SIP Development and Testing Tools

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TMC

SIP MAGAZINE™

The Authority On Session Initiation Protocol

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Q & A with NeuStar CTO Mark Foster

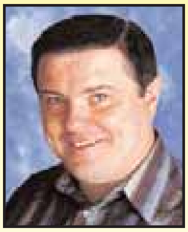
SIP & IMS:

Made for Each Other


He Said, She Said?

EDITOR'S NOTE

Winners & Losers



by
Richard "Zippy"
Grigonis

Having just got back from TMC's stupendous ITEXPO East show in Fort Lauderdale, Florida, I must admit I harbor a bit a jealousy for Ari Zoldan from Launch 3 (http://www.launch3.net) a company that invests in communications companies worldwide. In a drawing at the very end of the show on a Friday afternoon, Zoldan, an unprepossessing fellow denizen of New Jersey (Garfield, to be precise) won a Toyota FJ Cruiser SUV. (Yours Truly is precluded from winning anything at an expo, of course, owing to my esteemed position as Executive Editor and all of the "sacred trust of journalism" trappings that go with it.)

They say that "winners never quit and quitters never win," but, aside from everyone fervently pitching in and "winning one for the Gipper", it helps when the luck of the draw is on your side, as in the case of Mr. Zoldan. I can recall one day when I was back in college, listening to a classical radio station in Philadelphia. The great stage and movie director Joshua Logan was being interviewed and was talking about his favorite classical pieces of music (which happened to include one of my favorite selections, the second movement in A minor from Beethoven's Seventh). Logan suddenly stopped and remarked that "ambitious people say that they'll make their luck, but don't kid yourself, luck is real and you need it." I suppose one could point to VHS tape "winning" over Sony's Beta (alas, manufacture of VHS tape recently ended on the 30th anniversary of its first appearance) and other close calls in history. (Surely a case of "you win some, lose some".)

But then, there's SIP and its win over H.323. H.323 had a head start, appearing around 1996 as an International Telecommunication Union (ITU) standard and designed with a more traditional telecom mindset. SIP didn't appear until 1999 but it was introduced by the Internet Engineering Task Force (IETF) and so it immediately had more gravitas in the new-fangled packet-switched communications world. Both protocols allowed for voice and multimedia to travel via IP, but even though H.323 had quite a bit going for it, SIP (define - news -alert) turned out to be more scalable, less bloated and above all, was easier to work with than H.323 — of course, its detractors would say that's because the original SIP specification couldn't do much! However, over the years the SIP feature set has been extended, has gained tremendously in popularity, and so it has definitely "won its spurs" so to speak.

A major theme of this issue is SIP and IMS (IP Multimedia Subsystem). SIP is a key component of IMS, which takes call control and interoperability to unprecedented levels and serves as a common service architecture to both wireless and wireline communications. IMS involves a major overhaul of much of the world's telecom infrastructure, the cost of which worries many network operators. (I guess sometimes "you can't win for the sake of losing".)

This issue also contains articles on SIP development and testing tools. Interoperability is a major issue in the IMS world, and while every device gets along with every other device swimmingly in a perfect IMS network, the reality is that many devices and applications will have to go through several iterations before they can play together in the great IMS sandbox.

Dare I say, you can't win 'em all?

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

SIP MAGAZINE™

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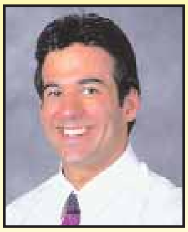
SIP: Ready for Prime Time



by
Rich Tehrani

The SIP market is maturing nicely as evidenced by the massive scale of today's SIP-based applications. After all if SIP applications are not able to scale and carriers and large enterprises aren't able to purchase off-the-shelf software to enable such communications, what is the point?

Sure you can write your own SIP-based applications but it is better to have choices in any market and not everyone wants to start from scratch.

We need SIP systems to scale to the largest implementations as inexpensively as possible so session initiation protocol-based solutions can be deployed all over the world and interact with one another.

I believe one day SIP (or a SIP successor) will be a ubiquitous protocol and when this happens the world will be better off as there will be widespread interoperability. Now, I am not an expecting instant communications utopia to magically appear, mind you - we all know full well that one implementation of SIP may vary from another making them incompatible.

Recently I was pleasantly surprised to see the results of a test by CT Labs of CommuniGate System's (news - alert) SIP-based CommuniGate Pro. If you aren't aware of CommuniGate, their technology allows the delivery of rich media-enabled Internet communications in a flexible manner that carriers require.

CT labs put the software through its paces and the results were very positive.

The test emulated an active subscriber base of unique SIP user agents engaging in real-world call activities. This was accomplished by generating realistic levels of SIP endpoint registrations, peer-to-peer VoIP calls, and application server traffic for various sizes of an active subscriber base.

We need SIP systems to scale to the largest implementations as inexpensively as possible so session initiation protocol-based solutions can be deployed all over the world and interact with one another.

For each tested CommuniGate Pro configuration, CT Labs discovered the maximum number of subscribers that could be supported without failed calls, excessive call answer or application navigation latencies, detectable voice or application prompt quality issues, or other types of service degradation that would be deemed unacceptable by a typical user.

The duration of each test varied depending on the goal and the traffic conditions created. Longer tests were executed to assess the long-term reliability of the CommuniGate Pro architecture.

CommuniGate Pro was found to provide excellent overall performance when subjected to the real-world residential traffic model. This two-server CommuniGate Pro All Active Dynamic Cluster configuration, utilizing an external NAS device for shared storage, supported over 220,000 active users without a single call or registration failure during the test runs.

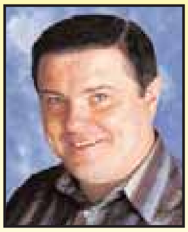
It would certainly seem the above test provides validation for service providers that CommuniGate Pro is a platform that can be trusted to perform under demanding conditions.

In a second phase of the benchmark with a single server, a test verified that the CommuniGate Pro (CGP) product supports a large subscriber base on a single low-cost server. This was proven through an examination of the performance results for call connectivity, registrations, and voice prompt quality. With a resulting call throughput of 7,200 calls per hour, the CGP call processing architecture was found to perform reliably during peak traffic conditions.

So there is great progress being made in the SIP server space and it looks like CommuniGate is helping corporations and service providers alike in furthering open, Internet communications regardless of media type.

SPECIAL FOCUS

SIP, Seamless Handover, and Multi-Mode Phones



by
Richard "Zippy" Grigonis

SIP is the key component to IMS (IP-based Multimedia Subsystem), the common service architecture that shall serve as the foundation to pretty much all of the world's future wireless and wireline networks. IMS in turn, leads us to the previously mythic world of Fixed-Mobile Convergence, where users can roam with any kind of service and any kind of communications device. The basis for such niceties as WiFi-to-cellular roaming, for example, are the 3GPP Voice Call Continuity (VCC) specifications.

One way to keep you from wearing a veritable tool belt of devices is to combine access technologies. Instead of carrying a separate WiFi phone, cell phone (GSM or CDMA) and WiMAX phone, why not wear a just a tri-mode (or "tri-band") phone?

Well, for one thing, a vast WiMAX (define - news - alert)network hasn't yet been deployed (but it soon will). As for WiFi and conventional cellular networks, dual-mode phones have received a lot of attention recently. One proverbial fly in the ointment - at least until recently - has been developing a dual-mode phone that will support "seamless handover" (or "handoff") so that one can simply walk from a corporate (or perhaps residential) WiFi network and continue your phone call on a cellular network, without losing the signal, dropping the call, thus forcing the user to place a call to the same party. With such a capability, great service provider empires could be built. . .

Now, however, there has actually been a veritable explosion of new product offerings in this area:

- (news - alert)AePona Ltd. (http://www.aepona.com), a service delivery platform vendor, demonstrated a standards-based WiFi-to-mobile call handover and continuity system at the recent 3GSM show in Barcelona.
- Alcatel-Lucent (quote - news - alert) has been developing carrier equipment that enables such WiFi/cellular handoffs.
- Avaya (quote - news - alert) acquired Traverse Networks for $15 million, an FMC software-for-cellphones vendor. Avaya, working with Motorola, now sells an add-on WiFi/cellular handoff package for an Avaya PBX.
- Cisco (quote - news - alert) and Nokia quote - news - alert) announced that the dual-mode Nokia E61i and E65 smart phones can either work as a corporate phone extension off of a Cisco Call Manager IP PBX and messaging infrastructure, or else utilize 802.11g and GSM radios (the software enables the phones to sense what kind of network has the strongest signal and to automatically switch to that network).
- (news - alert) Divitas Networks (http://www.divitas.com) calls their system a Mobile-to-Mobile Convergence (MMC) Solution. It consists of two components: 1) The DiVitas Mobile Convergence Appliance (MCA 1000) is installed on the customer premises that helps control and manage mobile devices as they roam among various wireless networks, such as WiFi, cellular and even public WiFi hotspots. 2) The DiVitas MCC software client that sits on the mobile devices and talks to the MCA, providing an interface for applications such as voice, video, IM, voicemail, and presence. The platform-agnostic MCC client can be downloaded "Over The Air" (OTA). The Divitas system works with existing PBX-based voice systems or in a standalone configuration (in which case the "IP PBX" is an internal open source form of Asterix).
- Siemens has recently released a system based on dual-mode mobile handsets, special client software and its "special sauce", the HiPath Mobile Connect Appliance, which sits behind the firewall and communicates with a corporate IP PBX via - you guessed it - SIP. (You were expecting E&M signaling, maybe?) The Siemens appliance houses a SIP registry of all local client handsets, both dual mode and more limited WiFi VoIP devices, and keeps running tabs on their status and presence information. The Mobile Connect Appliance has thus far been tested with the HiPath 8000 Communications Server, the HiPath WLAN product line, and Nokia E60 and Fujuitsu-Siemens Pocket LOOX dual-mode handsets. The appliance comes in three models, for 10, 20, or 150 users.
- (news - alert) SpectraLink (http://www.spectralink.com) has been doing work in this area and, lo and behold, (news - alert) Polycom (http://www.polycom.com), one of the great names in business speakerphones, paid a tidy sum to acquire them.

