendo pitch yet, but it's coming and more and more vendors are aware that they better do something about this and make sure their products can withstand some of these attacks."

Bajorek adds, "We're happy to be able to provide them with some of these test capabilities, such as being able to launch a 'SIP response spoof flood' — that's a mouthful — with packet rates exceeding 140,000 packets per second, which is quite a nasty attack against devices such as these. In the case of a session border controller, if it sees a SIP INVITE, for example, it can't just ignore that and pass it on through, it has to consider 'Okay, here's a SIP INVITE. Is this from a valid registered user? If so, do I allow it to go through?' So, every time the SBC sees a SIP INVITE it must deep-inspect that request and verify that it's a valid request for service, and then let it go through. That takes processing power. Therefore, a good way to test the limits of these devices is to flood them with a bunch of packets that look valid from all external respects, but may not actually be so. It's the job of these devices, whether they're firewalls or SBCs, or intrusion prevention devices, to only allow the valid packets through, even if the attackers are spoofing valid addresses that in the past have been used to create real SIP calls."

"These tests are done with two types of traffic," says Bajorek. "One, we send legitimate SIP traffic that the devices are supposed to recognize and allow through. We even validate voice quality during these attacks. But at the same time, we generate these different types of attacks and then monitor various things: the success, or lack thereof, of the valid SIP calls; the voice quality of the media streams; and we also monitor on the protected network side to verify that none of the attack packets make it through to the far end. To the degree that they do, the test will have failed in some way."

"It's always interesting to see how different devices respond," says Bajorek. "We've been able to demonstrate to most of our vendors where their limits are situated.

That's really the goal of any kind of test such as this."

"These SIP attacks are really a very hot area right now," says Bajorek. "The test requires not only the attack packet traffic but also very strong capabilities for sending the standards-based SIP calls through, and sometimes at very high call rates. We rely on the Empirix platforms for that and they do very well. Ultimately, it's a 'combination test' in the sense of the equipment needed to run it."

### Freeware SIP Testing

Scott Poretsky, is Director of System Quality Assurance at Reef Point Systems (http://www.reefpoint.com) the company that offers such things as security gateways and the Universal Convergence Gateway™ platform for service provider fixed-mobile convergence (FMC) networking.

"Clearly the SIP equipment vendors and the IMS equipment vendors are way ahead of test equipment vendors," say Poretsky. "So that's something we have to deal with, and we must come up with creative and innovative testing solutions in our labs so that we can properly test our products. There's a great freeware tool we use called SIPp. It's a free test tool / traffic generator for the SIP protocol available from Source Forge (news - alert) (http://www.sourceforge.net). It's commonly used in the industry. It's very often used in research labs, especially in universities, because it is freeware. It's excellent but it has its limitations — it doesn't scale well with the RTP [Real-Time Protocol] media and it doesn't do media testing, so it's strictly for control plane testing. I'm referring here to the SIP messaging — you set up the call but you really can't do much with the media. It also hogs the resources of the Linux machine it runs on. The thing with SIP TCP [Transmission Control Protocol] is that it really hogs system resources. So, you really have to test with it using only SIP UDP [User Datagram Protocol]. If you're doing SIP UDP control plane testing, it's an excellent tool, and we did make use of it. But when you're testing a device, if you push SIPp too hard, the resulting score you get will be lower because you're limited by the tool itself. It's not a limitation of the device under test's [DUT's] performance. That means that in a lab situation it's not good enough to solely use SIPp."

"Instead, there are even better commercial tools out there that are coming from companies such as Spirent Communications [http://www.spirent.com] with their Landslide product, and Ixia (news - alert) [http://www.ixia.com] is also releasing interesting tools," says Poretsky. "Where some commercial software packages fall short is with IMS (IP Multimedia Subsystem). Until now we've dealt with strictly SIP test tools, and now we need to be testing IMS, which has a lot of additional functionality and complexity, where you have IPsec and SIP working in conjunction with SIP signaling that's encrypted through IPsec tunnels. This is really generating demand for brand-new hardware technology in the test lab forms."

"Despite the lack of IMS test tools, we worked with Spirent, that offers the Landslide product," says Poretsky. "Landslide can take a device-under-test and actually generate its conformance for IMS and also its performance for IMS."

"Reef Point (news - alert) and Spirent did a demonstration at a recent VON show that demonstrated 50,000 simultaneous SIP-enabled mobile stations calls running in conjunction with our Reef Point Security Gateway and the Spirent Landslide," says Poretsky. "Two IPsec tunnels are necessary coming in and out of each virtual mobile set under the 3GPP IMS standard, so 100,000 IPsec tunnels were necessary. The test was a major one since it basically proved that IMS functions and concepts as specified by the 3GPP standard actually work."

