Prof. Henning Schulzrinne on Enterprise SIP

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SIP MAGAZINE™

The Authority On Session Initiation Protocol

On the Edge: SMB 2.0

Speaking SIP: A Protocol for all Sessions?

Deploying Next-Generation SIP Services

Beyond The Firewall

SIP Enables Global Connectivity

EDITOR'S NOTE

SIP: Spanning the Globe

by
Greg Galitzine

SIP is truly a global phenomenon. Carriers from across the world are taking advantage of SIP as they upgrade their networks to better serve their customers.

Russian VoIP carrier Tario Communications (news - alert) recently announced that they have implemented the CommuniGate Pro Internet Communications platform from CommuniGate Systems. (news - alert)

Tario's move was necessitated by the need to upgrade its internal network to meet the growing demands of serving its 300,000-strong customer base, while hosting over 60 million minutes of phone calls and messages.

The carrier chose the CommuniGate Pro Dynamic Cluster with SIP Farm, which in addition to providing five-nines reliability, redundancy, and scalability, enables Tario to allocate subscribers and telephony class of service based on traffic and regional geographic placement. The new system also provides the infrastructure for SIP trunking in addition to standards-based e-mail messaging.

Back to the good ol' USA, and right to the heartland… Time Warner Telecom recently announced the successful installation of its SIP IP connections for VoIP services to Boise State University.

Boise State University's campus-wide deployment of VoIP technology over its existing Time Warner Telecom metro Ethernet service encompasses over 14,000 telephone numbers and 4,000 handsets making it one of the largest university VoIP deployments nationwide.

Time Warner Telecom's 20 MB SIP trunk service replaces existing T1s to boost bandwidth by nearly 20 percent, as well as allowing Boise State's IT managers to connect directly to a VoIP PBX, thus enabling the retirement of six previously required gateways necessary to convert digital voice signals to IP protocol.

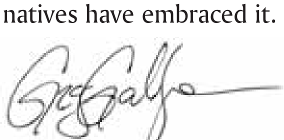
And Sawtel has decided to deploy Pactolus Communications' (news - alert) SIPware Broadband Telephony (VoBB) over their commercial satellite and wireless telecommunications network serving customers on the island of Nassau, Bahamas.

If nothing else, Sawtel (news - alert) is providing a showcase example of how to extend IP services across new networks and multi-network delivery paths.

SIPware Broadband Telephony delivers the advanced Voice-over-IP services for converged voice, data and video service for Sawtel's business and residential consumers.

SIPware services were selected for its capabilities, including prepaid billing, which enables Sawtel to serve all subscribers, be they successful businesses to individual consumers with, shall we say, "credit challenges." The service is initially being provided to users all across Nassau, and will be successively offered to user populations around the world needing to overcome service accessibility, reliability, and economy barriers.

So it seems that no matter where you go in this world SIP has been there, and the natives have embraced it.



SIP MAGAZINE™

Rich Tehrani, *Group Publisher and Editor-In-Chief* (rtehrani@tmcnet.com)

EDITORIAL
Greg Galitzine, *Editorial Director* (ggalitzine@tmcnet.com)
Erik Linask, *Associate Editor* (elinask@tmcnet.com)

TMC LABS
Tom Keating, *Executive Technology Editor/CTO/VP* (tkeating@tmcnet.com)

ART
Lisa D. Morris, *Senior Art Director*
Alan Urkawich, *Art Director*
Lisa A. Mellers, *Graphic Designer*

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Editorial Offices: 203-852-6800 Customer Service: For all customer service matters, call 203-852-6800.

ADVERTISING SALES
Sales Office Phone: 203-852-6800

Anthony Graffeo, *Sr. Advertising Director - Eastern U.S.; Canada; Israel* (agraffeo@tmcnet.com), **ext. 174**
John Ioli, *Advertising Director - Midwest U.S.; Southwest U.S.; International* (jioli@tmcnet.com), **ext. 120**
Drew Thornley, *Business Development Director - Western U.S.* (dthornley@tmcnet.com), **(480) 833-8836**

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feature articles

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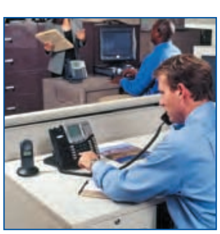

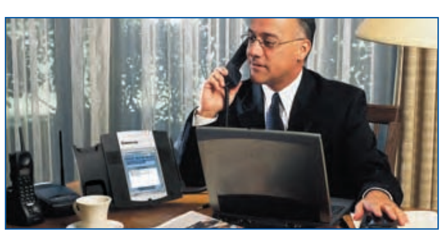

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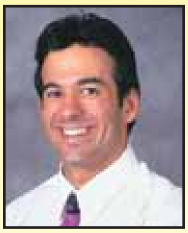


by
Rich Tehrani

My recent presentation at Tech Data's reseller conference was one of the most beneficial talks I have ever given. I always solicit audience feedback and this session had more feedback than any other I have given recently. When I say beneficial — in this case it was beneficial for me and, from what I gathered from the audience, they got a lot out of it as well.

When I got to the conference, I had a presentation ready to go as I always assume the audience will be shy. But by the fifth slide or so, someone asked about how he can assure quality in phone calls across the open Internet and I explained that you can't at the moment, but you can work with a service provider who will give you an SLA and help you achieve the quality you seek. Someone else in the audience then chimed in about managed networks. The room erupted into a discussion about this topic for a while and we touched on the fact that even an ATA has mechanisms to assure that voice is given priority.

We (the collective — the VARs in the room and I) narrowed this answer down to the quality of voice in the office and the quality once it leaves your office. These are two separate things. Some knew this already but it was beneficial for the newcomers to get this straight early on.

When I got to my slide on interoperability one of the resellers explained how there isn't much interoperability today and it is tough enough to get the equipment running and working through firewalls. He went on to say with all this complexity, forget convincing end-users that they have a need for SIP so things can work together. This VAR is installing very large systems, certainly not what a typical reseller would make a living doing. I explained how many vendors are wholeheartedly embracing SIP and some of these companies don't even sell phones but rely on Polycom and others. This to me is a sure sign of companies listening to customers.

> The resellers explained how there isn't much interoperability today.

At one point I mentioned that Avaya says they are supporting Cisco IP phones with full functionality. In fact I was in a meeting with Avaya's Lawrence Byrd recently when he told me this is happening. I said that the SIP interoperability issue seems like it is solved for the foreseeable future but we have some time before it all plays out. The resellers were skeptical but I promised them that Avaya was responding to customer demand. They asked if the support exists today and I said as far as I know it does.

Someone in the audience asked why you would need multivendor support and I explained that there are acquisitions that happen all the time that make this an important need. In addition, something I didn't mention is that having interoperability means you can buy equipment based on features and functionality and gives a better cost/benefit ratio. In addition it allows you to be secure in purchasing decisions — even if one of your vendors goes under.

Someone in the audience likened the Avaya (quote - news - alert) news to Apple supporting Intel chips. I agreed saying that Intel chips have allowed Apple computers to achieve better price/performance.

From there the discussion turned to the subject of SIP trunks, which everyone acknowledges are working well and moreover these trunks (so far) do not suffer from interoperability problems.

In my presentation, I explained to the resellers that I am in contact with vendors 95 percent of my time and the most refreshing conversations I have are with resellers who are actually implementing products. I learn more from a bunch of resellers in 30 minutes than I can in any other place.

I also explained that in the magical world of vendors — everything does indeed work perfectly. There was lots of head nodding and laughter at this point.

A few minutes later, one of the resellers raised his hand and said he has thrown all 3Com equipment out of his office and his house, and is looking for an alternative. I wish now that I would have asked why he had problems with 3Com, and more importantly why he had their equipment in his house.

Another reseller started to talk about the Linksys One product and said it was great. Yet another reseller mentioned Allied Telesyn (news - alert) and how he loves their products. At least one other person acknowledged how good their products are. The sole complaint was a lack of a call park button on one of their devices.

The group voiced their concerns about pushing products that are available on the Internet or through retail outlets. Most of the room agreed that it is better not to sell products that are available through some of these low-price retail outfits but some didn't think it was a problem to compete with retail channels as they see themselves as also having value in connecting equipment on the network.

I told the crowd about Epygi (news - alert) as a good alternative for the low-end market. No one seemed to have heard of them but many wrote the name down.

I explained that the voice business will not last forever — well it will, but margins will be squeezed and more products will go retail. I told them they need to focus on applications; if we don't do this now we as an industry are doomed, I said.

Everyone seemed to agree. I mentioned that vendors are

integrating their products into software such as MS Outlook or Internet Explorer. I further went on to talk about CRM integration and Just in Time Communications.

I gave an example of having an instantaneous conference call with a group of people — where they are connected via chat, or phone, and the type of phone (wired, wireline) is immaterial. I also said that we would have full screen sharing/collaboration capabilities and that this is the future and will make us much more productive.