When WiMAX is finally added to the mix, phones could in theory revert back to the 1980s thick-as-a-brick design for a while. Realistically, however, the public has become too style conscious. We may have to wait quite a while before a tri-mode phone becomes available that's thin enough and otherwise sufficiently good-looking to satisfy the public's fickle sense of design.

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

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Stanley Puts Interactive Intelligence's VoIP Solution to Work

By Greg Galitzine

New Britain, Connecticut-based Stanley Works has announced it has deployed Interactive Intelligence's (news - alert) Customer Interaction Center (CIC) to replace the company's numerous legacy PBX systems from Avaya, Siemens, and others. Stanley employs nearly 20,000 associates across nearly 40 countries worldwide, and in its early stages of deploying the Interactive Intelligence solution, Stanley is looking to roll out CIC across multiple sites across North America.

According to Scott Rudden, director of worldwide telecommunications and network engineering for Stanley Works, "We plan on expanding the deployment of CIC to additional sites across North America as part of our migration strategy to convert traditional legacy environments to VoIP and SIP-based architectures."

Customer Interaction Center gives contact centers a pre-integrated application suite and multimedia ACD capability to manage phone calls, faxes, e-mail and Web interactions from a single platform. Among the solution's key benefits, Interactive Intelligence delivers a lower total cost of ownership by replacing multi-box proprietary hardware with CIC's pre-integrated application suite and single set of business routing rules, anchored by a single administrative interface for PBX, ACD, outbound campaigns, recording strategies and other CIC functions.

In addition to CIC, Stanley Works has deployed Interactive's Interaction SIP Proxy, a server-based appliance used to route calls over IP networks, including the ability to dynamically redirect calls in the event of an emergency, and to adjust call loads across sites during spikes in volume.

http://www.inin.com

Siemens HiPath Xpressions Delivers Open, Standards-Based Unified Messaging

Siemens Communications (news - alert) announced the latest release of its award-winning HiPath Xpressions unified messaging solution. As part of Siemens' range of unified communications solutions, the new release includes native SIP integration with the high-performance Siemens HiPath 8000 IP softswitch, supporting up to 100,000 users. This demonstrates Siemens' commitment to open communication standards while offering new open capabilities such as the ability to emulate virtually any voicemail Telephony User Interface (TUI), and access to voice messages on-the-fly from any Web browser with no requirement for email client integration. These enhancements introduce more flexibility and choices, meeting the needs of diverse clients while improving business process optimization and worker productivity.

A key component of Siemens' Open Communications strategy, HiPath Xpressions V5.0 is a proven, enterprise-class unified messaging solution built on a standards-based and flexible architecture. The solution supports multiple Siemens and third-party communications platforms, using SIP, IP and ISDN, providing superior investment protection. In addition, it integrates with all industry leading groupware including Microsoft Exchange/Outlook, IBM Lotus Domino/Notes, Novell GroupWise, and Qualcomm Eudora, and supports simultaneous email systems on a single platform.

http://www.siemens.com

snom Unveils New IP Business Phone

snom technology (news - alert) announced the latest addition to its line of IP business phones, the snom 370. The snom 370 will provide users with the flexibility and features needed to maximize productivity and efficiency in a business environment. Larger display, expanded memory and VPN access offer business users a simple and secure way to manage calls.

The snom 370 features a larger display area with built-in grayscales to provide users with a clearer view and allow for more information on the screen. The expanded memory capacity enables the phone to support more applications, such as presence indication and presence displays of other users. The phone will also provide users with the ability to customize the display by depicting graphics and high resolution pictures to show the status of contacts. The snom 370 can support several audio devices simultaneously, such as the handset, headset, and loudspeaker. The multicast paging mode makes it possible to use the phone for large-scale public announcements.

With the addition of a new provisioning mechanism, the 370 allows for the dropping of configuration files and software loads into a file system, simplifying configuration and management for administrators. Updated security features include direct access to a VPN in order to secure communication to an insecure PBX system.

http://www.snom.com

Azaire Networks Introduces New IP-CNP with eXtended Mobility

Azaire Networks, (news - alert) provider of multi-access network solutions, announced the availability of its new IP Converged Network Platform with eXtended Mobility (IP-CNP with XM), enabling delivery of enhanced IMS/SIP voice services over both WiFi and cellular access networks. The new solution is fully compliant with 3GPP's TS 23.206 voice call continuity (VCC) standard. Azaire Networks customers now have the ability to offer a complete range of rich voice, data and IMS services across existing cellular networks and new broadband IP wireless access networks.

Azaire Networks new IP-CNP with XM fulfills the mobile users need to access subscribed services in a seamless manner whether at home, at the office or on the road. Additionally, operators can cost effectively deliver a combination of existing voice and data services in conjunction with new, rich VoIP and multi-media services. The new IP-CNP with XM leverages the SIP-based IMS architecture to extend operators existing voice services over WiFi access. It also delivers new SIP-based applications, such as IM, presence, push-to-talk, and video sharing over the same access network. Delivery of rich interactive applications over WiFi and cellular networks in combination with data services like push e-mail and broadband streaming video downloads/uploads, mobile TV and music downloads, ensures that users can experience All Services Everywhere.

The IP-CNP with XM solution is a complete, standards-compliant solution based on 3GPP IWLAN and VCC standards. 3GPP IWLAN functionality implements the highest level of security based on EAP-SIM/EAP-AKA and IPsec/IKEv2 and provides access to existing and new mobile packet mode services. VCC enables handover between WLAN packet access and macro circuit-switched cellular networks.

http://www.azairenet.com

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SIP-based VoIP Platform Enables Online Whiteboarding

By Erik Linask

damaka (news - alert) is making the communications process more efficient for users by way of its patent pending SIP-based Personal Softswitch, which is revolutionary in that it brings the VoIP platform to the end user, meaning that anyone, whether in the office, at home, or in a hotel, can take advantage of the benefits of SIP-based communications, regardless of what other equipment is available to them.

Now, in addition to phone and video calling, including presence features and IM capabilities, and desktop sharing and file transfer, damaka is introducing a secure whiteboard feature. With it, users can draw freehand with online contact globally, just as they can speak, share files, and instant message, enabling real-time collaboration on technical projects, document editing, or anything else that can benefit from live mark-up capabilities, like distance learning facilities and designers.

In addition to offering its services to the consumer market, damaka offers its platform to enterprise customers and service providers on a white label basis, allowing business users to also benefit from a secure, functional, and inexpensive set of VoIP and collaboration tools. Its IM tool enables connectivity with Yahoo!, AOL, Google, and MSN messaging tools.

http://www.damaka.com

Software IP PBX Offers Value and Flexibility

By Erik Linask

Massachusetts-based pbxnsip (news - alert) released version 2.0 of its Windows- and Linux-compatible software, which adds features users are commonly requesting and that make their communications simpler, unified, and generally more effective.

New features include: a user portal that houses call lists, click to play voice mail, personal address book; call recording; cell phone support; IM and presence capabilities; improved plug and play endpoint support; music on hold; new auto attendant functionality; integration with Microsoft Exchange 2007; ENUM support; multicast paging and intercom functions; and more.

"The new version contains some not so obvious features for management of the system," said Kevin Moroz, VP of sales at pbxnsip. "For example, it keeps track of the resource usage and reacts in real time to adjust performance limits and makes backup easy by locating all data in the file system relative to one directory."

