Inside The InterOperability Lab

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TMC SIP MAGAZINE™

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He Said, She Said?

EDITOR'S NOTE

Everything 2.0

by
Richard "Zippy" Grigonis

With all the verbiage over Web 2.0 generated in 2006 (and now into 2007), one of the few things one can pin down about it is that collaboration functions such as document and calendar sharing will be paramount and that these will be helped by the kind of sharing and presence-awareness made possible by SIP/SIMPLE support in IM platforms and business applications. (Note the article on SIMPLE in this issue of *SIP*.)

"2.0 mania" has continued with Voice 2.0, the idea that IM/presence and VoIP will converge — or is it that the web and IP Telephony will converge? In any case, you get the general idea. There's still a lot of mileage left in the term "convergence".

Alec Saunders, the visionary technology evangelist and blogger, even wrote a Voice 2.0 Manifesto back in October 2005. More recently, Saunders has written the following: "When the Voice 2.0 Manifesto was written, it identified presence as the enabler of conversation, allowing parties to easily determine each others willingness to engage, and by which technology. Presence, today, remains an unfulfilled promise despite the numbers of writers touting it as the future of communications for the better part of three decades now. . . The solution is user driven presence — the New Presence model. Presence geeks will notice the absence of discussion around standards like XMPP, SIP/SIMPLE, and IMS in this piece. My assumption is that we're going to get to a standard, and rather than debate the niki-norks of a particular protocol, this piece is about a vision for what presence might be when those details are completed. . . New Presence is a user-centric view of presence. Instead of merely reflecting the crude, device specific 'availability awareness' of today, New Presence systems understand our context, relationships, wants and desires. The New Presence model reflects the integrated conversation web we live in today."

Saunders' thoughts on the matter have kicked into high gear the usual army of bloggers. The general consensus is that we all have too many simultaneous examples of communications, IM and presence competing for our attention on a minute-by-minute basis (e.g. Skype, Google, MSN Messenger, Yahoo!, AOL, softphones), and that various devices and software should be aware of each other and of a master rich presence methodology. In other words, everything should be tied into one overarching standard and users should have one "identity" across all devices and platforms. That may take a while to come to fruition

In the meantime, the IM-to-Voice aspects of Voice 2.0 have struck a chord with the public and the technology has gained real momentum. Talkster has caused quite a stir as a service that lets you make free calls from your mobile phone to your buddies who use voice-enabled instant messaging services. No downloads or software installations are required to your phone or PC. The Talkster network (news - alert) relies on web services to handle both calling services and a dynamic contact list containing users' presence data ("status") utilized by instant messaging services. Talkster also offers international calling rates as low as 2 cents per minute, which many Europeans may find even more tantalizing than the IM-to-Voice link. One reason for this generous offer is that Talkster is still in beta — sign up now at http://www.talkster.com and start using the beta service today. The full "official" enterprise version of Talkster will be released in spring 2007 and will be a centrally-managed, policy-based, full-fledged voice and text communications solution.

Let's just hope Yours Truly doesn't get replaced with Editor 2.0.

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

SIP MAGAZINE™

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

EDITORIAL
Greg Galitzine, *Group Editorial Director* (ggalitzine@tmcnet.com)
Richard 'Zippy' Grigonis, *Executive Editor* (rgrigonis@tmcnet.com)
Erik Linask, *Associate Editor* (elinask@tmcnet.com)

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

ART
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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
Anne Powers, *Account Director - West/Southwest/Midwest U.S.; Europe* (apowers@tmcnet.com), ext. 167

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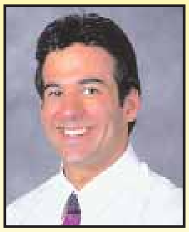


by

Rich Tehrani

The post holiday fog is leaving my brain and I am getting ready for another fantastic year in telecom. If I had to make predictions about the SIP market in the new year I would have to say there are numerous areas of growth.

Rather than give you my perspective exclusively, I contacted David Hattey, the President and CEO of FirstHand Technologies. As you may know, the company was formerly called SIPquest and has a product line fusing SIP and mobility.

I asked Dave for his perspective on SIP in 2007. He thinks '06 really set the groundwork for this year with Cisco announcing CallManager 5.0 with SIP support as well as every major vendor announcing SIP support. In 2007 he says 2 things are happening: First, SIP as a telephony protocol absolutely becomes prevalent. Every major company has announced support for it and it will permeate the space.