I continued that we need to wow our customers with enthusiasm. We need to show them where things are going. I cited the Nextel example and discussed how much more productive people are when they use these walkie-talkie like devices. I also — in full disclosure — said I hate these phones because the beeping is so annoying. Some of the room agreed and have actually switched back to regular phones due to customer complaints. But for applications where the noise is not a problem, these phones really speed time to decision.

In the end the audience got it and they seemed to agree. But they are not convinced of SIP interoperability. The discussion came up that many of the smaller vendors use SIP and do in fact interoperate. Surprisingly, many of the resellers in the room are concerned about reselling products from companies they have never heard of because they aren't sure these companies will be around tomorrow to support and service the products.

Tech Data is aggressively recruiting telephony resellers

The irony is that most small equipment providers don't realize how important it is to market to this group to ensure resellers will seriously consider them. Sometimes small VoIP equipment entrepreneurs are their own worst enemies as they have great products but think the world is obligated to find out about them and resell and buy their products without actually telling anyone about what they do.

Marketing and promotion is all about branding and giving off the image of a company of stature. Unfortunately many of these companies are led by engineers and will never understand how important it is to build their brand. For the majority of entrepreneurs, brand building is on the list of priorities just after opening a satellite office in Antarctica.

Being an engineer myself I understand how introverted we can be as a group — too bad so many VoIP entrepreneurs have the engineers making budgeting decisions in PR and marketing.

To be honest I was stunned at the extent to which VARs tend to avoid companies they never heard of.

Back to the talk, the group seemed very Cisco oriented with the exception of Allied Telesyn and Shoretel. Not surprisingly there was much Linksys knowledge in the room as well. The group actually seemed anti-Avaya and others echoed the negative 3Com sentiments to me after the talk was over. I am not sure if this group is isolated in their opinions or whether there is a trend forming here. I will be speaking in Atlanta soon to more resellers and hope to delve into the concept of brand perceptions further.

I did ask the room about telecom knowledge and about 30 percent of the room has sold telecom equipment for at least ten years. There were a bunch of VoIP newcomers as well.

In all, Tech Data did a great job getting these resellers to Kentucky to become educated on VoIP and other topics. Without a strong reseller market and without the proper training, there really is no enterprise IP communications or SIP market.

What's New With Tech Data?

I had a chance at the event to sit down with Charles Bartlett, Jr., the VP of Product Marketing of the Networking Division of Tech Data. I asked him how his business is doing and he told me companies are looking to outsource more and more of the things that are not their core competency. Some examples he gave were manufacturing and logistics. He says there is a trend to outsource sales and marketing as well.

I asked about how strong his reseller base was and he told me there are 90,000 resellers worldwide that purchase from them and about 30,000 of them are active at once.

One of the benefits the company offers their partners is access to a sales and marketing solutions center consisting of products and services from 80 Tech Data business partners. The telephony program is small but growing at the moment with 300 resellers in this program.

In the late nineties I witnessed the traditional computer distributors showing a renewed interest in telephony as well. At that time many vendors complained to me that the costs were too high to get involved with the traditional distribution channel. I told Chuck this fact and he said that his company can show a significant ROI for a product that the market wants.

They also can be flexible with smaller companies, coming up with an agreement that allows Tech Data to see less up front and more margin on the back end of the agreement.

But they are looking for the right type of partner, which means you need to have a product that fills a gap in their product line or delivers a price point not currently found in their portfolio.

Tech Data is also aggressively recruiting telephony resellers as this part of the business is doing very well.

In terms of the success of the telecom division, Chuck tells me they have experienced 35 percent compounded growth in the telephony SBU for the last three years. And, that's not so bad, now is it?

VoiceXML Gains Acceptance as Industry Standard

By Mae Kowalke

Microsoft (quote - news - alert) announced that Speech Server 2007 - the latest version of the company's speech and telephony platform - will provide full support for VoiceXML. VoiceXML is an open standards-based language used for developing speech applications, managed by the World Wide Web Consortium (W3C).

Microsoft said the addition of VoiceXML compatibility (Speech Server continues to support Speech Application Language Tags, or SALT) will "enable customers to choose the development standard that will work best in their environment."

According to Clegg Ivey, Vice President of Operations at Voxeo, Microsoft's adoption of VoiceXML is a good indication that the language has matured into the industry standard. "When Microsoft signs on to a standard, you know you're in good shape," he said.

Because of the VoiceXML standard, companies can deploy speech products knowing that when they upgrade or switch vendors down the road, they won't have to re-write all their applications.

VoiceXML can be used to create any type of interactive speech application, including voice-enabling of call centers, interactive tracking applications, and outbound notification.

http://www.microsoft.com

Back to the Future: MindSpring Redux

Can a blast from the past ring in the future? EarthLink (news - alert) thinks so. Bringing back one of the 90's most successful Internet brands, the company relaunched MindSpring. The new MindSpring is EarthLink's softphone client that combines Internet voice and instant messaging on a consumer's PC.

Integrating popular Internet applications and Voice over IP (VoIP), MindSpring delivers high-quality, crystal-clear voice calling capabilities and real-time text (instant messaging) and access to free voicemail and EarthLink Web Mail. This combination allows MindSpring users to have one identity for e-mail, instant messaging, and phone calls over the Internet.

MindSpring is built on the SIP (session initiation protocol) Internet standard, a platform that makes it compatible with any other open SIP-based network. MindSpring and Google Talk are also interoperable for instant messaging.

http://www.earthlink.net

Juniper Develops Tool for MSOs to Secure VoIP, SIP Services

Juniper Networks Inc. (news - alert) has unveiled a new product, the Cable Dynamic Threat Mitigation tool, which helps cable MSOs protect VoIP and SIP services against network security attacks.

The Cable Dynamic Threat Mitigation solution works with any PacketCable Multimedia-compliant cable modem termination system. It integrates Juniper's routing platforms, intrusion detection and prevention devices, and SDX Service Deployment System so MSOs can identify and isolate security threats, then notify operations staff and infected customers.

The intrusion detection and prevention device monitors for network attacks on a per-user, per-application, or on-demand basis, and detects offending traffic or a DDoS attack. It then notifies the SDX-300, which sends the appropriate policy to any Juniper Networks routing platform and/or third-party PacketCable Multimedia-qualified cable dynamic threat mitigation or metro routing platform. According to the nature of the threat, MSOs can choose to rate limit or filter traffic, drop the call, or redirect the user to a Web portal, and automatically re-establish service once the threat is eliminated.

Juniper's Cable Dynamic Threat Mitigation tool also can provide security for SIP-based services, such as online gaming, video conferencing, instant messaging, and push-to-talk.

http://www.juniper.net

Woize International Ltd. to Launch Symbian Client

Woize International Ltd., (news - alert) the digital communication service provider today announces plans on launching a Session Initiated Protocol (SIP)-based Symbian Woize client for the benefit of worldwide users during Q2. Symbian is an operating system based on open standards for smartphones.

By launching a SIP-based Symbian 9.1 client, Woize creates an opportunity for worldwide advanced phone users to benefit from the Woize services, thereby reaching even further into the opportunities of the market of free mobile phone calls enabled by WiFi networks.

The Symbian Woize client is planned to have similar functionalities as the SmartphoneWoize client, allowing users to make free calls to other Woize, PocketWoize, SmartphoneWoize, and Symbian Woize users, making inexpensive calls to users outside of the Woize community, and receiving phone calls to the client anywhere in the world.

http://www.woize.com

Dialexia and Grandstream in SIP-eroperability Testing Deal

By Erik Linask

VoIP solutions provider Dialexia Communications (news - alert) and Grandstream Networks, (news - alert) which provides IP telephone hardware, have successfully completed comprehensive interoperability testing between Dialexia's IP-PBX Dial-Office and Grandstream's GXP2000 SIP IP Phone.

The partnership is set to allow future service providers and enterprise customers easy VoIP deployments. With the need for connecting distant branch locations, while maintaining the functionality of existing PBXs, customers are looking for an edge, which often is found in ease of implementation - and cost, of course.

Dialexia's flagship product, Dial-Office, is one of the lowest cost VoIP solutions available to enterprises. It is designed to easily integrate with legacy TDM and IP networks, thus offering customers the dramatic savings that often drives them to a VoIP solution.

Grandstream Networks' GXP-2000 is a next-generation enterprise IP telephone based on open SIP standards. Built upon unique and proprietary technologies, it features superb audio quality, rich functionalities, and excellent manageability at affordable prices.

The Grandstream GXP2000 IP-Phone combined with Dialexia's Dial-Office brings a rich feature set and extraordinary value to both consumer and corporate markets. With adherence to the SIP standards the products are well suited for enterprise users of IP PBX systems, remote office users, and many other cost saving and productive uses.

http://www.grandstream.com
http://www.dialexia.com

Ubiquity Launches Ubiquity Developer Studio to Speed Deployment of SIP Applications

Ubiquity Software, (news - alert) developer of standards-based SIP (Session Initiation Protocol) application servers, announced the launch of Ubiquity Developer Studio (UDS), a development and debugging platform for its SIP Application Server (SIP A/S).

UDS provides a simplified set of tools that enables SIP developers to build highly modular applications and services that run on top of the Ubiquity SIP A/S.