In addition to the new features introduced with version 2.0, the pbxnsip product offers all the traditional IP PBX functionality, from end to end security to remote management, from trunking support to PSTN interconnect, from support for remote users via a built-in session border controller (SBC) to P2P communication. It also enables a wide range of user functions, like call control features, multiple appearance, hunt groups, conferencing, push2talk, IVR trees, and more, all of which help make both inbound and outbound communications from efficient.

http://www.pbxnsip.com

Avaya Broadens Commitment to SMBs

Avaya (quote - news - alert) has announced significant enhancements to its Intelligent Communications suite of converged IP telephony solutions specially designed for small and mid-size businesses (SMBs). The enhancements give even the smallest firm an easy, affordable way to take advantage of advanced, IP-based communications to improve productivity, customer service, and virtually any business process. Avaya' s new developments also enable companies to access less expensive call services from service providers offering SIP trunking, which can cut calling costs in half.

Based on Avaya developments in converged communications technology, the company is introducing two versions of enhanced software for Avaya IP Office, the company's flagship solution for SMBs. The Standard edition is designed for very small businesses and serves up to 32 users. The Professional edition enables firms to take advantage of enhanced mobility, multi-site networking or customer service intelligence and scales to support up to 270 users.

In addition to new IP Office software, Avaya is expanding its current portfolio of servers by introducing a new IP Office 500 communications server. Its compact, modular, and flexible design supports telephony, voice messaging, and a complete customer service suite -- all at an attractive price for small and mid-size businesses.

Together, the new IP Office software and communications server deliver a solution that enables SMBs to buy only what they need today, while easily adding users or functionality via software licenses that are also compatible with existing IP Office solutions.

http://www.avaya.com

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CASE STUDY

Leading Edge Provider Seeks Leading Edge Solution

by
Erik K Linask

Quilogy is a professional services company based in St. Charles, Missouri, and a Microsoft Gold partner. In addition to five buildings in St. Charles, its some 250 employees are scattered among 13 other cities across the Western half of the United States.

Growing Pains

(news - alert) As the company grew - it initially had just two buildings in St. Charles - it realized that its ancient traditional PBX system no longer was capable of the functionality or capacity that was required.

"I would say the system we were using was bought used before I got here, and it had probably been heavily used before we started using it," said systems technologist Adam Swank. "I would say it was probably 20 years old."

There were also technical and financial limitations. When the company acquired three additional buildings in a historical district of St. Charles, the traditional PBX was extended through just the two original buildings. Quilogy was then faced with a choice: purchase three new systems for the new buildings or find a way to extend the existing network to the new facilities.

According the Swank, the company felt there must be a better alternative, and when it fitted the new buildings for connectivity, it did so with the intent of installing some variety of network-based telephony system.

Quilogy finds itself on the leading edge of technology and, as such, is keenly aware of the benefits of SIP and was looking for a SIP-based platform that could communicate with its Windows environment and could also integrate into Microsoft's IM platform. The company looked at several companies, some large, some small, some hosted, some premises-based, but found nothing that met its requirements.

Hosted solutions, while popular, faced regulatory challenges in that, to have a phone system in each of its headquarters buildings, FCC regulations would have required running a T1 to each site. Hosted providers must assign each T1 identifier to an address, and they are not able to handle multiple addresses over one Internet connection. In order to comply with these 911 processing regulations, Quilogy would have been forced to radically alter its connectivity.

The Solution

So, after investigating several options, Quilogy finally found a premises-based, Windows-compliant system with the required features that also offered a trial period - and so the relationship with pbxnsip was born.

"There had been rumors about a Microsoft-based phone system that never really came to fruition," Swank explained. "We wanted to use something that would be Microsoft-compliant in the future, but also something that we could use now. Pbxnsip definitely fit that bill."

Because Quilogy had recently rehabbed several buildings and already made a number of network improvements - it already had the service capacity, the bandwidth, and the network - when it came down to pbxnsip, it was just a matter of the software and the phones. In fact, according to Swank, the deployment was so easy that they were able to place calls on the network the same day the server and software were installed.

Since the pbxnsip IP PBX is based on SIP, it easily integrates with a wide range of vendor hardware, which meant Quilogy was free to use additional hardware of its choice, like IP phones from Polycom and an IP gateway from AudioCodes. With pbxnsip, Quilogy has access to all the features of traditional IP PBXs, including end-to-end security, remote management, trunking support, call controls, conferencing, and much more. The latest release of pbxnsip, version 2.0 has added features like cell phone support, call recording, IM and presence capabilities, ENUM support, and more.

The Result

Since deploying pbxnsip in St. Charles more than a year ago, Swank says they are patently pleased with both the platform and the service from pbxnsip. There has been little need for support, but even when concerns arose around the phones or the gateway, pbxnsip provided internal contacts for Quilogy at the appropriate vendors, hastening the resolution process so that Eulogy's business would not be negatively impacted.

Quilogy has also deployed pbxnsip in its Seattle, Chicago, and Des Moines locations, and will add Nashville this spring. In Des Moines, in fact, Quilogy was able to save more than $1,000 per month on communications costs, but in addition to the communications capabilities cost savings, Swank notes that the relationship works because, "They are in line with what we are thinking as far as the future of SIP and Microsoft integration. They think the way we do and they're a very agile company like we are, so it was a good fit from that perspective."

He added that, "In addition, it's an enterprise-level phone system at a non-enterprise price. They are definitely out there ahead of the curve."

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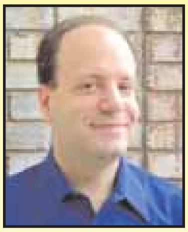
SPEAKING SIP

Peering Outside: Federating Presence



by

JD Rosenberg

***Presence.* When I say that word to folks outside our industry, they think I'm talking about the stuff you get during the holiday season that's wrapped in shiny paper and bows. Just a few years ago, even folks within the VoIP or telecom industries hadn't heard of presence. Now, it's a common part of the lingo of modern telecommunications. In case you haven't heard of it, *presence* refers to someone's ability, willingness and desire to communicate across different devices, modalities and services. The simplest manifestation of presence is the traditional "buddy list," such as the ones provided by Yahoo, AOL and MSN. In those applications, presence shows up as those icons and status messages that say something about your ability and desire to receive instant messages.**

Instant messaging is just the beginning, though. I can also be reached on my desk phone, my home phone, my cell phone, through email (at multiple addresses), through push-to-talk services, through the postal service and, of course, face to face. With so many modalities, it becomes hard for someone to figure out when I should be contacted, and how. That's where presence comes in. Presence tells other people about all of the ways I can be reached and gives them clues about how and when I can best be reached. Presence helps bring order to the mess of modern communications. It makes the experience richer and facilitates more frequent and more effective communications. That has value to end users and the service providers that offer it.

Unfortunately, presence exists in islands. Subscribers of one provider cannot see the presence of subscribers of other providers, unless they obtain identities and clients for each of those providers. Worse still, enterprises that deploy presence systems see only the presence of other users within the same enterprise. Presence is particularly valuable in the enterprise. At work, communicating effectively is important. Everyone is busy, and getting the most out of your day means quickly connecting with the people you need to connect with and moving on. For example, when I have a question that can be quickly answered by any one of a number of people, I quickly scan my buddy list, see who is online, and shoot that person an instant message. I get a response, and I'm done. Presence helped me choose whom to contact. Without it, I would have had to call people randomly or email them randomly, and most of those attempts would have been wasted.

My business contacts don't end at the front door of my office building. I, like many others, have colleagues and customers at other enterprises with whom I interact. Communications with those contacts is even more valuable. They have less time for me than for their own internal colleagues, and the information they provide is also usually more valuable. For this reason, being able to see their presence would be even more useful than seeing the presence of people within my own enterprise.

So why doesn't it work? What makes it difficult to extend presence outside the boundaries of an enterprise? Why can't we federate?

There are many challenges. One of the biggest ones is scale. Presence generates a LOT of traffic. Consider two large enterprises, each with 100,000 employees. Each of those employees has a buddy list of 50 people, 10 of whom are in the other enterprise. Each user changes his or her presence state 10 times a day. This results in 1 million subscriptions reaching between the two enterprises, generating 10 million notifications a day, or 347 notifications per second. These numbers are one to two orders of magnitude larger than the typical call volume between enterprises of this size. The problem gets even worse for extremely large service providers whose subscribers number in the millions; inter-network notification rates there can be measured in the tens of thousands per second.

Security is another challenge. How do enterprises make sure that only authorized recipients can get and see the presence data? How do they make sure that their presence systems don't get used for launching denial-of-service attacks? Those kinds of problems are not specific to presence systems, but the scale of these systems makes them more attractive targets.

The final challenge is making the value proposition clear. Many enterprises still don't utilize presence and IM systems at all, many actively block it, and even more have users who make use of consumer systems without the approval or support of IT departments. Presence and IM have yet to catch up to voice and email as recognized essential tools for today's information worker. However, it's just a matter of time. Once they have caught up, unfederated presence will look as antiquated as email or voice systems that don't work outside the enterprise. I look forward to that day.