The second thing Dave sees happening is SIP being extended into mobility. He admits the comment is a bit self-serving as this is the space his company plays in — but this is going on with every major vendor, he says. The companies are extending SIP call control out into the mobility space as a data service over cellular packet, WiFi or WiMAX. He adds that several carriers and equipment vendors are looking at convergence of packet voice and packet cellular. He sees HSPDA and EVDO Rev. A as being drivers for this evolution of SIP in 2007.

In summary, Dave sees 2007 as the year that SIP permeates fixed telephony and mobile SIP will gain a foothold and emerges in the marketplace.

...2007 as the year that SIP permeates fixed telephony and mobile SIP will gain a foothold and emerges in the marketplace.

It is tough to disagree with Dave as my Publisher's Outlook (http://www.tmcnet.com/434.1) in the last issue of this magazine — November 2006 discussed the acquisition of Orative by Cisco which coincidentally will also help Cisco bring presence information to the mobile user.

FirstHand Technologies (news - alert) hopes to have a new announcement in time for Internet Telephony Conference & Expo (http://www.itexp.com) Jan 23-27, 2007 in Ft. Lauderdale, Florida. Going forward, expect to see them focus on compatibility with more call control products as well as integrating their mobility offerings to enterprise IP PBXs. They will also have deeper feature sets, allowing cell phones and dual mode devices to have more PBX-like functionality.

Expect them to announce some new vendor partnerships very soon.

Speaking of ITEXPO, I happened to be looking at the SIP conference sessions coming up at the show this month as a guide to what the opportunities in SIP will be. I figured this was a great starting point for such an article as the conference sessions generally set the tone for the future of the industry. So here they are in all their glory...

There will be a load of SIP education in the SIP track. Sessions will focus on everything you may be interested in learning from an introductory course to the ever-so-hot SIP trunking space. In addition there will be a session on deploying SIP on a global scale, p2p SIP and early media's effect on high volume applications.

For those who are interested there will be a full suite of IMS sessions at the collocated IMS Expo (http://www.imsexpo.com/). Sessions discussed in this conference will range from IMS billing to FMC and video.

In short, I am looking forward to an amazing 2007. I expect to see much innovation, more consolidation and a smattering of new "killer" must-have SIP applications permeating fixed, mobile telephony and the Internet.

Keep your eyes peeled on your electronic and real mailboxes for future issues of *SIP* magazine, now in its second year — thanks to you and our advertisers.

INDUSTRY NEWS

Solegy Launches Free, SIP-based Softphone

By Mae Kowalke

Hosted VoIP solutions provider Solegy (news - alert) announced the launch of its free softphone, designed to run on Microsoft Windows. The softphone is based entirely on open source code, which can be accessed through Solegy's OpenSourceSIP initiative.

Solegy says the SIP-based softphone is completely compatible with the company's hosted ServicePDQ platform. The softphone also can be customized to include branded skins, instant messaging, and advanced account management features (e.g. pre-paid balance, one-click credit purchasing).

Further flexibility is provided by the softphone's ability to be embedded in a Web page or distributed in a Windows or Mac OS X install file.

"We are making the Softphone available to everyone who wants to experience the ease of making VoIP calls through their computer," Solegy's CEO and co-founder, Eric Hernaez. said.

Hernaez continued: "This is just the tip of the iceberg in terms of what we can create for service providers who want to adapt the Softphone for their customers. The basic Softphone we are offering can be customized to not only create a unique look, but to take advantage of the many service management features available in our hosted environment."

Solegy explained that SIP, which it describes as a peer-to-peer processing protocol, "allows next-generation services including VoIP, video, instant messaging and online games to take place through session initiation, modification and termination."

http://www.solegy.com

damaka Launches Video Mail with Its Peer-to-Peer VoIP Software

Communication and collaboration software company damaka launched its video mail service, which enables users to leave a video (and audio) message for their peers on damaka (news - alert) and to any other contacts using a regular Webcam. Video mail is a new addition to damaka's existing set of features, which includes IPTV, DialOut, video conferencing, desktop sharing, and FastFile transfer.