UDS is designed to work in conjunction with the Ubiquity Appcelerator product set, an optional add-on to SIP A/S. Appcelerator provides a unique SOA including a Service Oriented Object Framework (SOOF) with reusable code components. And by using Eclipse, developers get an enterprise or "IT" perspective on real-time telecommunications applications software development, which makes it easier to develop converged applications that work with carrier services or within the enterprise.

Applications created with UDS can be built tested and debugged in Ubiquity Developer Edition, so you can test and debug right on your desktop.

http://www.ubiquity.net

Genesys Readies Enterprises for IMS World with Release of 7.2

By Robert Liu

With its release of the newest version of its flagship contact center software, Genesys Telecommunications (news - alert) Laboratories is betting that enterprises will eventually fix their IP communications switching infrastructure on the Session Initiation Protocol (SIP) and incorporate elements of the IP Multimedia Subsystems (IMS) architecture.

The centerpieces of the Genesys 7.2 release are the Genesys Customer Interaction Management (CIM) platform, a framework that adapts to any media channel, and the Genesys SIP Server, which acts as a software-based automatic call distributor (ACD) to direct calls to up to 30,000 IP agents through multimodal customer contact channel options that include voice, e-mail or chat.

This new standards-based approach to call control and signaling means Genesys 7.2 can freely communicate with media gateways and softswitches without having to conform to a myriad of different variations on the Media Gateway Control Protocol (MGCP). In addition, the company has incorporated the Real-time Transport Protocol (RTP) to manage the media elements within the network, as outlined by the IMS specifications.

Meanwhile, the Customer Interaction Management (CIM) platform features an Open Media software component that tightly integrates IP into key elements of the suite, such as the Genesys Voice Platform to support voice self-service, video and voice-enabled applications. As such,

Genesys 7.2 is able to incorporate business process routing (BPR) capabilities by applying intelligent real-time routing, tracking, reporting and management capabilities to work items and interactions over any channel inside the contact center and across the enterprise.

http://www.genesyslab.com

DiVitas Networks Announces Interoperability with Trapeze Networks' WLAN

DiVitas Networks, (news - alert) a provider of seamless, unified mobility solutions for enterprise networks, announced that it has successfully tested the interoperability of its platform with Trapeze Networks, (news - alert) the provider of the wireless LAN (WLAN) Mobility System, MP-372 access point, and new MX-200 and MX-216 series of switches to enable enterprise campus mobility - and the industry's first demonstrable Fixed Mobile Convergence for WiFi and Cellular.

DiVitas and Trapeze successfully tested a VoIP-over-WiFi application, handing off VoIP calls from the WLAN to the cellular network and back for seamless roaming over the best available network.

Leveraging cellular networks, Internet/Wi-Fi technologies, and dual-mode handsets, DiVitas delivers mobile voice and text on a single platform, both within the enterprise and outside it. The DiVitas solution also serves as an application platform for enhanced communications capabilities at extremely attractive cost points. Above all, the DiVitas solution is controlled by the enterprise. Administrators can now centrally manage mobile communications independent of the network that delivers them.

http://www.trapezenetworks.com
http://www.divitas.com

RADVISION Releases New SIP Toolkit With Java APIs

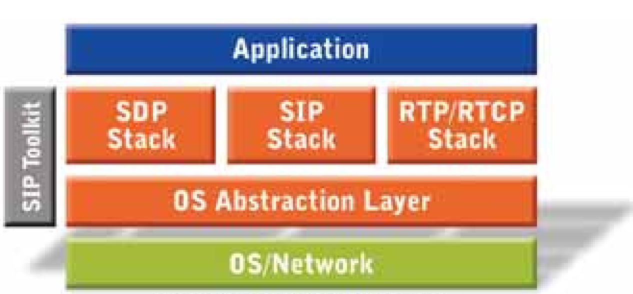
RADVISION, (news - alert) a provider of multimedia conferencing and communications platforms, has released a Java version of its Session Initiation Protocol (SIP) toolkit for real-time communication applications, such as VoIP and IM. The new toolkit significantly shortens development cycles, thereby enabling even faster time-to-market for a variety of new and innovative services.

Java programming language, which is rapidly gaining popularity among developers, can be used to write software on one platform and run it on another, create programs that run within Web browsers, develop server-side applications, and write applications for cell phones and other consumer communication devices. Java SIP is a fundamental building block for Web-based endpoint-oriented communication applications.

The Java SIP Toolkit can be used to develop click-to-talk applications, as well as a variety of applications for web call centers, and soft phones for laptops and PDAs. It also can be used as a major building block for innovative and time-critical advanced telecom services on the server-side.

Similar to all of RADVISION's SIP toolkits, the Java SIP toolkit addresses carrier-side applications for softswitches and application servers. The Java SIP Toolkit supports operator-grade loading of busy hour call attempts (BHCAs) and has enhanced carrier-grade features, including advanced SIP signaling functionality. The small footprint allows efficient operation in SIP terminals and IP handsets, as well as network-side implementations.

http://www.radvision.com



SIP Toolkit Architecture

UPC Broadband Signs One Millionth VoIP Customer

By Erik Linask

UPC Broadband, (news - alert) the European broadband division of Liberty Global, Inc., whose networks pass more than 11 million homes and businesses in Western Europe is using solutions from both Nortel and ARRIS, as well as several others, in growing its European network. In fact, over the past 15 months, UPC's telephony deployment has doubled and it has now signed its one millionth VoIP customer.

UPC is also one of the first cable operators to deploy SIP (Session Initiation Protocol) -based VoIP services, laying the foundation for new, feature-rich voice and multimedia services, like presence, instant messaging, video and mobility, and taking the first step towards the converged services.

"As consumer and business demand for multimedia services grows, operators want scalable, reliable and open VoIP solutions that deliver services in a timely, reliable and cost-efficient manner without abandoning existing network investment," said Steve Pusey, executive vice president, Nortel and president Nortel Eurasia.

UPC provides high quality VoIP service that is attractively priced to offer a real alternative for consumers to their existing telephone service. Its deployment of VoIP-based equipment - like softswitches and gateways from Nortel and customer premise equipment from ARRIS - allows UPC to provide enhanced triple play services to its customers. There are plans to develop a VoIP/SIP-based converged product portfolio over a single IP network to both fixed and wireless networks. The goal is to introduce integrated fixed/mobile services in Switzerland and other key markets based on these SIP softswitches.

http://www.lgi.com

The New PacketCable: SIP ... and More

The new specs announced this week by CableLabs Inc. (news - alert) for its PacketCable system bring the cable industry closer to adopting SIP, but still refrain from specifying SIP as the only option for cable operators deploying VoIP services.

The current signaling system used by cable operators, called NCS, does not allow the portability that SIP supports. Unlike SIP, customers cannot plug their IP phones or terminal adaptors into the Internet anywhere and get service. Therefore, the services closely resemble the PSTN, with a few extras, such as Web interfaces to configure features.

The changes to PacketCable signal that the leading operators in the cable industry will continue their relatively conservative approach to VoIP in the short term, but the specs also open the door for future services, such as WiFi VoIP, and combinations of voice and other media, that take advantage of the portability of SIP and its support for multimedia.

Also included are select aspects of IMS, chosen for their relevance to the cable industry.

Cable operators now have a mechanism to do both voice and video telephony. They can combine services across platforms, for example, sending caller ID to a set-top to display on a television. There are also ways to integrate fixed and mobile services. "It gives operators ways to integrate across all those services," Miller says.

The specs developed by CableLabs, the technology development center for the North American cable industry, also deal with some of the knotty deployment issues of SIP VoIP, namely security and network address translation (NAT) firewall traversal. Other new features address QoS and device provisioning.

http://www.cablelabs.com

App Server Vendors Prep for Imminent Combination of IT, Telecom

By Robert Liu

Application server vendors are accelerating their rollout plans to prepare for the imminent blending of information technology with the telecom world, brought on by the influx of IP Multimedia Subsystem (IMS)-based architecture.

Oracle (quote - news - alert) has outlined its roadmap to deliver a standards-based Service Delivery Platform (SDP) for the telecommunications industry to transition from current silo-based network investments into a service-oriented architecture (SOA) to help reduce time and costs to deploy next-generation communication Internet Protocol (IP) voice and data services. Oracle SDP plans to extend its Oracle Fusion Middleware platform with technologies obtained through its recent acquisition of HotSip, adding elements like Session Initiation Protocol (SIP), Presence Server, Proxy Registrar, and Location components.

Oracle's announcement underscores its desire to buy its way into the lucrative telecom market, which is witnessing an economic upswing as standards like IMS and Linux enable the convergence of delivery platforms.