"We then actually did SIP from the Spirent box through the Reef Point Security Gateway that has a SIP ALG [Application Level Gateway] for its firewall," says Poretsky. "The Spirent (news - alert) box on one interface sent out IPsec traffic to the Reef Point Security Gateway that makes the security gateway think that there's really 50,000 simultaneous mobile stations, not just one Spirent box. So SIP signaling ran through the IPsec tunnels encrypted, got unencrypted at our box, where authentication also occurred for access to the network. The SIP signaling gets forwarded to the P-CSCF [Proxy-Call Session Control Function, a SIP proxy that is the first point of contact for the IMS terminal]. The Spirent box also emulated on another interface the P-CSCF. I believe we're the first to do that, where you go from a mobile station with two IPsec tunnels through a firewall with a SIP ALG, to a P-CSCF. And we even did it at scale — 50,000 simultaneous mobile stations and SIP sessions. The SIP sessions were in the IPsec tunnels for the calls originating from the mobile handset. So the Spirent box had its emulated mobile handsets signal the SIP through our security gateway, which terminates the IPsec tunnels and has a SIP-aware stateful firewall. But it's not like a session border controller because it doesn't terminate the SIP. That's why the Spirent box in the demo also emulated the P-CSCF out of the other interface."

"At Reef Point, we, more than any other vendor, put very strict requirements on high performance," says Poretsky, "because our firewall can do one million stateful sessions. We support 150,000 simultaneous registered mobile subscribers, with support for up to a million simultaneous media flow connections. We're talking about a huge scale with many protocols running simultaneously and with the signaling per IMS, all running in a single chassis. Note that session border controllers support only one tenth that capacity."

So, any way you look at it, SIP testing is increasing in both scope and sophistication.

*Richard Grigonis is Executive Editor of TMC's IP Communications Group.*

Q & A

# 60 Seconds with Professor Carol Davids

**Carol Davids is Director of the VoIP Laboratory at the Illinois Institute of Technology (IIT) in Chicago. She worked in telecom for over 30 years: 22 years at Illinois Bell/Ameritech, where her projects included engineering packet switched data networks and gateways to the Interexchange carriers. For the past decade she has focused on methods for providing telephone services over broadband networks. She has designed and delivered various academic courses in telecom and Internet protocols and architectures, both as an independent consultant for service providers and vendors and as part of her current work as professor of ITM. At IIT's Center for Professional Development (CPD) she designed and built the VoIP lab, and as Director of VoIP Activities there she is now developing programs and projects linking that lab and its students with industry partners.**

Richard "Zippy" Grigonis recently spoke with Prof. Davids about SIP testing, one of the themes of this issue.

*RG: For a small lab you've dealt with some impressive companies.*

CD: I have ongoing relationships with Reef Point, Lucent, Motorola, and a small company called NESS [Network Expert Software Systems, Inc.], which does surveillance software and makes the monitoring software for AT&T for their 911 surveillance group. They monitor the network, so their concern is less with the 911 messages than it is with the health of the network. Their software is in my lab because one of my student projects this semester is to expand the capabilities of that software so that it can be applied to an IP network which is doing SIP phone calls.

We have different relationships with different companies depending on their needs and resources. For example, Lucent (quote - news - alert) has a sort of 'helpmate' relationship with us. We have inherited from them a lot of infrastructure equipment, much older computing gear and even some of their older VoIP gear. They'll give us a suggestion, like, 'Why don't you develop some SIP application that will be of use to people who were disabled'. So the students built a SIP phone for the hearing impaired. That was a project that took about four semesters. By the time they finished it they had developed and demonstrated code that a hearing-impaired person could use to make a phone call to a pizza delivery service and the pizza delivery guy didn't have to do anything special to get the call. They ordered the pizza via speech-to-text and text-to-speech translation, which is all moderated by SIP signaling. There was no need for the old TTY [TeleTypewriter] technology.

A company can simply suggest something to us and we'll go off and develop it. What's great about that kind of work is that a company sometimes really can't make the case and come up with a good ROI for dumping a lot of money into certain research, and so they can come to us and the students are delighted to do it. I don't need to demonstrate ROI, I just need to have a nice piece of code at the end of a semester or two, and a demonstration of it.

*RG: In this issue we mention Reef Point and Spirent. You've dealt with both companies.*

CD: Reef Point came to us and they were interested in third-party independent testing. They brought a device into our lab and their trained our first semester students on how to operate it. The students work in a tight loop with the Reef Point engineers, so that when they find that we've crossed some threshold in performance, they'll be shooting emails immediately to Reef Point's Scott Poretsky, who in turn finds the developer to fix the code. So, the students really have an experience here of being a test group associated with a product. The kids love it when I give them a real focus and help them innovate.

We also have a Spirent Abacus on permanent loan. It enables us to create fairly large and intense assaults of SIP telephone calls across an application layer gateway.

*RG: What's happening now?*

Recently we started working with Motorola. (quote - news - alert) They have little MultiMode, MultiMedia SIP/IMS endpoints. We're going to do performance testing on them, and we're setting up our infrastructure in the lab now. In our first semester we'll build a simple WiFi network over which we can then observe the performance of these endpoints and then subsequently we'll probably do some work in Motorola's labs. First of all, we'll benchmark just how these things operate, and then subject them to different stress patterns.

Keep in mind that we don't charge people to do this, because we see ourselves as an education process. We ask for the equipment, frequently, and we're delighted to get donations to support the lab and its valuable work. Companies always figure out one way or another to reimburse us.