damaka offers its white-label, SIP-based, peer-to-peer platform for companies interested in jumping into the triple play solution with minimal infrastructure investment. According to damaka CEO Siva Ravikumar, Videomail is just one example of how damaka can help a small and medium-sized operators or new entrants in the telco market offer services that the incumbent telcos have yet to provide. Why spend time selling a voice mail service when video mail is already there for the taking?

damaka recently released its IPTV product, which provides users free access to BBC News, CNN-IBN, HSN, and many other channels. The company also recently released a best-in-class peer-to-peer multi-party video conferencing (H.263/H.264) as the cornerstone of its video offering.

http://www.damaka.com

SIPBox Named Polycom Certified Reseller

By Cindy Waxer

Specializing in Asterisk, the industry's first open source telephony platform, is just one of the ways SIPBox (news - alert) is attempting to establish a stronghold in the open source market. The end-to-end telephony provider, which designs, implements, and supports VoIP solutions for medium to large enterprises, announced it has become a certified Polycom reseller.

Polycom (news - alert) is a provider of unified collaborative communications solutions and offers a complete line of SIP-based desktop phones that are fully interoperable with leading IP PBX platforms. SIPBox recommends the use of Polycom VoIP phones in all of its telephony deployments, specializing in Asterisk, the industry's first open source telephony platform.

"Aligning ourselves with a leading technology vendor in the industry such as Polycom allows us to continue to provide our customers with flexible and cost-effective VoIP solutions," said Chad Agate, co-founder and CEO of SIPBox. "We believe that Polycom's line of SIP phones will support the business telephony features of Asterisk and meet the growing needs of the market."

http://www.sipbox.net
http://www.polycom.com

WorldGate Communications to Debut Travel Version of the Ojo Video Phone

WorldGate Communications (news - alert) announced that it will debut a sample of a travel model for the Ojo video phone product line in January. Ojo is the first broadband vide-phone to feature a friendly and intuitive display that enables clear, realistic, face-to-face, personal communications experiences. Its features, sound and video quality are unmatched in other video phones; users enjoy full motion-smooth video with no jerkiness. In addition, Ojo customers are able to easily and simply connect their Ojo personal video phone to their broadband networks to experience true-to-life video communications.

The Ojo video phone is designed to conform to industry standard protocol (SIP) and utilizes enhancements to the latest technology for voice and video compression (H.264) to achieve superior quality at data rates as low as 100 Kbps. WorldGate has been awarded a patent for its distinctive design and has patents pending for its specialized technology. This unique combination of functional design, advanced technology and use of broadband networks provides real-time video communication experiences that bring families and friends closer together, and for the first time provides consumers with a high-quality, affordable video phone.

http://www.wgate.com

BEA Systems' WebLogic SIP Server Certified Interoperable with NEC, Huawei, and Convergin

By Patrick Barnard

San Jose, Calif.-based BEA Systems (news - alert) is continuing to play a key role in helping service providers all over the world quickly and efficiently roll out next generation services for their customers. The company has announced that it has completed interoperability testing between its WebLogic SIP Server - the industry's first converged Java EE-SIP-IMS application server - and the IMS service platforms of NEC, Huawei Technologies, and Convergin. (news - alert) The successful completion of this interoperability testing means these network equipment providers (NEPs) can now combine their solutions with those of BEA to deliver fixed mobile convergence (FMC) and telco SOA for carriers' pre-IMS and IMS networks.

"Operators will benefit from the industry's leading IMS application server platform, which provides a converged, single container architecture at their IMS Services Layer (versus multiple containers to achieve the same capabilities), which gives operators a lower TCO structure," Ken Lee, director of product marketing for BEA's WebLogic Communications Platform product family, told TMCnet. "BEA WebLogic SIP Server gives operators an edge with the most highly available platform, and a converged container with access to Java, Internet, SOA, IMS capabilities, which means more innovative, revenue-generating services."

BEA's partnerships with NEC (news - alert) and Huawei, (news - alert) in particular, will strengthen its presence in Asia. Lee pointed out that the company's penetration in the telecom OSS/BSS and IT side in the APAC region is already quite high.

http://www.bea.com *http://www.necunified.com*
http://www.huawei.com *http://www.convergin.com*

Microsoft Unveils VoIP Solution as Part of Desktop Communications

Microsoft Corporation (quote - news - alert) has opened a private beta of its new enterprise voice communications server, Microsoft Office Communications Server 2007, to 2,500 IT professionals. Office Communications Server 2007 allows companies to integrate voice over Internet protocol (VoIP) technology into existing telephony infrastructure, eliminating the need for expensive network overhauls and also extending the useful life of existing investments. The new voice server will also allow workers to instantly launch a phone call from 2007 Microsoft Office applications, such as Office Word 2007, Office Outlook 2007 or Office Communicator, by simply clicking on a colleague's name to determine his or her availability and initiate a person-to-person or multiparty call.