"IT-standards-based service delivery platforms offer compelling value to operators as the basis for developing interactive, media-rich, next generation data services," said Philip Marshall, director, Wireless/Mobile Technologies, Yankee Group. "Service delivery platforms that enable immediate ROI through out-of-the-box services and integration with [operational and business support systems] through standard interfaces are likely to be particularly compelling. Vendors who are able to provide a broad portfolio of products that are stable, mature and carrier-grade will have a definite edge over the competition."

http://www.oracle.com

Cisco Expands Unified IP Phone Portfolio

By Sandra M. Gustavsen

Cisco Systems (quote - news - alert) has introduced a number of new or enhanced IP telephone models, including the 7911G (Global) and 7941G-GE and 7961G-GE (Global and Gigabit Ethernet) models. All support IEEE power and advanced SIP capabilities.

Recently, Cisco also announced Cisco Unified CallManager 5.0, a new software version of the company's IP PBX that runs on the Linux operating system and adds native support for SIP (SIP line and trunk support) in addition to Cisco's existing protocol, Skinny Client Control Protocol or SCCP. With native support for SIP, Cisco customers and channel partners will have a choice of endpoint devices from various vendors for a customized solution. Further flexibility is available to Cisco customers since they can choose devices based on SIP or Cisco's SCCP

Cisco's IP Phone 7911G is the company's newest basic IP phone with the same hardware design of Cisco's earlier IP Phone 7912G, including one line and four soft keys for access to popular telephony features and functions. Designed for business employees with low to medium telephone usage such as those in retail, school or manufacturing environments, the 7911G adds enhanced memory, IEEE 802.3af Power over Ethernet and advanced security. XML support enables text-based applications (including double-byte and unicode characters) to be viewable on the monochrome display. The 7911G works with Cisco's CallManager 3.3 software (or higher) and CallManager Express 4.0.

http://www.cisco.com

Openwave Joins MobileIGNITE

MobileIGNITE, the leading industry association fostering collaboration to accelerate fixed-mobile convergence (FMC), today announced that Openwave, a leading independent provider of open software products and services, is joining MobileIGNITE.

Openwave has agreed to the group's tenets of working on open standards-based interoperability and leveraging SIP (Session Initiation Protocol) with a commitment to IMS (IP Multimedia Subsystem), the industry's leading blueprint for convergence.

MobileIGNITE's Interoperability group is working to define common use cases, interoperability best practices, and test cases from the baseline open standards on which FMC and other IMS-related applications are based.

"As the line fades between mobile and VoIP networks, carriers will be able to give consumers adaptive messaging services that are independent of network, device, or protocol used," said Richard Wong, senior vice president, products and solutions, Openwave. "We look forward to becoming active in the MobileIGNITE association to ensure industry interoperability and accelerate development and deployment of converged services."

http://www.openwave.com

SMB 2.0 — On the Edge

by
Erik Lagerway

Ever since the arrival of the Internet we have been focused on the desktop, after all — this was our interface into the next generation of communications.

While building Xten (now Counterpath), I was exposed to many business plans and product strategies that were years in the making. Many of the carriers' softphone plans were focused on the Enterprise. Near the end of my time there I saw a shift towards the SMB. Now some of the largest operators on the planet are integrating softphones, Outlook plug-ins, browser toolbars, and SDKs that enable rich IP Communications on SMB and Enterprise user's desktops.

We have come a long way since NetMeeting although it took a while before we actually saw the same features in SIP that H.323 applications could offer. If you think about it, NetMeeting was a pretty good app for its time, offering a fairly complete SDK and let's not forget the "callto:" html tags which Skype, and others, have since used as their own.

SIP (define - news - alert) is now leading the way, and we are seeing a plethora of vendors who are enabling VoIP, Video, IM,
and Presence (VVIP) at the desktop using open standards. For a long time I had to listen to naysayers bash the idea of using the PC as a phone but I think those days are all but gone.

Now that SIP can deliver more of the rich enterprise features like Simultaneous Line Appearance we can move graciously between our PC desktop, mobile phone, and desk phone with great simplicity.

There are hundreds of millions of Outlook users out there and most of them are business users ranging from SOHO to large enterprise. Integrating reliable and quality VoIP into Outlook is a no-brainer. Most of the application feature server vendors are on board with this idea. Broadsoft is now offering their Broadworks Assistant in conjunction with their Web-based Comm Pilot and stand-alone Communicator softphone. Sylantro is big on Web 2.0 and is leveraging their close relationship with Microsoft to deliver integrated solutions for Hosted Exchange. I am happy to say that even Counterpath has jumped on board with their new Eyebeam Release 1.5 that has Outlook support out of the box.

I call this shift in technology for the Small and Medium Business… SMB 2.0.

What is SMB 2.0? It is open standards-driven communications technology that is simple, reliable, powerful, and affordable. It is technology which drives services that the SMBs can easily understand and leverage to grow their businesses effectively.

Hosted VoIP is not reliable you say? HA! Hosted SMB VoIP is reliable. I would urge those naysayers to take a closer look at WAN link multi-homing.

Our phone systems of today are old and antiquated — analog from front to back. Few PBXs today have any IP interconnect let alone CTI. This is SMB 1.0.

You may remember this thing called e-mail… How about ATTmail, MCImail, CompuServe, X.400? Remember them? They failed. Something open won. It was SMTP. It became ubiquitous. SIP is to VoIP what SMTP is to e-mail.

Then we have the Unified Messaging era, which boasted converged communications. It was far too expensive and IT departments were not all that interested in opening up their networks to integrate these solutions. Exit SMB 1.5.

Waking up to the dawn of a new era in communications are millions of users within Small and Medium Business, spanning the globe. With hosted SMB VoIP and SMB SIP trunking there is a huge opportunity to deliver every feature a PBX user is familiar with and so much more, at a reduced cost with nearly zero integration into the core of the existing network.

Integrating Hosted SMB VoIP with Outlook or an Internet browser are great ways to leverage an interface users are familiar with to enable VoIP for the SMB. Click to call on Outlook contacts makes a lot of sense. Combine this functionality with feature-rich desktop and mobile IP handsets and we have the makings of a very formidable communications revolution aimed squarely at the small and medium businesses — that's SMB 2.0!

Hosted SMB VoIP and business SIP trunking is already changing the way SMBs do business and the way they communicate. The question is now, "When will your business get on board?" It's here today and it's here to stay.

Erik Lagerway is Chief Technology Officer at Shift Networks. For more information, please visit http://www.shiftnetworks.com. (news - alert)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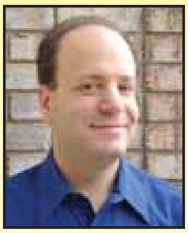

by JD Rosenberg

The Internet Engineering Task Force (IETF) met for the 65th time in late March in Dallas. The meeting was historic for many reasons. It was the 20th anniversary of the IETF, and it was also coincident with the largest rainfall Dallas had seen in 20 years. The downpour was so bad that the parking lots were flooded with two feet of water, and new guests could not check in at the hotel. This downpour was also symbolic of the meeting itself.

Talk of SIP was seemingly everywhere. There was a new IETF area dedicated to SIP and SIP-related topics, new area directors were pulled from the SIP community, and a new pre-working group was looking at peer-to-peer SIP. This trend is happening in the industry, as well. SIP is everywhere. We've now got a magazine devoted to it (this fine publication), and the IP Multimedia Subsystem (IMS), a SIP-based network for service providers, is the new darling of the telecom industry, attracting even more hype than SIP itself.

This deluge has, unsurprisingly, raised a flurry of interest in using SIP for all kinds of applications. SIP is obviously well-suited for interactive multimedia communications, including voice over IP (VoIP) and video telephony. It's also been extended to support presence and instant messaging. These days, it's starting to get mentioned in the context of other applications — interactive TV, gaming, messaging, video on demand, and virtual private networks (VPNs), to name a few.

One of the drivers behind this is IMS, and the argument goes something like this: Service providers are spending a bundle of cash in the development and eventual large-scale deployment of IMS. Its purpose is to serve as a control layer for SIP-based applications in the network. Once they've spent the money, it would be nice to use that infrastructure for other types of applications too. Indeed, most of the applications on that list do have the notion of a session — there is a start and a stop, and usually the delivery of some kind of media throughout that session. Since SIP is all about managing sessions, couldn't it be used there too? Indeed, it's not an unreasonable argument.

Native Versus Encapsulated Use

SIP can be used to support those other applications in two ways. One is natively, where SIP is extended with the new functions necessary to support those applications. The other is encapsulated, where SIP is merely used as a wrapper to signal the beginning and end of a session, while some other protocol is used for the actual application-specific communications. Native solutions are frequently problematic, since they can require substantial changes to SIP and may be unachievable given SIP's design. As an example, one might try to use SIP for Web browsing. Rather than send an HTTP GET command for content, SIP would send an INVITE command to establish a content-transfer session. SIP would work poorly in this model. SIP INVITE requests cannot be pipelined the way HTTP requests can be, and that leads to poor Web performance. SIP also lacks any capabilities for caching, a key piece of HTTP functionality.

The more interesting approach is to encapsulate another protocol in SIP. In the example of Web browsing, HTTP would still be used. A SIP INVITE command would establish an HTTP session; it would be followed by HTTP requests for the content; and the session would conclude with a SIP BYE command. A similar model has been proposed for streaming media, traditionally done with Real Time Streaming Protocol (RTSP). An INVITE command would be used to start the streaming media session, followed by RTSP requests to perform the "trick plays" (fast-forward, rewind, pause, etc.), and ending with a SIP BYE.