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

PRESENCE ENABLED

Jump In, the Water's Fine

by Dave Uhlir

Instant messaging is at once the Trojan Horse and Achilles Heel of *presence.* IM has introduced people to the power of presence in delivering more synchronous interpersonal communications. At the same time, IM has helped balkanize the technology as vendors and customers came to understand presence as something living exclusively within applications (like chat) that they can wholly own. Fragmenting the IM market has slowed and complicated efforts to integrate presence within mainstream network communications technologies, while scaring away some customers with a false choice between VHS or Beta. In the early days of networking, it took HTTP, HTML and Web 1.0 visionaries to convince mainstream IT that the water's fine and its time jump in because there is more to networked computers than sharing some files and a printer. A similar challenge confronts the Web 2.0 visionary as it is a common sentiment in IT that *if I don't care about IM or collaboration software, I don't have to care about presence enabling my network.*

Lying at the heart of Web 2.0, live services, mashups, *et al.,* is a comprehension of presence as living everywhere, but never wholly somewhere. That presence is collectively owned by a dynamically shifting hierarchy of people and systems exchanging variable information about their status that affects the flow of data and media. Real-time information on the status of people, information, and things has great value to a wide range of specific applications — but it is not, should not, and I submit technically cannot be owned by a specific application.

This is an important architectural distinction. When thinking about presence solely in terms of your application, the tendency is towards bolting a presence engine onto existing application servers. This passageway is full of traps and dead ends. One is in trying to leverage an existing application server's online transaction processing techniques. . . Really? Are you suggesting that the fact that my presence changed from available to on the phone requires transaction processing levels of verification? Okay, what happens when we add these layers of verification, will running this architecture push presence updates fast enough to be relevant? Will it scale to handle all the nodes of a unified communications system? Ultimately you have to ask yourself if you want to know what my presence *was*, or what it *is!*

As it turns out, building your own presence server is really, really hard. You can do it, but developing the know-how to make it scale is costly, and don't forget to ensure your application can exchange presence with other networked entities lest you create yet another messaging island, and what's the point in that.

While scalability and latency shouldn't be much of a concern for things like chat, even when adding the simple automated presence changes that can be pulled from a PBX or Outlook calendar, they matter a great deal once you start considering a world in which anything that is IP-addressable can be given the ability to publish its current status and subscribe to events relevant to its process(es). Moreover, people will interact with all of these systems in a lot of different ways and from a lot of different places for all types of reasons. At that point, your application could be spending all of its cycles just trying to keep up with the presence changes published to the network.

Walking towards the light, we start to see applications outfitted with simple XML transceivers while the heavy lifting is done by dedicated presence servers that don't require the overhead of a runtime environment. Applications are freed to communicate only what has changed, leaving it to the server to determine who cares and why. As better and more efficient consumers and publishers of presence information, applications can exchange more information with more points of presence in real-time, while driving more value into the aggregate of information exchanged.

Today, as it was back when IT came to understand the value of networking, we have standard protocols such as SIP, RTP, and XMPP that have warmed the water. The visionaries are experimenting and already tapping into this new pool of information drawn from existing network services to create even more powerful applications and add-ons. Now it's time for the mainstream to jump in and discover that there's more to presence and real-time collaboration than IM, a PowerPoint presentation, and a conference call.

Dave Uhlir is vice president of marketing and product management for Jabber. (news - alert) *For more information, visit the company online at http://www.jabber.com.*

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FEATURE ARTICLES

SIP and IMS

By Richard "Zippy" Grigonis

SIP (Session Initiation Protocol) and IMS (IP Multimedia Subsystem) are huge topics these days, the subjects of endless discussion. SIP defeated its rival, H.323, in the call control signaling area for IP Communications, but it was the harbinger of more grandiose things to come — namely, the grandiose IMS architecture, which is based on SIP and started out as a small part of the 3G wireless world. Both wireless and wireline operators soon realized that IMS could become a universal service architecture for the world's communications services, and that IMS enabled operators to hatch new services by the dozen in a multi-vendor friendly network. In an IMS universe, such services can roam with users, which leads us to the Shangri-La of fixed-mobile communications (FMC).

Over at the legendary telephony hardware/software vendor (news - alert) Dialogic (http://www.dialogic.com), the Director of Product Management for Media Products for the Service Provider Markets, Chris Fullam, says, "We've bought into both SIP and IMS, no question about it. We've pretty much built our product strategy around allowing people to migrate from the current networks to IMS. We don't think it will happen overnight. Indeed, things seem to be moving slower than was originally forecast. Still, our strategy that we formulated a few years ago about using host media processing [HMP] as our basic media engine and SIP for call control, has worked out very well. Both HMP and SIP really provide a good foundation so that we can evolve with the market. We've been tuning that architecture so that by adding DSP [Digital Signal Processor] offload to HMP, we think that we've got a really good combination of factors: First we have the best performance. Second, the price is good because we can really take advantage of off-the-shelf servers."

"So, we're buying into SIP, IMS and HMP," says Fullam, "and that's the basis for our product strategy. It seems to be playing out really well too, though as I said things are moving a bit slower than we expected. The whole rollout on IMS and 3G networks takes time. A year ago we thought that we'd see a lot of deployments in 2007. As it happens, we're seeing pilot projects, but it now looks like the big deployments will start occurring in 2008 or 2009. In China, for example, there are some pilots, and the 2008 Olympics is acting as a sort of driver to get things rolled out."

"Some of the key benefits that IMS provides makes sense to everybody," says Fullam. "Lowering your costs, being able to rollout new applications more quickly, having a distributed environment and modularizing components — those are all really great benefits that no one can argue with. Everybody plans to move toward IMS but the realities of the situation are that operators have to keep their revenues flowing and they need to leverage existing investments. Still, our approach seems to be working in this environment. We have network operator customers that have deployed things such as color ring back, the Short Messaging Service, or just a media solution in today's network. They can move to our newer platforms, such as our Multi Media Platform [MMP] that uses HMP with the DSP offload, and they can keep using the same APIs. Essentially, they can leverage the vast majority of their existing application development work. Over time, they can move to some of the other standard protocols such the Media Sessions Markup Language [MSML] or H.248 and use the same servers. That's a key piece of our approach, to provide a migration strategy that allows people to leverage their existing infrastructure."

"The 'M' in IMS is Multimedia, and for years we've felt pretty good about HMP as an excellent approach to handling media processing. I think most of our competitors and many people in the industry have said, 'Yes, it's a great idea, but it doesn't scale, and it will only work for small-scale deployments or where there's extreme cost pressure'. And I think we've now seen people recognize that our combination of HMP and DSP boards is highly scalable. In particular, if you take the most MIPS-intensive functions, such as the transcoding processes for wireless, and you move them to a DSP board, now you've got the best of both worlds. You do the majority of the media processing — playing prompts and call control activity and all that — on an off-the-shelf, high-end server, and you relegate the processing-intensive transcoding processes to the DSPs. That mix gives you the best overall price performance in building equipment that will function in the greater SIP and IMS environment."

The Great IMS Migration

So, companies known for their telephony hardware, such as Dialogic, are gearing up and jockeying for position in preparation for the upcoming Great Migration to IMS.

At (news - alert) AudioCodes (http://www.audiocodes.com) VP of Systems Yehuda Hershkovits says: "IMS uses a common IP core with SIP-controlled endpoints. SIP is used to set up, maintain and terminate multimedia sessions. Key elements in the IMS architecture are media gateways for interconnection with the PSTN, MRF [Media Resource Function] for advanced multimedia services, and SBCs [Session Border Controllers]. AudioCodes provides all key elements."

"AudioCodes' media gateways have been inherently designed as IP-based systems to support wireline [carrier VoIP & enterprise VoIP], wireless [CDMA, UMTS, GSM, WiFi, & WiMAX] and cable access networks, making the transition to a combined, converged IMS-based core network a relatively simple task," says Hershkovits.

Hershkovits elaborates: "AudioCodes' enterprise gateways [Mediant 1000/2000/3000], access gateways [Mediant™ 5000/8000] and media servers [IPmedia™ 2000/3000/5000/8000] fit into any IMS-compliant network on which fixed-mobile-convergence [FMC] is based. Since the inception of VoIP, the company's gateways have operated in distributed networks, recently adopted by IMS. In addition, its field-proven, IMS-compliant H.248 control protocol supports those functions necessary for IMS deployment — IMS uses H.248 as the transport call control. AudioCodes products already have a choice of multiple vocoders — GSM-EFR, AMR, EVRC, iLBC, G.729A, and so forth — needed to build a truly converged core network supporting GSM, UMTS, CDMA, cable and wireline access networks, to enable interoperability with an IMS core network.

In addition to standards — or multi-standards — compliance, AudioCodes teamed up with NEPs [Network Equipment Providers], SIs [System Integrators] and Service Providers, offering VoIP equipment with a full commitment to open interfaces and interoperability with third party network elements, which is at the heart of IMS philosophy. AudioCodes is closely following 3GPP SIP standardization, assuring IMS compliance with no hardware changes, thereby protecting infrastructure investment."

Will SIP Keep Evolving?

One wonders if, after all of this ramping-up to deal with SIP and IMS, whether SIP is in fact suited to working in an IMS environment as it is, or whether there may be any more official, major extensions to the protocol.