With native support for Session Initiation Protocol (SIP), Communications Server 2007 and Microsoft Office Communicator, part of the 2007 Microsoft Office system, interoperate with products from industry partners including Nortel Networks, Alcatel-Lucent, Avaya, Cisco, LG-Nortel, Mitel Networks, NEC Philips Unified Solutions, Polycom, and Siemens Communications. Through these relationships, customers worldwide will be able to support VoIP using their existing desktop phones, data networks and time division multiplexing (TDM) or Internet protocol (IP) private branch exchanges (PBXs). Customers will also able to leverage the softphone capabilities of Office Communicator to make and receive phone calls from their PCs, eliminating the need to purchase expensive IP-compatible phones.

http://www.microsoft.com

Zoom Technologies Lets Users Change their VoIP Service

By Patrick Barnard

VoIP (define - news - alert) equipment maker Zoom Technologies (news - alert) has begun shipping the Model 5800 Zoom VoIP Freedom, a nifty little analog telephone adapter (ATA) that lets a user choose which VoIP service they want to use based on their needs.

As the company points out, most VoIP providers tend to hold their customers hostage by furnishing hardware that only works with their service. Zoom's new VoIP Freedom, though, lets them choose from up to 25 SIP-based providers using a web-based chooser. Using the chooser (which allows for very quick and easy change in service provider) a user can choose VoIP services from providers in the Americas, Europe, and Asia. The chooser's menus are available in several languages including English, Spanish, German and Vietnamese. Once a user selects a service, the Zoom VoIP Freedom device automatically configures itself for that service. Best of all, there is no charge for registration - and the device itself costs only $59.

One advantage of the device is that it supports both VoIP and traditional phone service, and allows the user to use one or the other - or use both simultaneously. For example, someone can talk on the VoIP line but someone else can still make calls on the traditional phone line at the same time - plus it is easy to switch between the two. Thus the device enables a "second phone line" capability that isn't always found on other ATAs. Most VoIP customers (especially new ones) tend to keep their traditional phone service so that they continue to have access to 911 services. In the event of a power outage, the device will automatically connect to traditional landline phone service.

Another huge advantage of the device is that it lets cell phone users use the device as a bridge to connect their cell phone to their VoIP line, thus letting them make long distance calls using their cell phone via low cost VoIP.