The encapsulated mode certainly works. The question is, does it bring any real benefits? It has clear drawbacks: It increases the volume of messaging over access links and within the network, since it is adding to what was used previously. It can also increase the amount of time it takes to establish the session, and it increases the overall processing load in the network.

Benefits Not Assured

The benefits depend on whether the processing performed by SIP servers in the network, such as the Call Session Control Function (CSCF), can now be usefully reused. There are two in particular worth considering: allocation of network quality of service (QoS) and service authorization. In the IMS, the CSCFs enable QoS by communicating with a policy server. That functionality can be reused only if the Session Description Protocol (SDP) body is present in a SIP message. SDP was designed to

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describe multimedia communications sessions using codecs; it cannot readily describe more complex sessions like real-time gaming, which have different traffic properties from those of interactive media, and are not really represented by a codec at all.

Another functionality provided by the CSCF is service authorization; applications are invoked if the subscriber profile indicates that the application is permitted for that subscriber. This is effectively a flag of sorts, indicating whether an application is enabled or disabled for a subscriber. These mechanisms work well for typical interactive communications applications. However, other services may have other needs. For example, authorization for streaming media services is likely to require the use of digital rights management (DRM) technologies to authorize access to content. This kind of functionality is not provided at all by IMS. Thus, the CSCF authorization functions would not help here.

In essence, one must always choose the right protocol for the right job. Different applications have differing requirements, and these demand different protocols. SIP is not well-suited to be the common layer for all IP-based applications. To quote my wise colleague Dave Oran, "There is a reason IP is at the waist of the hourglass, not SIP." The "hourglass" refers to the IP design philosophy, where IP is at the middle, numerous link-layer technologies are under it, and many applications are on top of it, all unified by IP. This hourglass model for IP has proven itself to be right time and time again, with IP serving as the common basis for new applications each year.

As the Internet continues its growth, touching more people and moving faster and faster, the deluge of IP-based applications will continue. Although SIP-based applications will be a part of these, they will only be a part, not the whole.

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. (news - alert)

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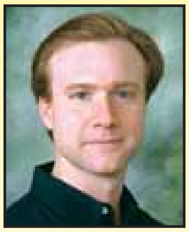
Speaking with Equinix's Lane Patterson



Lane Patterson

***SIP Magazine* had the chance recently to interview Lane Patterson, Director of Research and Development of Equinix. Equinix is a global provider of network-neutral data centers and Internet exchange services for global enterprises, content companies, and network service providers. The company offers co-location, traffic exchange, peering, and outsourced IT infrastructure services.**

Equinix Internet Business Exchange (IBX) centers serve as core hubs for critical IP networks and Internet operations worldwide. With direct access to more than 200 networks, including all of the top global Tier 1 networks, Equinix customers can directly access the providers that serve over 90 percent of the world's Internet networks and users.

Last October, Equinix announced a partnership with Neustar, (news - alert) whereby they would jointly develop a new generation of services to enhance the interconnection of networks providing advanced services under Session Initiation Protocol (SIP).

SIP Magazine editorial director Greg Galitzine asked Patterson about that relationship, about SIP, and about VoIP Peering in general.

GG: What is the nature of the relationship between Equinix and Neustar?

LP: Equinix and NeuStar have entered into a partnership agreement, whereby Equinix will leverage its exchange and peering services at layers 1–3, and NeuStar will connect with SIP/ENUM services at layers 4–7, in order to provide a comprehensive infrastructure for inter-provider VoIP and SIP interconnection. Equinix is well-recognized as a global authority on peering and inter-provider routing, and NeuStar has expertise in SIP/VoIP, ENUM, e911, PSTN numbering, number portability, and call routing. Both Equinix and Neustar are committed to carrier and provider neutrality, and to developing a long-term infrastructure for the industry. The goal is to provide scalable and interoperable infrastructure allowing next-gen services to work seamlessly across networks.

GG: In your opinion, how important is SIP to the future of communications applications?

LP: SIP is extremely important and quite flexible. Because it is easily extended for VoIP, gaming, video, chat, and other services, it will create a unified and scalable signaling infrastructure for a variety of real-time communications, just as HTTP provided a unified protocol for the exchange of hypertext and other content. SIP supports both one-to-one and multiparty communications. SIP hides the private identity of end users, but provides an easy public way of reaching people using a simple "sip://user@domain.com" address that can hide multiple devices and communications capabilities behind it. SIP provides automatic negotiation of capabilities between endpoints, so that the most preferred settings, such as voice or video quality, can be negotiated between callers. SIP has extensions for security and identity, to solve caller-ID spoofing. SIP also adapts well from simple peer-to-peer communications to more controlled carrier environments, where you may have to traverse several proxies and border elements.

> SIP will create a unified and scalable signaling infrastructure for a variety of real-time communications.

GG: How does SIP interoperate with ENUM?

LP: Quite easily. ENUM is based on DNS, and is used to simply translate a TN (Telephone Number) into a SIP Address such as sip:bob@example.com, which gives the IP address of the caller's SIP proxy. So it acts as an IP-enabled phone book. ENUM just handles the lookup for the caller. Once it knows the IP endpoint of the callee, SIP takes over and actually negotiates the end-

to-end session, including all the capabilities such as audio/video codec and security settings between callers, and the tracking of the beginning and end of the call.

GG: What are some of the more exciting applications you are seeing in early deployments or in early trials?

LP: An important early step is the successful use of ENUM to automate IP-to-IP routing of calls so calls no longer have to be handed off to a PSTN switch. I'm also excited about some proposed extensions of ENUM to allow it to return SS7-type information for calls that still must go through the PSTN. During the transition, this will be the majority of calls, but ENUM can gradually replace many of the roles of more costly SS7 infrastructure. Of course, the first application of ENUM that's already in use is to enable users across multiple wireless providers to exchange pictures or rich media between their phones. And SIP-Identity promises to make sure SIP communication does not get filled with SPIT (SPam for Internet Telephony), like our e-mail system has. It can also help run interesting services like age verification that can drive purchasing from SIP-enabled devices. And finally, NeuStar understands e911 for VoIP better than anyone.

GG: How much of an impact will VoIP Peering have on the PSTN in the short term?

LP: In the short term, it will be very gradual. However, if MSOs get organized, they have the potential to cut hundreds of millions of dollars from their settlement costs by routing calls to each other over IP — even walled-garden IP. And it gives them a great infrastructure to launch features such as video calling that can provide a feature hook to continue to lure folks away from their old-fashioned POTS lines. What remains to be seen on the wireless side is if they will try to wait for IMS, or are willing to move forward with SIP peering. And native SIP communications (e.g., calls to sip:user@example.com, and not to TNs) will probably start emerging with the potential to erode PSTN minutes up to about 20 percent. Other elements, such as Google federating Google Talk with other trusted SIP providers, may develop enough push for this at the right scale. It is important to realize that SIP-IX was built with both early-stage and late-stage PSTN-to-VoIP transition issues in mind and with both open Internet and walled-garden architectures considered.

In short,
it will be a long road
for IMS.

GG: What are your thoughts on IMS as a roadmap for next generation networking?

LP: In short, it will be a long road for IMS. There is great concern about the complexity involved, and there will have to be proper integration of multiple IMS "building blocks" into integrated platforms. The good thing about IMS is the discussion being created, and the care being taken to discuss things like authentication and security, QoS and measurement, customer-network and network-network interfaces, etc. However, there is much that can be done with SIP and ENUM, without presuming IMS.

SIP, NAT Traversal and Firewall Issues

By Olle Westerberg

The market growth of real-time IP communications is expected to be the next big wave of Internet usage after e-mail and the Web. SIP has quickly become the standard signaling protocol for this traffic, including VoIP.

However, SIP-based communications cannot reach users behind firewalls and Network Address Translation (NAT). Firewalls are designed to prevent inbound unknown communications, while NAT hides the private IP addresses on the LAN, stopping users on the LAN from being addressed from the outside. Very few, if any, communications are received directly from outside our local-area networks. This provides us with comfort in knowing that only authorized users can gain access to our networks and the valuable information stored on our local servers and computers.

The NAT created on the firewall or by routers is also a part of the security fabric. NATs are necessary primarily because the Internet IPv4 standard does not support enough unique IP addresses to allow all of the devices connected to the Internet to have their own distinctive IP address. With Network Address Translation, only the firewall or router is given a publicly routable IP address. Each device on the inside is then assigned a private IP address that is only known inside the firewall-protected space. This architecture prevents inbound communications from reaching the intended recipient behind the firewall because the IP address of the client device is unknown and not routable.

Finally, most firewalls do not support SIP. Just as with all other protocol types, the firewall must recognize the format of the signaling in order to admit it to the network. Since most firewalls installed today do not support SIP, the inbound traffic will be stopped for this reason alone.

But why is this important?