AudioCodes' Yossi Kurzman, Director, Product Marketing, Systems, says, "SIP is an evolving standard, making interoperability a challenging task. The IMS is an IP-based core network used with multiple access networks — wireline, wireless and cable— to provide a converged service to wireless, wireline and cable subscribers. Each new access type brings along a whole new set of standard activities that impact the IMS core network. *Access agnostic* is the goal, but not yet the reality. Each new access type presents a different set of requirements and challenges. It will take time to overcome them, but, as with what happened following the early days of VoIP, standards eventually stabilize and reach equilibrium."

"All customers seek unique applications to justify their large investment in IMS deployments," says Kurzman. "Voice is still the leader application, generating a significant portion of revenues. Video could play a definite role in the future."

"MSOs [Multiple System Operators] are now offering subscribers mobility," says Kurzman, "such as quadruple play, in addition to the converged voice, video and data services over IP [triple play] offered by wireline, cellular and VOBB [Voice over Broadband] operators."

"IMS, with its SIP architecture," says Kurzman, "enables video over IP communications, presence services, Instant Messaging, Multimedia services and mobility, such as handoff between VoIPoWiFi to GSM/CDMA. IMS is the architecture that will help MSOs offer quadruple play services, secure their competitive status, expand their subscriber base and increase the ARPU [Average Revenue per User]. IMS' SIP architecture needs to continuously adapt to these market forces and changes."

Where the Legacy PSTN Meets the IMS Future

SIP and IMS will be stuck with legacy PSTN technologies for some time to come. Mindful of Dialogic's HMP/DSP offload combination to handle MIPS-intensive tasks such as transcoding, Yours Truly asked Rich Poole, AudioCodes' Director of Business Development whether he thought extensive interworking and transcoding would be needed for SIP and IMS to interoperate with legacy PSTN technologies.

Says Poole, "A fundamental asset of the IMS framework is its ability to transparently interconnect different types of network access, viz., cellular access [AMR coders, EVRC coders] wireline access [G.711, G.729], PacketCable access, WiFi access and WiMAX access, each of which uses different coders. To perform interworking with the PSTN, intensive, high-quality, high-capacity transcoding capabilities are required, so extensive transcoding is a given."

"In the case of AudioCodes," says Poole, "our gateways are at the heart of media/bearer processing within IMS-based networks, providing the appropriate voice processing required to complete and handover calls between divergent access networks and the existing PSTN, if required."

"These divergent access networks typically require different vocoders, protocol and signaling support," says Poole, "so *maximizing voice quality* will naturally dictate minimizing the number of vocodings required in a given call and naturally increase the number of vocoders required to be supported on a given IMS media gateway."

Poole proudly notes that, "In the 3rd Annual ETSI [European Telecommunications Standards Institute] Speech Quality Test Event, AudioCodes was the only vendor out of 10 who participated which did not score below the standard on a single test and exceeded the standard significantly on a number of tests. AudioCodes submitted a Mediant™ 2000 digital gateway for the testing. The Mediant 2000 shares core enabling technology with a wide range of AudioCodes gateway and media server platforms, testing the core of a large number of AudioCodes products."

FEATURE ARTICLES

Integration of SIP & SS7 for Voice-over-IP

Opportunities & Challenges

By James P. Rafferty

There is strong momentum in the telecom industry as carriers make the shifts to providing voice and other services over an emerging SIP network infrastructure. However, there is a large class of customers whose phones are circuit-based and a tremendous amount of existing circuit-switched infrastructure that will be in the carrier networks for years to come. One of the challenges for carriers is to transition in stages to a SIP-based core network and related applications, while leveraging existing infrastructure and providing a package of services for both traditional and IP phone users. One of the keys to accomplishing this is by providing support for interworking between the traditional circuit-switched signaling methods and SIP. In particular, the SS7 (Signaling System 7) signaling approach is widely deployed on telephone switches that were deployed during the past twenty years.

A variety of standards groups have taken on the challenge of providing interworking between the large SS7 install base and SIP. (define - news - alert) In this article, we will review the various efforts that have been made in standards bodies to address these needs and consider typical use cases on how SIP and SS7 can be used together to provide connectivity and voice services for both IP and circuit switched users.

SS7 and Voice-over-IP

The SS7 network emerged in the 1980's as carriers began to design their networks in a way that would be more friendly toward the development of various enhanced services. SS7 followed earlier technologies like the Integrated Services Digital Network (ISDN), but was different in that SS7 utilized a separate overlay network for signaling, so that it would be possible for signals and media (for example, voice or fax) to follow separate pathways when traversing the public circuit-switched network. By separating the signaling from the media path, it was possible to improve network efficiency, evolve more complex services and offer extra protection for signaling within network equipment.

The greatest success story of SS7 to date has been in providing the signaling infrastructure that is used for virtually all cellular wireless calls worldwide. However, SS7 is also widely used in the landline circuit network, also known as the Public Switched Telephone Network (PSTN). The use of SS7 can vary greatly. For example, the SS7 ISDN User Part (ISUP) is a call control protocol and is primarily used for setting up and tearing down calls, albeit with much more information being contained about the calls than was commonly the case on earlier networks like ISDN. Another common use of SS7 is for database access applications. Here, the Transaction Capabilities Application Part (TCAP) plays a key role by enabling the lookup of database information that is commonly used for applications such as local number portability.

As SIP has become the mainstream protocol for VoIP, the need arose for SIP-based equipment to interwork with virtually all of the existing PSTN protocols. Among these, SS7 is arguably the most important, since it was the last of the major PSTN protocols to emerge and is widely used for all kinds of existing enhanced services in both landline and wireless applications. In line with this, there has been a great deal of standards work done by a variety of different bodies to enable effective interworking between SS7 and SIP (see Table 1). In the sections to follow, we will examine the most important of these standards developments and discuss the current state of deployments.

Table 1

Standards Organization	SS7 — SIP Interworking Standard
IETF	RFC 3398
ITU-T	Q.1912.5
3GPP	TS 29.163

In the Land of RFCs

The Internet Engineering Task Force (IETF) is the place where SIP originated and where most of the related standards development takes place. The standards produced by the IETF are known as standards track Request for Comments (RFC) documents. The primary RFC document on SIP to SS7 interworking is RFC 3398. This standard was one of the early efforts of the SIPPING working group and its focus is on mapping ISUP messages to SIP messages, and ISUP parameters to SIP headers. This kind of mapping is often implemented via a media or signaling gateway for cases where phone calls are being transported between an SS7 network and a SIP network.

One of the challenges in ISUP to SIP mapping is the existence of several SS7 variants. The most common variants are those of the International Telecommunications Union (ITU) and the American National Standards Institute (ANSI). However, other variants are also possible. RFC 3398 makes some simplifying assumptions to deal with this; it provides its mapping based on the ITU version of ISUP and then offers guidelines on some differences which may occur in other versions.

An example of the mapping of a SIP session to an ISUP session using these conventions can be found in Figure 1.

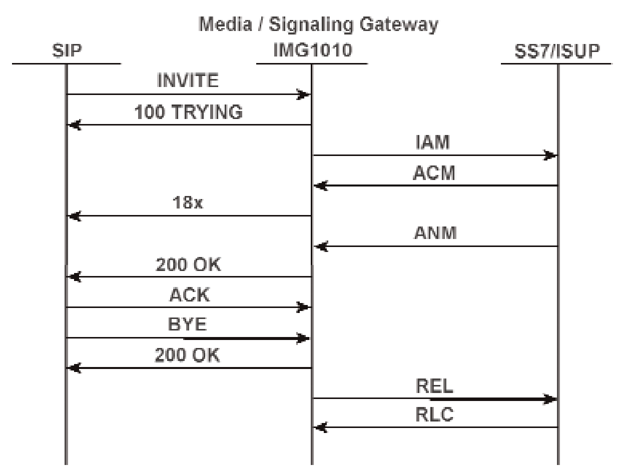

Figure 1. Basic SIP to ISUP Call Setup

A key part of this translation is to take the headers of the SIP Invite and then use mapping conventions in order to populate the parameters of the equivalent ISUP setup message, which is called the IAM, or Initial Address Message.

RFC 3398 provides a basic guide to interworking between SIP and ISUP, but it turns out that there are also other RFCs that can help in providing a more robust implementation. For example, there is an RFC 3326 that provides a SIP header that allows ISUP cause codes to be transmitted via SIP, thus offering some additional information to the gateway about what is actually taking place at different stages of an ISUP call, such as at the time of a call release. Support for these RFCs is provided in some existing gateway products, such as the Cantata IMG1010.

Union of the Parts

The International Telecommunications Union was the key originating standards body for SS7 and has produced its own standards document offering guidance on translation between ISUP and SIP. The document is known as the Q.1912.5 recommendation. It maps the two protocols and addresses additional details that enable additional ISUP functionality to be translated for use in the SIP network. For example, there has been a lot of work done in both SIP and ISUP to support asserted identities and address related privacy issues. If a subscriber has requested a service that calls for suppression of caller identification information, there are ways to reflect this in the ISUP messages. Further, there are equivalent mechanisms in SIP which are described in RFCs 3323 and 3325. The challenge is to be able to effectively translate between these two representations. The Q.1912.5 document provides implementers advice on how to do this, but stopped short of fully supporting the ISUP service known as COLP/COLR which addresses these matters.