http://www.zoom.com

AccessLine Migrates to SIP Network

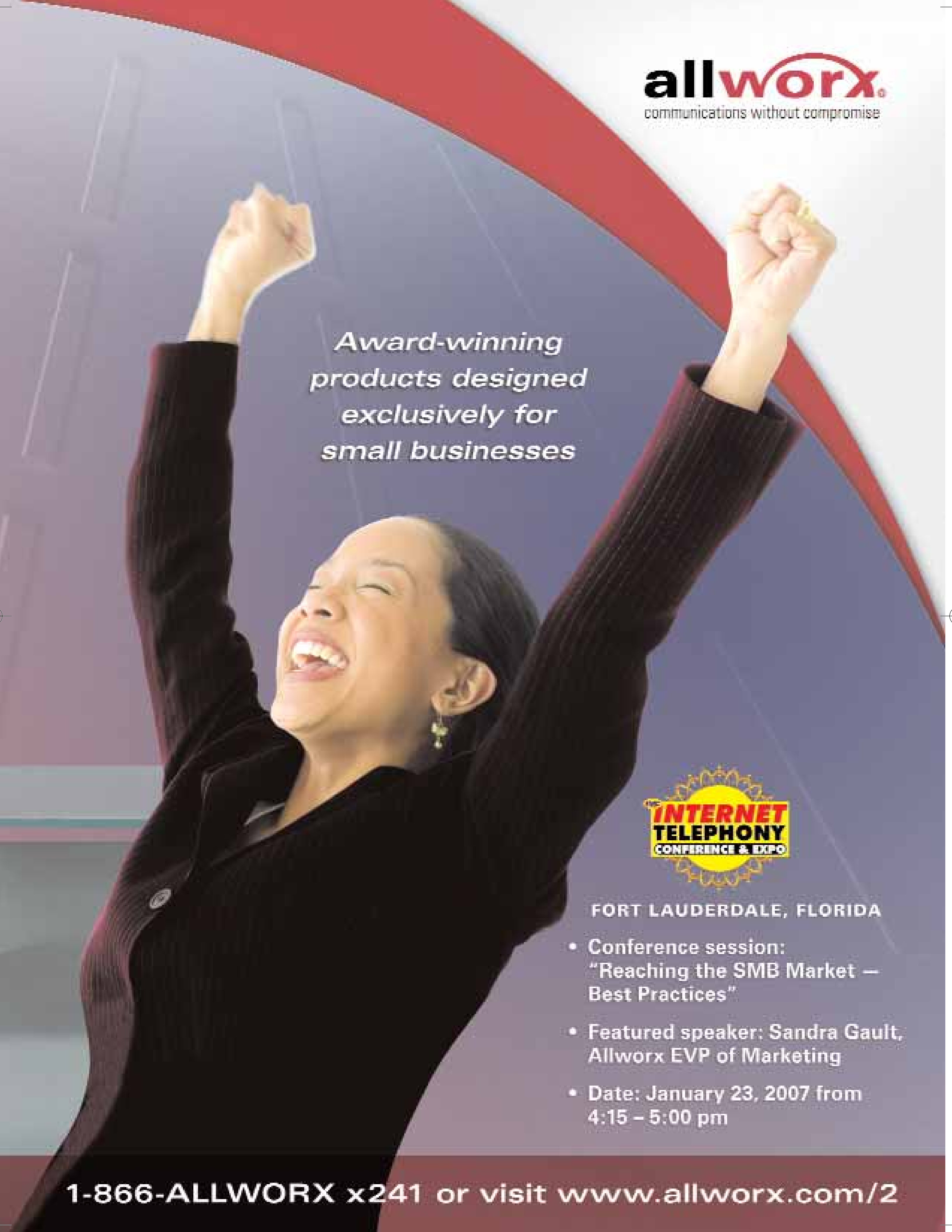
By Cindy Waxer

Only a month after being awarded two U.S. patents for VoIP call processing and security, AccessLine Communications, (news - alert) a provider of hosted VoIP services, has announced the completion of a multi-year migration of its telecommunications network to the Session Initiation Protocol (SIP) standard. Through leveraging its SBC technology and internal signaling, AccessLine can normalize complex SIP interoperation nuances and offer unified access to multiple VoIP domains via a single interconnection with AccessLine.

"Each network service provider presents a different combination and permutation of the myriad options and capabilities that can be supported over a SIP interface," said Jerry Knight, chief technology officer for AccessLine. "AccessLine's customers can reap all the benefits in reach, availability, features and cost of multiple carrier interconnections via a single interconnection with AccessLine. This can save the customer years in time to market and technical effort that can be re-deployed on other more valuable initiatives. This is of particular importance to customers with large call volumes, where redundancy and availability are primary concerns."

In addition to the unique "normalization function" that is inherent to AccessLine's SIP Network, AccessLine is able to deploy a much larger number of SBCs (Session Border Controllers) at a much lower cost than most other carriers. This simultaneously gives AccessLine both a cost and quality advantage. For customers of AccessLine's hosted SmartVoice Service for business, this means that the benefits of advanced business telephone service are delivered at a fraction of the cost of competing services. AccessLine's investment in its network migration to SIP means increased call quality, high availability and improved traffic efficiency.

http://www.accessline.com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SPEAKING SIP

State of Emergency: VoIP and 911

Voice over IP is a technology that has had to jump numerous technical hurdles on its way towards success. Voice quality, ease of use, and NAT traversal have been three of the biggest so far. The next one, which VoIP is in the midst of handling, is emergency services.

by
JD Rosenberg

VoIP (define - news - alert) has the ability to far exceed existing wireline and wireless emergency services in terms of quality and features. With VoIP, wideband speech can be utilized, providing extremely high quality sound. This allows emergency call takers to hear background noises, which can provide additional information about what is happening where the caller is calling from — a good thing for a trained ear to hear. It can also be beneficial for separating fraudulent calls from real ones. Video can allow for fire and police to see what's going on at the site of the emergency before they even get there. Text-based media streams allow the hearing impaired to have improved access to emergency services. IP-based networks supporting VoIP will also allow emergency calls to carry additional information about the caller — links to their medical records, for example, to allow responders get the right help to the scene. VoIP will thus be an enabler that allows us to save more lives.