The vision of true Global Connectivity over the Internet is for SIP to become a universal protocol supported on all devices — from computers to phones — and that we will be able to use those devices to reach anyone, anytime, wherever they may be located at that moment. And it's not just voice. When SIP is widely deployed, the interaction will become more collaborative, with partners, vendors, employees, and even customers using the most effective tool for every occasion whether it is Instant Messaging, presence, voice, video, application sharing, whiteboarding, or file sharing.

The SIP servers providing these functions are often placed on the LAN, but for these to communicate over IP with the outside world they must traverse the firewall.

Eventually, all firewalls will need to be SIP capable in order to support the widescale deployment of real-time communications. In the interim, several solutions have been proposed to work around the firewall/NAT traversal issues that limit the SIP communication.

SIP-Capable Firewalls, SBCs and Edge Devices

SIP ALG-Based SIP-Capable Firewalls

The majority of all SIP-capable firewalls today use the SIP Application Level Gateway (ALG) architecture, which solves firewall traversal by "taking care of the SIP packets on-the-fly," making sure that they reach the right destination on the LAN. It does not provide the full protection and flexible functionality necessary for today's secure enterprise.

SIP Proxy-Based SIP-Capable Firewalls or SIP-Enabling Edge Devices

The SIP proxy is designed to briefly stop the signaling packets so that each one can be inspected before the header information is rewritten and the packets are delivered to the appropriate endpoints. This provides the enterprise with a complete, flexible, controlled implementation of SIP-based communications.

The SIP proxy can offer benefits not available with the ALG architecture:

- Far-end-NAT traversal (FENT) to support remote users;
- Encrypted SIP signaling (TLS) and media (SRTP);
- Authentication;
- Advanced filtering;
- Advanced routing and control features; and
- Intelligence to enable the firewall to act as a backup for a hosted or centralized IP-PBX.

SBCs at the Service Provider Edge

Session border controllers (SBCs) control a firewall from a distance. All SBCs share the same basic feature in that they create pinholes in the NAT/firewall through which SIP signaling and media can pass through. Typically this far-end NAT traversal solution is implemented by continuously sending dummy packets through the firewall to keep pinholes open for the media to cross, or by asking the client to re-register in short intervals to keep those ports available.

SBCs are popular with service providers for their far-end NAT traversal ability. SBCs leverage FENT by creating special VoIP networks within their IP network.

Traversing an enterprise firewall with a tight security policy is a far different challenge. FENT solutions remove control from the firewall, and require the firewall to be accessed from the inside. Also, security-conscious enterprises will not typically allow open ports. The SBC's centralized approach also provides a single point of failure.

Customer Premise SBC

Many enterprise customers are reluctant to replace their existing firewalls with new SIP-capable firewalls because they've already set up security policies and trust the equipment that they have. Yet, many enterprises need to overcome the limitations of their existing firewalls whether they have firewalls with no SIP functionality or SIP ALG firewalls with limited SIP functionality.

This has triggered the development of a new type of product: the "customer premise Session Border Controller" or "SIP-enabling edge device. Essentially the SIP-enabling edge device assumes control of SIP traffic without involving the existing firewall in the process.

STUN, TURN, ICE

STUN (Simple Traversal of UDP through NATs) requires a STUN client on the phone or other endpoint device, which sends packets to a STUN server on the Internet. The STUN server replies with information about the IP address and ports from which the packets were received and detects the type of NAT device through which the packets were sent.

The STUN client at the endpoint uses this information in constructing headers so that external contacts can reach them without the need for any other device or technique.

STUN requires that the NAT device accept all traffic that is directed to a particular port, and forward that traffic to the client on the inside. This means that STUN only works with less secure NATs, so called, "full-cone NATs." This also means the internal client will be exposed to an attack from anyone who can capture the STUN traffic. STUN is not generally considered a viable solution for enterprises. Additionally, STUN cannot be used with symmetric NATs. This may be a drawback in many situations as most enterprise-grade firewalls are symmetric.

TURN (Traversal Using Relay NAT) allows an endpoint behind a firewall to receive SIP traffic on either TCP or UDP ports. This solves the problems of clients behind symmetric NATs which cannot rely on STUN to solve NAT traversal. TURN connects clients behind a NAT to a single peer to provide the same protection as that created by symmetric NATs and firewalls. The TURN server acts as a relay, any data received is forwarded. The client on the inside can then be on the receiving end, rather than the sending end, of a connection that is requested by the client on the inside.

This method is appropriate in some situations, but since it essentially allows inbound traffic through a firewall, controlled only by the client, it has limited applicability for enterprise environments. It also scales poorly since the media must go through the TURN server.

ICE (Interactive Connectivity Establishment) uses STUN, TURN and other methods to solve the NAT traversal issue. ICE allows endpoints to discover other peers and then establish a connection. ICE is a complex solution to the problem of NAT traversal, relies on client/server-based approaches, and removes control from the enterprise.

Universal Plug-and-Play (UPnP)

UPnP lets the Windows client control the firewall. The concept of allowing a software client to control the firewall is risky from a security standpoint. This technique is appropriate only for users with complete confidence in the LAN.

Conclusion

Although firewall/NAT traversal is still an impediment to widespread adoption of SIP, many companies today are looking for the right solution to bring the benefits of converged communications to their network.

Several solutions exist today, all of which have a place in the convergence toolbox. Some of the options are useful in situations where security demands are light and where there are no SIP servers on the LAN. But in those cases where security concerns are greater, and in situations where tight corporate firewall environments prevent the use of some of these simpler solutions, a more robust technique should be employed.

Whichever solution is chosen, the resulting advantages that come from adopting SIP-based real-time global communications offers productivity benefits to every company.

Olle Westerberg is Chief Executive Officer at Ingate Systems. For more information, please visit the company online at http://www.ingate.com. (news - alert)

Leveraging SIP to Support New Services

By Al Brisard

IP networking has become pervasive, as integral a part of our lives as the way we access data. But we've only just begun to explore and exploit the possibilities IP networking can offer in terms of how we connect to others. The IP network provides the ideal conduit for all forms of communication from simple Web browsing and e-mail to instant messaging (IM), voice calls, and video services, all packaged for easy consumption.

But the IP network isn't perfect and it's by no means homogenous in terms of the technologies used to provide the connections that enable voice, video and data services. But that is changing as SIP becomes part of the IP network's fabric acting as the unification force. SIP is the unique enabling protocol that will foster that exploitation to provide significant value in the form of real-time communications value. As the basis for real-time communications, SIP provides the logical bridge between enterprise and residential, wireline, wireless and WiFi networks, setting the stage for seamless mobility between devices, networks, and providers.

With regard to new services, SIP by itself is still only one hand clapping. The other hand is presence. The combination of SIP and uniform presence enables a unified virtual network that is presence aware. This catalyst will foster a new wave of development, bringing new and innovative types of services and features to market that can be leveraged across online personalities; at home, work, and play. As a major benefit, existing services and capabilities can now be extended to all devices wherever they are, via any network aware device depending on user policies. Additional benefits are realized via new services and capabilities that can be achieved such as real-time collaboration, file transfer, alarming, IM, video escalation, classes of service, chat, "Bots," intelligent routing, and many more in the works. In order to understand the new, it is important to understand where we are today as a baseline and to identify any limitations or obstacles that still must be overcome.

Serving SIP Today

SIP momentum continues to build as it moves beyond niche applications to more widespread, mainstream applications while proprietary applications become increasingly marginalized. On the consumer side it has manifested itself in the form of Vonage-like services as well as part of softphone-based peer-to-peer networks. On the enterprise side, where it is achieving the greatest traction, it has manifested itself as a PBX replacement in the form of SIP call control software running on a standard server and registering several SIP compliant endpoints (hard phones, softphones, media gateways, and so on). Additionally, many enterprise applications are integrating SIP to not only make real time communications within the applications possible but also to establish a standards-based protocol to be used to interact with other applications and devices. In the wireline carrier network it has manifested itself mainly as IP Centrex terminating SIP endpoints and most recently as support for SIP trunking. For wireless carriers, SIP is playing an increased role in the development of wireless handsets that support dual mode (cellular and WiFi). In each case above, SIP technologies support a specific market need and offer additional choices to the proprietary, single vendor solutions that once dominated the market. Across all the markets and where native SIP was not available, traditional networks and devices must be SIP normalized by leveraging gateway technology. In the short term, this is necessary to “SIP enable” the connection or device but as we continue to move to native SIP as a matter of course this will no longer be necessary, resulting in yet more opportunity as well as cost savings.

An overall accelerant to SIP-based real time communications solutions is the fact that SIP, delivered like an IT application, is implemented and deployed as a Web service as part of a SOA (services oriented architecture). This is true whether it is a traditional SOA that is enterprise-based or a carrier-based SOA, which is basically an IMS (IP Multimedia Subsystem) framework. Well architected SOAs have an HTTP interoperable nature and support clearly defined interfaces such as SOAP (simple object application protocol) and XML (extensible markup language). SIP-based real time communication solutions also supporting these interfaces seamlessly plug in and immediately provide service options and value.