There is also a method called overlap signaling that is used in some variants of ISUP. The mapping of this signaling was not addressed at all in RFC 3398, but guidance is provided within Q.1912.5 on how to take this information and take appropriate actions on the SIP side.

The Q.1912.5 document also makes reference to an approach which is known as SIP-T by the IETF and SIP-I by the ITU. In this instance, SIP is used as a tunneling method to transport the full set of ISUP headers from one SIP-T enabled entity to another. SIP-T is a preferred approach for bridging ISUP networks via IP, since none of the ISUP header content is lost in translation to SIP headers. By contrast, the typical mapping of ISUP to SIP causes about 80% of the ISUP parameter information to be left behind.

All in all, this recommendation provides a useful complement to RFC 3398 that will allow a SIP — ISUP gateway to offer a richer, more complete translation between the two protocols. There is deployment of this recommendation, but its use has been less prevalent than that of RFC 3398.

IMS — Convergence of the Species

In recent years, there has been a lot of discussion about a new approach to using the SIP standards that is known as the IP Multimedia Subsystem or IMS. IMS attempts to take the SIP standards and put them in a framework that is suitable for developing a variety of enhanced services, not only for voice but for other types of media. One of the key concepts of IMS is that the services should be provided independent of the type of access that is used. So, for example, one could have a voice mail service in IMS that can be used whether a call originates from a cellular wireless, traditional landline or WiFi hot-spot.

IMS began as an effort to provide services for next generation wireless networks, but there was an understanding from the beginning that many calls would originate on the existing SS7 network. In IMS, there is a Media Gateway Control Function that is used to intermediate between the worlds of ISUP and SIP. A key focus of this function is to perform signaling conversion between ISUP and SIP. Some implementers also choose to combine both this function and the Media Gateway media conversion function within the same system. The basic concept is to take a call as it comes in from the SS7 network and then convert it to SIP on the edge of the IMS network. From this point forward, the signaling is based on SIP and the media has been packetized. This allows the call to interact with various SIP enabled Application Servers and other IMS entities such as the Media Resource Function (MRF) to actually execute a service such as voice mail or video mail (see Figure 2).

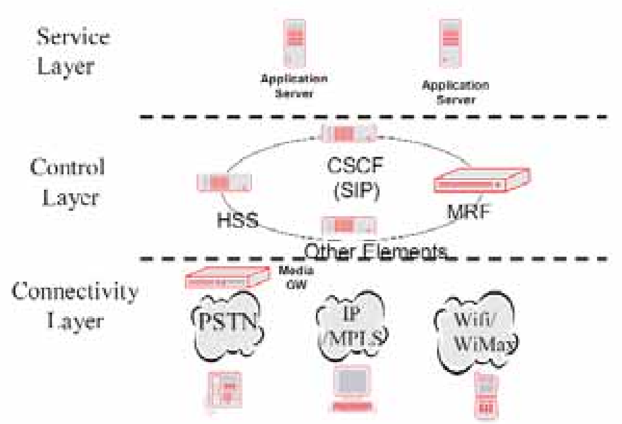

Figure 2. IMS Network Overview

In order to accomplish the interworking between ISUP and SIP, the Third Generation Partnership Project (3GPP) has devised its own set of documents that guide the translation. Some of the key documents in the IMS that are related to this activity are 29.163 on interworking between the IMS and circuit networks, and 24.406 on voice call continuity between cellular and IMS networks. From a technical perspective, the 3GPP work builds on the ITU-T Q.1912.5 recommendation and is closely aligned to it. However, it has gone beyond the earlier document in some respects. For example, it has taken some steps to formalize mapping support for asserted identity and privacy services on the ISUP side that were not fully addressed in the earlier specification.

Since the related 3GPP standards work is still underway, deployment of these specifications is at an earlier stage than the previously discussed documents, but implementers of the SIP to ISUP interworking as defined in Q.1912.5 are already close to compliance with the 3GPP document.

Summary

In the circuit-switched world, SS7 signaling has been used to provide a wide variety of services for both wireless and landline users, including such popular features as *local number portability*. Since there will be a co-existence of the circuit-based and SIP networks for many years to come, it is important that devices such as media and signaling gateways can provide ways to support calls that need to cross the boundaries of these two networks. As reviewed in this article, the IETF, ITU and 3GPP standards organizations have all developed standards to aid implementers in converting SS7 ISUP into SIP and vice versa. As a result, vendors have been able to provide products that provide ISUP to SIP signaling conversion at the boundaries between networks. This allows carriers to leverage their current SS7 infrastructure and execute a measured transition in the deployment of SIP-based network elements and services.

Industry legend James Rafferty is the Senior Product Manager for IP Telephony at Cantata Technology (news - alert) (http://www.cantata.com). He is responsible for the company's IP and carrier voice product lines. James is also active in the IP Telephony standards process through his work with the MEGACO working group of the Internet Engineering Task Force and related contributions to ITU-T Study Group 16.

FEATURE ARTICLES

SIP Development Tools

By Richard "Zippy" Grigonis

SIP has become the premier the call control/signaling protocol for IP Communications, unseating its predecessor, H.323. Developing products that rely on SIP is less difficult than with H.323, and working with SIP code, while not a walk in the park, is nevertheless relatively straightforward compared to other types of protocols.

SIP is the core signaling protocol of the upcoming IMS (IP Multimedia Subsystem) network, which will unite wireless and wireline networks, enable fixed-mobile convergence (FMC) and allow for a multitude of services to be developed almost willy-nilly. The SIP protocol and SIP servers are the enablers for IMS advanced services. However, developers have discovered that there's a big difference between SIP compliance and IMS SIP compliance. Basic SIP with all of its extensions is fairly complex and IMS SIP is an even more complex signaling protocol. Many changes and a large number of complex extensions to the original IETF SIP standard were introduced for IMS, making IMS extended SIP even more complicated than ever before. Moreover, some of the IMS SIP interfaces are still under discussion. It's a major challenge to developing SIP systems containing advanced features for IMS deployment.

The differences between IMS SIP and non IMS SIP are as follows:

First there are the IMS SIP extensions on the access level:

- SigComp (RFC 3320)
- P-headers (RFC 3455/3325) — P-preferred-identity, P-access-network-info, P-asserted-identity, P-called-party-id. When discussion IMS SIP, many vendors think that IMS 3GPP has defined a bunch of P-headers needed for the IMS network to be capable of providing specific services that the IMS network should bring to the world, but the P-headers are a minor issue in being IMS compliant. Many P-headers were defined to pass relevant information in the network, but having these capabilities is not enough to give you the capability to develop IMS compliant elements.
- AKA-MD5 (RFC 3310)
- Security agreements (RFC 3329)
- IPsec
- Media authorization (RFC 3313)
- Service-route mechanism (RFC 3608)
- Reg-event package (RFC 3680)
- IPv6

Then there are security extensions (too numerous to list here) and miscellaneous extensions such as IMS resource reservation (RFC 3312), path headers (RFC 3327), SDP extensions (more attributes, grouping of media lines, more codec support, etc.), XML schemas usage including those in SIP message bodies (registration event package, XCAP, PIDF etc.). Finally, there are extensions by the IETF SIMPLE group with their work on presence and IM optimization (partial notifications/publications, notifications filtering, resource lists (RLS)/SIP exploders, and MSRP)

Fortunately, vendors have evolved their SIP development tools to keep pace with transitions in both basic IETF SIP and IMS SIP.

"Basically, the SIP Developer Toolkit, it really focuses on SIP tools for developing SIP network elements," says Subocki. "We also offer a larger SIP Developer Suite, or which comes with an IMS package [See Figure 1]."

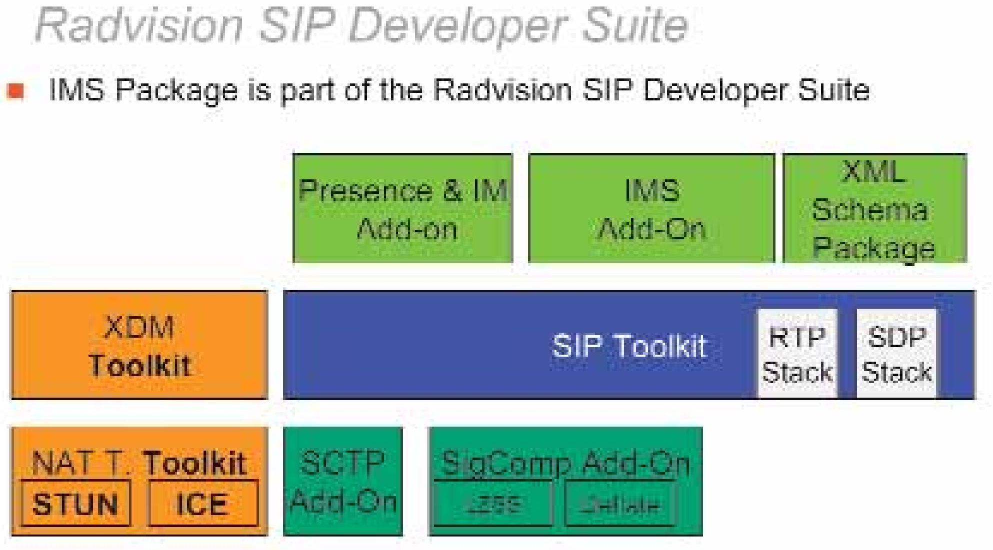

Figure 1.