But that's the future. Today's state of emergency services is only part of the way towards this vision of the enhancements that IP-based networks could provide. Many VoIP providers do offer emergency services, but it is not yet ubiquitous. Fixed access is provided, but mobility is not yet supported. If the user does move it around, they have to manually update their location with their service provider. Though calls can be routed to the correct emergency service center, in many cases the caller's location is not provided to the operator. Though connectivity exists to most emergency service centers, in some cases it is done through the "backdoor" — the administrative lines that are meant to be used for non-emergency connectivity.

What stands in the way of removing these limitations and going beyond what can be done today? There are three factors in play:

1. Properly routing calls, and then connecting them to the right emergency services center, requires access to databases and closed access points in the PSTN that have not always been available to the VoIP provider.

2. Emergency services require the geographic location of the user (caller) to be determined, a capability which is virtually non-existent in today's Internet.

3. The Internet allows the separation of access from service, so that one provider can offer the VoIP service, while another provides IP connectivity.

The first issue can be remedied in part by defining new IP-based access to emergency service centers, and providing standardized IP-based access to the databases needed to route calls to those service centers. Fortunately, work is underway to achieve these goals in several standards bodies, including the IETF. The IETF has defined a new protocol, called LoST (Location to Service Translation), which provides a simple XML-based interface that can be used to map a callers location (included in the call setup message) to a SIP URI for reaching the emergency service center for the caller's location. The IETF has also defined standards for representing location information in SIP messages.

The second issue will require both devices and the network to be upgraded to be "location aware". The technologies that will play a role in that will be highly variable, including GPS for certain devices, link layer technologies such as LLDP-MED for other devices, and new DHCP options for other types of devices. Unfortunately, providing location isn't just a technical challenge, it's a business one, and that challenge is intimately coupled with the third big issue.

Perhaps the greatest strength of the Internet is that allows a complete decoupling of the underlying network from the applications that run on top of it. Consequently, the user can access an IP-based application from anywhere that IP connectivity exists. This means that, for a given phone call, there are really two providers in play — the one that provides the access network, and the one that provides the application service. Unfortunately, proper operation of VoIP requires participation from both of these providers. The provider of the access network is the only one that knows the location of the user, but the VoIP application provider is the one that needs to route the VoIP call based on that location information. For the system to work, the access provider must allow the VoIP provider to obtain and use this location information.

Why is this unfortunate? It is unfortunate because, in many cases, there is little business incentive for an access provider to give this location information to the VoIP provider for the purposes of completing the emergency call. Clearly, for privacy reasons, users wouldn't want this information to be given out arbitrarily without their consent. However, the user will need the information to be provided for the emergency call to complete. If the access provider also has a VoIP service, they would be helping a potential competitor offer better service. If the access provider doesn't offer VoIP service, it would require them to deploy infrastructure for applications they aren't providing. It is possible that VoIP application providers can work out business arrangements to compensate access providers for location information. If that cannot come to pass, government regulation might be required to help move things along. To be successful, those regulations must recognize the separation of roles and responsibilities between the access and application providers.

But, fear not! Just as VoIP has handily dealt with numerous challenging hurdles in the past, this one too will be met and overcome. Once it does, VoIP will not only change the way people communicate, it will save lives in the process.

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. (quote - news - alert)

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PRESENCE ENABLED

Present and Accounted For

by Joe Hildebrand

There is a transition underway that is revolutionizing the way people and applications interact and it is summed up in a word: PRESENCE.

No longer merely an indicator of availability, PRESENCE is a universal router of messages and media streams and universal interface to live information.

In our last column we asserted that due to several factors, the wrapping of context around the content we consume now delivers more value than the content itself. Person-to-person presence, as represented in most instant messaging systems, today provides adequate context around some interpersonal communications, i.e. it certainly helps me limit the number of interruptions I receive — when I'm really busy — to only the most important ones. Similarly, by presenting me with the availability of my contacts, it helps me negotiate phone calls.