Serving SIP Tomorrow

The current state of deployed SIP from above can be summarized as the existence of pockets of SIP technology and corresponding presence functionality that provide specific value. The number of pockets will continue to grow over time and with critical mass provide the foundation the next phase of development. The next step is to interconnect these pockets into one virtual group with completely shared presence information. Imagine the benefit to being able to understand the presence of your co-workers, key suppliers, customers, relatives, teachers, etc., regardless of what device or network you were utilizing and regardless of what device or network they were utilizing. Also imagine that not only did you understand their presence but you also understood their preferred mode of real time communications (IM, Voice, Video). This would be extremely productive just for this capability. But wait… what if you could also leverage common applications such as collaboration, conferencing, call recording and others? This would be even more productive and begin to create an inherent virtual community with common resources that you actively participate in and is one that is tuned to your persona of the moment (i.e., business, personal, etc.).

In this world, standards-based peer-to-peer communications is the norm — not the exception. Proprietary peer-to-peer models and pseudo peer-to-peer models such as Skype are interesting but in the end do not expand beyond their initial intention and remain closed. There is no question that Skype’s success has caused the market to stand up and take notice and has subsequently proved to be a valuable market accelerator. It also made the market understand how a closed proprietary model is a limited approach at best.

Getting There from Here

Many things still need to happen to make what is possible, probable. And many of these elements, as you may or may not expect are more political than technical:

Standards: The SIP standard has been complete for some time now with only minor evolutions being discussed. SIP as the primary supporting protocol for standards based instant messaging is still unresolved and being debated. In the IETF, XMPP and SIMPLE are the two standards contending with proprietary solutions out in the market such as AOL’s version of IM. Solidifying the direction for SIP-based IM and widespread adoption/implementation of these standards is critical to realize a plug and play tomorrow.

Fixed Mobile Convergence: In order to leverage the mobile network, mobile carriers need to be willing to allow dual mode devices that leverage the best connectivity option based on signal strength as well as feature set. SIP makes it possible for two different networks, in this case IP and mobile, to transition a current session or call from one to the other near seamlessly. Technically, fixed mobile convergence is possible today but a lack of clarity around business model, revenue impact, and long-term customer retention has carriers stalling to ponder the situation.

Presence Sharing and Control: For tomorrow’s capability to happen, users must be able to easily construct presence policies around their actual presence. In other words, a user may want to have different groupings of contacts that would be presented with different details regarding their presence. For example, a spouse may be presented with presence of always available and a supplier or casual business partner may be presented available from 8am to 9am. Communicating this presence real time across potentially many networks and pockets will be necessary to achieve presence ubiquity.

Security: Becoming more open and connected lends itself to more threats and vulnerabilities. In tomorrow’s world, SIP signaling and the associated media path will be secure. In fact security may be different depending on presence, class of service, connected network, or maybe even on-demand.

SIP Peering: Peering between carriers covers the frontier perspective and makes end to end SIP possible. Eliminating the need for SIP calls to traverse the PSTN does away with all the transcoding and associated inefficiencies. It basically settles the last frontier and finishes off the transition from many pockets to a single virtual SIP based community.

Timing

The most exciting thing is that this is all this is happening at a rapid pace and the key forces including innovation competition and time to market pressures are accelerating the momentum. This means that news services and features as a result of SIP adoption are right around the corner.

Al Brisard is vice president of marketing at Pingtel. For more information, please visit the company online at http://www.pingtel.com. (news - alert)

FEATURE ARTICLES

Advanced SIP Services: Developing Next Generation Applications

By Doug Tucker

The Telecommunications Service Provider market is rapidly moving to embrace SIP (session initiation protocol) as the next-generation "dial tone." This move is necessary to support the change in revenue focus from lowering operations costs to expanding revenue through offering new and innovative services.

For several years the telecommunications industry has focused on forcing the SIP IP infrastructure to emulate the legacy TDM-based, audio centric, telecom service operation. The goal of everyone involved in this effort (standards bodies, equipment manufacturers, etc.) was to achieve a level of parity between SIP service networks and the legacy TDM service networks. This has been achieved and, as a result, the industry is now looking beyond simple emulation of legacy voice services and is starting to redefine the service provider application infrastructure to accommodate new services that include rich multimedia content, as well as unified communications.

As service providers design their new infrastructure, the constraints that have been built around SIP to help it conform to the legacy service model are being broken down. The result is that SIP is finally being recognized and allowed to fulfill the functionality it was originally designed for: to enable multi-modal sessions to be opened between subscribers and services in a dynamic and flexible framework.

The service providers' focus on building out their SIP networks has turned increasingly toward forming an Application Layer on top of SIP that can provide the type of flexible multi-modal applications that can take advantage of their SIP infrastructure. The IMS and 3GPP efforts are a good example of the types of architecture being considered as service providers re-architect their infrastructures.

IMS defines a SIP Application Server as its open service creation environment. While several vendors have taken license with the definition of what a SIP Application Server is, the official definition of it is covered under JSR116, the SIP Servlet Container. This SIP Application Server provides an environment that closely emulates the Web Servlet container used commonly in data centric Web applications. The SIP Servlet Container provides easy and performance-effective access to the underlying SIP service network. Real-time, performance grade telephony applications can be written and deployed in this environment using common Web application development techniques. Building the application layer in this way provides easy access to a large population of skilled Web application developers.

Opening up SIP application development to Web developers through JSR116 is just the initial step however. IMS and JSR 116 together provide only the starting point for developing the rich multi-modal applications that will assist Service Providers in expanding their revenue targets. The JSR116 standard is great for creating applications such as Conferencing, IVR and announcements, ACD operations, video conferencing, etc., but only the portions of that application that involve call control and routing, session establishment, media control, and message routing. Other elements of a complete application such as Web portals, complex data manipulation, user presentation, calendar management, etc., all of which involve extensive data management and non-real time interaction with external elements (subscribers), can severely impact the real-time performance of a Java-based system. Because of this, telephony applications that require these non-real-time operations will be written in a way that makes use of the JSR 116 container for the telephony or real-time aspects, and a non-real-time platform such as J2EE or .NET will be used for the data-centric operations.

The combination of a J2EE or .NET layer on top of the SIP Servlet container provides the best of all worlds for telephony service creation. The only limitation to this combination is the lack of a clear demarcation point when architecting applications. It is too tempting to build most of the application in the Web tier to simplify the developer's task. This results in poor performance and scalability in that the performance-sensitive aspects of the application are compromised. The solution to this problem is the recent adoption of Service Oriented Architecture (SOA) techniques and the Web Services standards by service providers.

Web Services provides a set of standards that support the distribution of components of an application in a standard framework. Because it is based around standards, there are many vendors that support this infrastructure and provide tools that help develop, organize, debug, and deploy these applications. In the Web Services architecture, the SIP telephony portion of an application is exposed as a service. This SIP Telephony Service is then blended with other services at the Web tier to create a unified, logical distributed application (Figure 1).

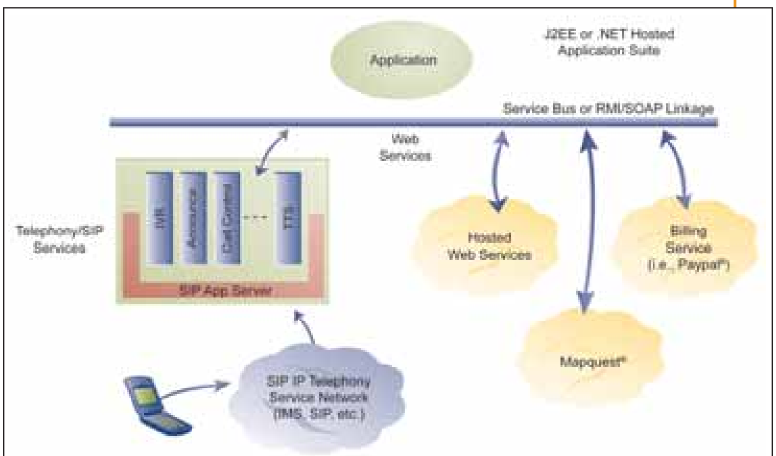


Figure 1. The diagram illustrates a deployment architecture that integrates a SIP Application Server into a Web Services distributed application.

The JSR 116 standard alone does not specify how to build a Web Services interface within a SIP Servlet container. This is a relatively new requirement that several companies are starting to address in their product lines. The standardization of a Web Services development environment on top of a SIP Servlet container will delay early implementations. This is a necessary step towards evolving the SIP telephony infrastructure toward a truly open and flexible service delivery environment. I would expect the standardization of Web Services on top of a SIP Servlet container to be well under way before the end of the year.

An example of the type of application that is possible in this infrastructure is a Driving Direction service (Figure 2).