One of the major vendors of SIP and SIP IMS-based development tools is RADVISION (news - alert) (http://www.radvision.com). Sagi Subocki, head of the company's IMS Segment in the Technology Business Unit and Manager of the SIP, Megaco and DIAMETER toolkits, says: "We are focusing on building developer tools, including SIP Toolkits and Megaco / H.248 and DIAMETER toolkits too. We provide a framework, both to the client side with our multimedia terminal framework and a SIP server development platform. We have a set of testing tools, both for H.324M technology [the mobile version of ITU H.324, defined for visual PSTN phone terminals] and for SIP. IMS [IP Multimedia Subsystem] is a strategic vertical of the business unit on which we have focused for a few years now."

"We also now have a full IMS Developer Suite, developed to focus on the IMS market," says Subocki. "The IMS Developer Suite consists of various components; each of which is dedicated to a different protocol, and has IMS, Megaco / H.248, DIAMETER, and testing tools designed for IMS network elements. As part of this Developer Suite, we have a SIP Developer Toolkit which of course involves the SIP side of the IMS network. It has the basic SIP stack with a complementary RTP and SDP stack and add-ons to the SIP stack which give the specific capabilities that each customer might need. There are also complementary toolkits such as a NAT [Network Address Translation] Traversal Toolkit for developers who need capabilities of dealing with network borders and firewalls."

To be precise, the RADVISION Developer Suite offers the following:

- The Presence and Instant Messaging add-on — RADVISION calls it the SIMPLE add-on.
- The IMS add-on
- An XML schema package that gives support for specific XML schemas needed for more advanced services.
- The NAT Traversal Toolkit, which has both STUN and ICE capabilities.
- The SigComp package, which is needed in IMS. This is mandatory and very important in the IMS network because of the issue of access through wireless and 3G where bandwidth is sometimes limited and is therefore a valuable resource. (This is why compression of the signaling protocol is mandatory.)
- An SCTP add-on
- The XDM toolkit.

"With our SIP Developer Suite, customers can basically mix-and-match the specific add-ons and toolkits they need for the specific network element which is being developed," says Subocki. "You get our product in source code, very extensive documentation and of course sample code and GUI test applications."

SIP Tool Potpourri

There are all kinds of SIP-related tools Yours Truly has run into in recent years, such as the following:

- The SIP-H.323 Converter by Cosystech (news - alert) (now CoSystems, http://www.cosystems.com),
- DC-SIP a complete package of portable source code, sample applications, full reference designs, test applications, and customization tools, by Data Connection (news - alert) (http://www.dataconnection.com).
- Various testing products by GL Communications (news - alert) (http://www.gl.com) that provide extensive SIP-based call emulation, analysis, SIP call trace, and call monitoring.
- The SIP Stack from HCL Technologies (news - alert) (http://voip.hcltech.com) that smoothes the progress of SIP-based VoIP product and solution development. It comprises SIP User Agent and SIP Server reusable components that have been tested and ported on to multiple platforms.
- An API to develop SIP-based applications from Hellosoft (news - alert) (http://www.hellosoft.com).
- Hotsip's (news - alert) (http://www.hotsip.com) Service Creation Environment (SCE) for the their M2CE SIP application server with access to open APIs, making it relatively easy to add and customize new and existing applications.
- A full SIP Stack and SIP Lite from Hughes Software Systems (news - alert) (http://www.hssworld.com).
- A SIP stack from Mediatrix (news - alert) (http://www.mediatrix.com).
- Nine-9s (news - alert) (http://www.nine-9s.com); their TeleSoft product includes their CompactSIP SIP stack, 3GPP / WiMAX / IMS SIP stack, SIP-ISDN and SIP-PSTN interworking software.
- Netbricks (news - alert) (http://www.netbricks.com) offers SIP-BRICKS, a high-performance SIP implementation. Netbricks is a leading developer and supplier of Portable Protocol Stacks compliant with Protocol standards.
- The SIP Toolkit from ELUON (news - alert) (http://sip.eluon.com/index.jsp).
- The EmSIP embedded SIP Stack from Parrish Systems (http://www.parrishsys.com).
- PJSIP, an Open Source SIP Stack, supporting many SIP extensions/features (http://www.pjsip.org).
- MjSip is a complete java-based implementation of a SIP stack available as open source under the GPL license (http://www.mjsip.org).
- The VBVoice service creation environment from Pronexus (news - alert) (http://www.pronexus.com).
- The SIP control interface and a management agent from SandCherry (news - alert) (http://www.sandcherry.com).
- The HiPath Software Development Kit from Siemens (news - alert) (http://www.siemens.com).
- The High-Performance SIP Stack from Jugphoon (news - alert) (http://www.juphoon.com).
- VQmon from Telchemy (news - alert) (http://www.telchemy.com) is integrated into IP Phones, VoIP Gateways, Residential Gateways, SLA monitoring systems, routers, OSS, Probes and Analyzers,. and provides reporting metrics using SIP, RTCP, XR QoS Reports and other key protocols.
- The TsGATE software SIP to PSTN Gateway design kit from TeleSoft International (news - alert) (http://www.telesoft-intl.com).
- The VeriCall Edge is VoIP embedded software platform that offers the media processing; and SIP call control functionality needed to develop a video and VoIP enabled client device.
- The Fusion SIP Software Suite from Unicoi Systems (http://www.unicoi.com/fusion_net/fusion_sip.htm), designed for use in embedded devices, has a small footprint and is completely ROMable and re-entrant. The Fusion SIP stack provides a simplified API for quick integration with the application and transport stack (TCP/IP/UDP). (news - alert)
- The SIP Protocol stack from Wind River (news - alert) (http://www.windriver.com).

As developers create new communications services in existing wireline and wireless networks — as well as the upcoming IMS network — many of these tools will be getting quite a workout (and will doubtless evolve along with SIP and IMS). Even more new tools will appear as a plethora of mobile devices and technologies make their way into enterprises and consumer residences.

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

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SIP Test Tools

By Chris Bajorek

Editor's Note: Developing and deploying SIP applications and services demands superlative testing tools for proof-of-concept, load, security and interoperability testing. Professional testing labs are obviously treasure troves of such technology, so we decided to take a peek into one of the best known of such labs, CT Labs of Rocklin, California, an independent operating unit of Empirix, headed by long-time testing guru Chris Bajorek. In this article, Bajorek takes us on a guided tour of his tool collection.

Here's a rundown on the various classes of test tools we use at CT Labs for SIP-oriented projects. Because of the range of products that we test here (from consumer to enterprise to carrier class), it's a comprehensive list.

High Density Call Load Generators

When it comes to verifying call handling performance at rated capacities, we use a number of different call load generators. Load generators need to issue high numbers of simultaneous calls but not necessarily navigate complex telephone or voice-controlled user interfaces. A key feature to us: the ability to estimate voice quality for every active call. By the way, TDM is not dead yet: Gateways still exist in all sizes from small to very large that need to be tested at rated loads.

Specific tools we use for SIP-oriented tests:

- Empirix (news - alert) Hammer NXT-IP for ultra-high density VoIP testing. The NXT-IP gives packet performance metrics as well as a voice quality score for every active test call.
- Empirix Hammer NXT-TDM for ultra high density TDM load tests. We use this mostly to test high capacity gateways.

Call Feature Testers

Call feature testers can also be used as load generators for lower density SIP products. But their primary use is to validate call and application features. For example, an IP PBX with integrated voicemail can be thoroughly regression tested using a call feature tester. Test scripts can be created that not only verify the PBX's ability to route calls and execute a variety of call features correctly, but also can fully verify the voicemail application itself under real-world traffic conditions. We have developed scripts that will even verify the quality of the voice messages recorded and played back by emulated users under peak load conditions.

These call feature testers come in two flavors: VoIP and TDM. We still use our TDM feature tester/call generators in a surprising number of projects. One recent example: we created an automation test suite that can verify a wide range of call features for terminal adapter devices that interface POTS phones to broadband VoIP services (e.g. adapters that work with Vonage, AT&T Call Vantage, etc.).

Specific tools we use for SIP-oriented tests:

- Empirix Hammer FX-IP for VoIP feature and moderate density load testing.

- Empirix Hammer FX-TDM for TDM feature and moderate density load testing.

Network Emulators

To verify real-world performance, you cannot simply test a VoIP product in a 100% clean network environment. We have seen too many early to market products work fine on a sterile network but have serious difficulties when presented with typical impaired network conditions. To get a true picture of how a SIP or IMS device will operate under real-world network conditions, you need a device that will emulate conditions of packet loss, latency, and jitter. We use our network emulator quite often for this purpose.

Specific tools we use for SIP-oriented tests:

- Empirix Hammer PacketSphere XG

Registration Generation Tools

When you think of testing a SIP device or a staged IMS network for performance, you don't usually think of registrations as a major traffic component. But in fact, the number of registration-oriented SIP messages that can exist on a network under certain conditions can cause serious performance problems. For example, a registrar that is serving a regional VoIP access network can be overwhelmed when power to a region has been restored after an outage. When power comes back on, all those SIP devices that were off are now alive and trying to register at the same time — which can easily swamp out an ill-prepared server. Our SIP registration generator has been very busy lately in a variety of performance tests.

Specific tools we use for SIP-oriented tests:

- Empirix Hammer DEX with registration template.

Device Emulation Tools

Many tests that we run in our lab involve staging the test target, a variety of supporting VoIP network devices, and emulation of the subscriber base endpoints via feature and call load generators. There are two basic ways to stage a "supporting-role" VoIP device: (1) buy a 3rd party device and integrate it into the test setup, or (2) use a tool that allows you to emulate the features and functions of that device. For example, staging an IMS network in our lab requires the inclusion of an HSS (Home Subscriber Server) device. With the tool we are now using, we can emulate the HSS and get the benefit of deeper and easier to access metrics. In addition, we only have to learn the device emulation tool interface as opposed to learning to configure and use many different third-party devices. This is quite a powerful concept.

Specific tools we use for SIP-oriented tests:

- Empirix Hammer DEX (device emulator).

Application traffic generators

When we stage certain types of tests, we not only need to generate VoIP traffic but also complementary traffic that is handled by the system under test. For example, testing a service provider-grade firewall with a SIP-oriented ALG (Application Layer Gateway) might require staging of VoIP traffic in combination with other "Internet mix" traffic such as http, POP3, ftp, etc. Other types of tests might require generation of more application-specific traffic such as IPTV or video on demand. So the goal of adding these traffic components is to create as close to a real-world traffic mix, including VoIP, as is possible in a lab setting.

Specific tools we use for SIP-oriented tests:

- Shenick diversifEye 8400 (http://www.shenick.com).

Monitoring/Analyzer Tools

We use two kinds of monitoring and analyzer tools: a SIP trace/protocol analyzer and a device resource metrics/monitoring tool. The protocol trace analyzer is an absolutely essential tool that allows you to debug low level problems with the handling of SIP messages and their associated media packet streams. The device resource monitor that we use enables us to hook into the system under test platform(s) and see in real time the effect of the various traffic loads on the device's available internal resources (CPU usage, memory usage, disk usage, etc.).

For example, we recently completed a test on a highly scalable SIP-based VoIP communication system that was staged on a series of servers connected to a high-capacity storage device. Our resource monitoring platform allowed us to record the real-time utilization of CPU, memory, and storage resources while we varied the number of emulated subscribers and types of traffic. When the test was done we were able to graphically line up the results, explain to our customer where the bottlenecks were, and point them in the right direction for improvements.

Specific tools we use for SIP-oriented tests:

- Empirix OneSight (device resource metrics/ monitoring tool).
- Empirix Hammer Call Analyzer (protocol trace analyzer).

Security and Robustness Test Tools

This general class of tools (often referred to as protocol "fuzzers") includes tests designed to flush out problems and vulnerabilities with lower-level SIP stacks and parser mechanisms. The need for SIP-oriented robustness validation is important now, but will be even more important as the proliferation of SIP-based communications grows and malicious VoIP denial of service attacks become more prevalent.

Using these tools you set up a special test that attempts to launch call sessions toward the system under test using SIP messages that have been "fuzzed" (intentionally corrupted) with the purpose of exposing weaknesses in the lower-level parsing and SIP message handling layer. The device under test is monitored for the appropriate response to each test case.

The first and second tools listed below take a bit of setup effort but the price is hard to beat: free.

The third listed tool — from Mu Security — is a relatively new entrant in the robustness testing marketplace; in the short amount of time that we've been using their platform we have come to respect its importance in terms of better preparing our customer's products for the real world. In effect, their tester allows you to expose a wide variety of protocol and attack vulnerabilities (and not just for SIP) that could otherwise leave your product gasping for air. I strongly suggest you check out their website for more info.

The CT Labs SIP Attack Tool (not for sale) is designed to launch a range of randomized SIP messages at up to gigabit wire rates toward the system under test. The tool has proven to be extremely effective in validating products for protection against this special class of SIP-specific attacks.

Specific tools we use for SIP-oriented tests:

- PROTOS Test-Suite (http://www.ee.oulu.fi/research/ouspg/protos/testing/c07/sip/).
- IETF SIP torture test suite (http://www1.tools.ietf.org/html/draft-ietf-sipping-torture-tests-01).
- Mu Security Mu-4000 (http://www.musecurity.com).
- CT Labs SIP Attack Generator.

Chris Bajorek is the Director of CT Labs (http://www.ct-labs.com), an independent operating unit of Empirix Inc. (http://www.empirix.com).

ADVERTISING INDEX

Q & A

60 Seconds with Mark Foster, Senior VP and CTO of NeuStar

by
Richard "Zippy" Grigonis

Mark Foster has been with NeuStar (http://www.neustar.com) since November 1999. He was an independent consultant from 1994 until November 1999, during which time he was the lead inventor of local number portability, conducted the first industry field trial of local number portability in the Seattle area, and was heavily involved in industry technical, policy and regulatory discussions leading to the adoption of local number portability. Foster has also held important positions with Stratus Computers, Phone Base Systems, and Quest Communications. He holds a bachelor's degree in physics and computer science from the California Institute of Technology.

Richard "Zippy" Grigonis recently spoke with Mark Foster about SIP, IMS and ENUM.

RG: Most people probably don't realize that NeuStar is not only heavily involved with SIP but also with ENUM — tElephone nUmber Mapping, or "Electronic NUMbering" as it's also known.

MF: NeuStar helped define SIP at the IETF. NeuStar's Jon Peterson wrote most of the core RFP documents for SIP. Obviously we've been one of the lead developers and contributors to the core SIP standard, as well as ENUM. We're also active with a different group called the Liberty Alliance, and that also deals with identity management. Indeed, we've brought a lot of that work into the IETF, and one of the most recent areas of work on SIP itself relates to closing the final security issues around SIP, which is what we refer to as SIP Identity. We think this is the critical final part of the core SIP standards, to make SIP truly secure on an end-to-end basis across a large number of differing usage scenarios.

RG: Will SIP retain its identity as IMS [IP Multimedia Subsystem] takes off?

MF: Frankly, an area of very strong commercial and deployment interest we see now around SIP is actually in a pre-IMS context. IMS is fairly large set of technology standards, protocols, hardware — a whole huge architecture. Many industry segments are trying to understand what the business case is; in other words, what new revenue or cost savings can be achieved by undertaking large potential IMS infrastructure investments. While this massive IMS work is underway, we're seeing a quiet revelation among various developers and operators that SIP technology is actually far more deployable today in terms of addressing immediate problems and generating very real revenue streams. Not coincidentally, this is a major theme of NeuStar's work; specifically, our recent acquisition of FollowApp, which specializes in next-gen messaging products for mobile operators. Mobile messaging is the second biggest revenue stream for mobile operators — about $60 billion year generated by 1.3 billion users. Most mobile operators offer text messaging, which has remained unchanged for about 10 years, and which in turn is built on 30-year old SS7 technology. FollowApp's 19 mobile network operator customers worldwide have already bought, deployed and are in the process of launching a next-gen messaging product that uses SIP at its core and will be used to upgrade existing mobile messaging products, such as text and multimedia messaging. FollowApp adds essentially instant messaging features, such as presence, to mobile messaging. The ability to use SIP-based technology to upgrade an existing revenue segment for mobile operators is a huge opportunity. It does two things: The addition of presence and multimedia boosts the mobile messaging volume of individual subscribers 4 to 6 times over current usage rates. At the same time, it drops the cost of using existing SS7-based technology, which adds up to a third of the total cost.

At 3GSM in Barcelona we saw that the biggest single area of activity in the mobile industry today is the emergence of this next-gen mobile IM capability. It's exciting because it's SIP-based at the core and we think that over time it will not just revolutionize messaging, but also will help to introduce in a gradual, phased way, the core essence of IMS, without the need for making available the total IMS infrastructure from the outset.

This is a completely different approach from what most operators in the industry have focused on as it relates to SIP and IMS technology deployment. It's much more phased, seamless and organic — and revenue- and cost-oriented. We're talking about making a huge revenue and cost savings impact today, as opposed to groping for the IMS promise of the future. So what's not to like about that?

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

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