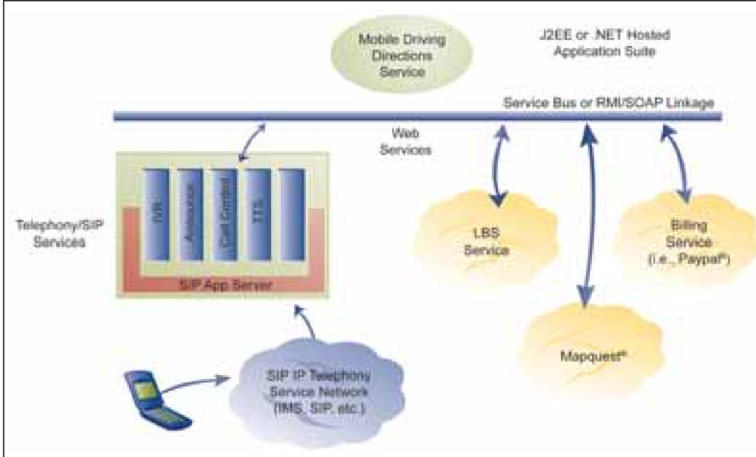


Figure 2. The diagram illustrates a distributed Web Services-based deployment architecture for this type of application.

In this service, a subscriber calls the service and is directed to an auto-attendant that is implemented within the SIP Telephony Service. The auto-attendant interacts with the subscriber through voice prompts and collects DTMF digits. It may also connect the subscriber to a Speech Recognition server to record the subscriber's desired destination. The SIP Telephony Service then would pass the subscriber's location and desired destination up to the Web tier through a Web Services interface, where the higher layer application would interact with a public Web Service such as MapQuest to generate the requested driving directions. The Web tier application would then pass the driving direction list back to the SIP Telephony Service where it will use a text to speech server to play the directions out to the subscriber.

Applications are becoming more complex and rich in terms of the number of available options and operations. Service providers are moving rapidly to adopt new standards and architectures to better enable them to capitalize on new revenue potential. The SIP IP network that has been built within these service providers is now being awakened to its full potential and is poised to deliver on the true promise of flexible converged infrastructure.

Doug Tucker is Chief Technology Officer of Ubiquity Software. For more information, please visit the company online http://www.ubiquitysoftware.com.(news - alert)

Q & A

60 Seconds with Professor Henning Schulzrinne

Henning Schulzrinne was a member of technical staff at AT&T Bell Laboratories, Murray Hill and an associate department head at GMD-Fokus (Berlin) before joining the Computer Science and Electrical Engineering departments at Columbia University, New York. His research interests encompass real-time, multimedia network services in the Internet and modeling and performance evaluation. Protocols co-developed by him — including SIP — are now Internet standards, used by almost all Internet telephony and multimedia applications. His research interests include Internet multimedia systems, quality of service, and performance evaluation.

I had the chance to ask Professor Schulzrinne a few questions on the matter of SIP's role in enterprise communications.

GG: Recent news announcements from some of the leading companies in the Enterprise IP Communications space have all been focused on support for SIP. Cisco, Avaya, Sphere, IBM, 3Com, and Microsoft have all been making headlines with their SIP initiatives.

Why the recent focus on SIP for enterprise applications?

HS: I suspect that there are probably several reasons, such as enterprise managers having been burned before by proprietary systems. However, in this particular case, one reason is that many vendors seem to specialize into each of the core components of an enterprise IP communications systems, such as Ethernet phones, proxy servers, PSTN gateways and border devices, such as VoIP-enabled firewalls. Even many companies that offer products across this spectrum may not offer the precise equipment that a particular company needs, such as conference room-quality speaker phones or 802.11 phones. For example, we recently rolled out a SIP-based system for the Department of Computer Science at Columbia, which includes two brands of desk phones, a software infrastructure from one major vendor, a soft phone made by a small company, and ATA devices for fax machines made by yet another specialized vendor.

GG: What benefits does SIP offer the enterprise?

HS: There are probably two core benefits, namely having a standards-based technology and using a mature technology that has been widely tested.

A standards-based technology makes it far easier to evolve a system, without having to do forklift upgrades every few years. In the past, it has been fairly common that upgrades were postponed as long as possible as such an upgrade would be extremely disruptive and might have involved replacing hundreds or thousands of hardware devices, along with the central infrastructure and maybe even the wiring. Now, it should be possible to slowly evolve the infrastructure as needs evolve and functions get added. For example, as managers get their late-model IP phones with color screens, older phones can then migrate down to users having more modest needs (or that simply rank lower on the organizational totem pole).

Having a standards-based technology also makes scaling much easier, e.g., when two organizations are merging.

The technology itself is now fairly mature; unlike many proprietary systems, issues such as scaling and security have been addressed, even if they are not always implemented yet.

Longer term, currently only SIP offers an integrated vision of communications that includes not only voice and video, but also instant messaging, presence, and advanced services.

GG: What's your take on SIP Trunking?

HS: SIP trunking avoids the need to run PSTN gateways locally to make any outside calls, so this is clearly a step in the right direction of an all-IP, end-to-end SIP infrastructure. Ideally, SIP trunking should not require any new protocol mechanisms or infrastructure, just the inbound and outbound delivery of calls destined for calls to the outside. Longer term, the combination of ENUM or URL-based routing will reduce the need for the explicit notion of trunks. After all, we don't use e-mail trunking to send messages from one company to another.

The market for SIP trunking services seems to be fairly young, particularly for smaller to mid-size enterprises, as service providers appear to be more interested in selling IP Centrex hosted services rather than just gateway services. Besides the possibly lower per-minute costs, the ability to have more burst capacity seems very attractive. For example, in my department, we have a PRI, for about 200 users, which is usually more than enough capacity. However, we'd like to be able to host larger teleconferences, without jeopardizing the ability of others in the department to make and receive calls. We have plenty of IP bandwidth, hundreds of Mb/s for the campus, but cannot justify buying an additional PRI for the occasional use it would get.

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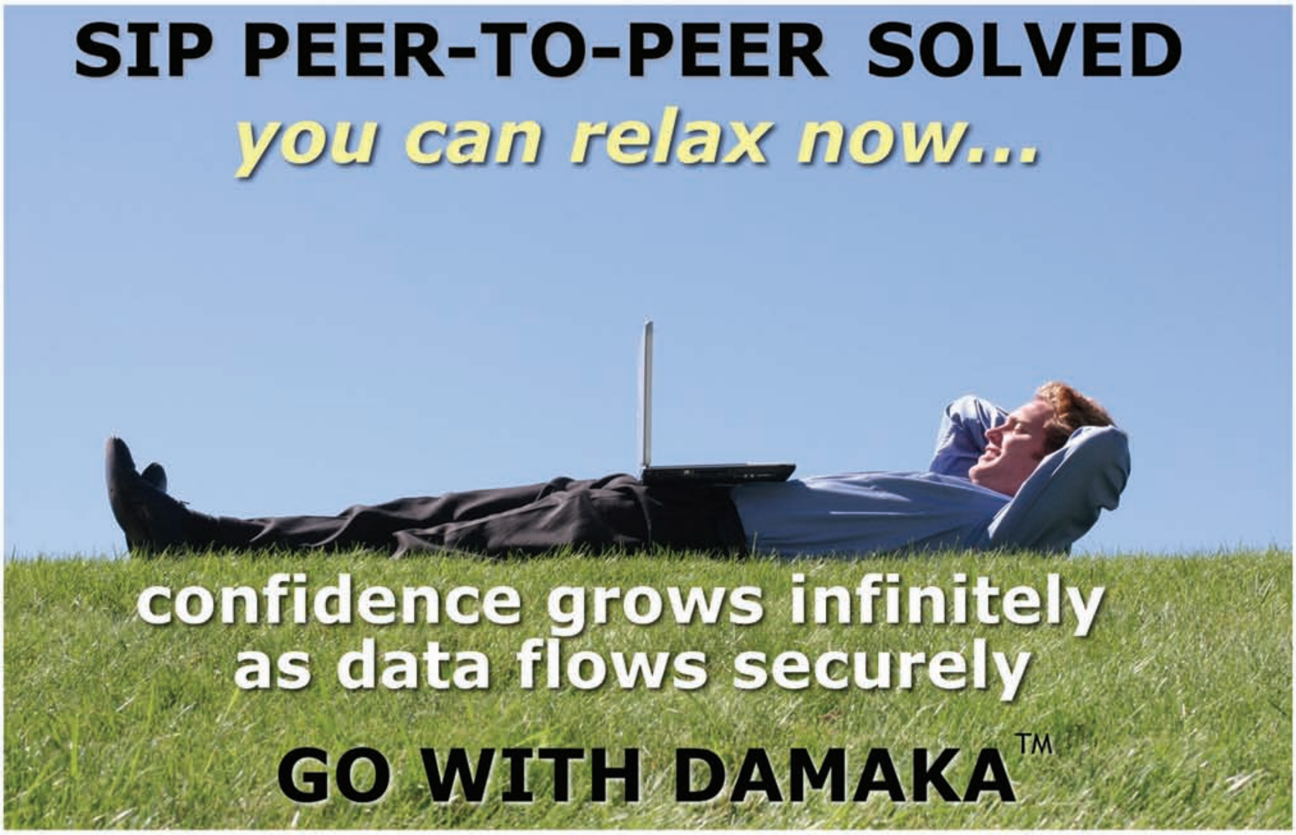
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