For most people-centric communications, this is a manual process that ignores the presence of applications and the rich context they have to add. Peering into the future, we look no further than the sophisticated environments we currently find on Wall Street and in the federal government where applications and people collectively use presence to dynamically route messages between people. For instance, bots and agents roam chat rooms looking for new information posted by and for people. These bots and agents feed off keywords, policy, and the identity information stored in directories to determine which people need to see what information immediately, or conversely, what information needs to be kept from which people.

We see related — albeit less sophisticated — applications in consumer IM services too. One example I'm familiar with is the Swiss Phone book tel.search.ch@swissjabber.ch bot available on the XMPP network. This bot, which appears on my contact roster, allows me to do natural language queries of the Swiss phone directory. I have seen similar bots for movie times, World Cup scores, and other useful functions, but if you've ever used any of them you've already guessed my next point.

Humans and machines don't speak the same language. We can sometimes understand each other but at least as frequently simply frustrate each other. The fact is that applications require message structure that is not native to human language. Given the right tools, people can readily speak the language of applications today. Those tools include forms and other constructs which allow for crisper semantics to be embedded within messages and media streams. Adding structure to communications allows programmers and ultimately machines to know more about how to handle and deliver them in the right context.

Consider first responders who file incident reports from the scene of an emergency. Instead of providing free-form textual information (which is not machine-friendly) or requiring them to adhere to a machine-readable format, a template for an incident report appears on their screens as a basic form. It asks quick questions that probe for context and directs the answers into a format that can be understood and ultimately best routed by a machine. The result is the ability for information to find recipients based on their presence, which includes their organizational role and identity, and not necessarily based on their relationship with the sender. For example, an officer on the scene of a spill doesn't need to know anything about a chemical expert, nor need a dispatcher to track one down. Instead, based on the information in the form, a presence-based router can scan directory information and dynamically reach out to and involve experts in developing situations far faster and more accurately than a person could. We also see this in call centers, where customer information is presented in a dynamic form that can be pushed easily between support staff, automatically alerting supervisors to escalating situations based on the information in the form.

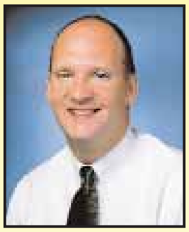
This is the next step in the transition, forcing more structured data and better semantics into person-to-person and person-to-application messaging. The final step is to enable presence-based application-to-application communications. For instance, a commercial real-time traffic application could have a software agent resident in the first-responder's network. The agent could subscribe to incident reports as they are published, syndicating only that information which is for public consumption (i.e., a lane closure), while omitting more sensitive information, such as the names of those involved in a crash. Embedded within the agent's presence profile is business logic that accounts for policy and enforces its compliance. This type of cross-application messaging can be deployed today.

There remain challenges to achieving this future. People and organizations must use forms more consistently and effectively and forms must become a more inviting means of collaboration. There is also much to do in terms of embedding directory information and policy with the syntax that can be easily read by people and machines alike. The good news is that much of the infrastructure we've already built for person-to-person presence can be rapidly recycled in this transition. How much and how soon is the subject for another column but I'll give you a hint. It depends upon how adequately today's architecture accounts for the needs of people and applications to communicate with one another in real-time. How you balance those needs will likely define your strategy as live or dead.

Joe Hildebrand is CTO of Jabber, Inc. (news - alert) For more information, please visit the company online at http://www.jabber.com.

SPECIAL FOCUS

SIP in America's Heartland — NNU Deploys the First Ever Inter-Tel 7000



by
Greg Galitzine

Northwest Nazarene University (NNU) is a Christian liberal arts university fully committed to an educational process that pursues both intellectual and spiritual development. Founded in 1913 as a college, and having attained university status within the last decade, NNU today serves over 1,200 undergraduate students and approximately 500 graduate students at its Nampa, Idaho campus. There are also approximately 260 employees. The University comprises eight regional campuses in the United States, and over 52 such centers globally. Within the United States, NNU has campuses in 7 States: Alaska, Oregon, Washington, Montana, Idaho, Utah, and Colorado.

NNU is also home to about 20–30 International students who have come to the Nampa campus from around the world.

I had the good fortune to meet with Eric Kellerer, Director of Information Services at Northwest Nazarene University to discuss some of the challenges facing this growing educational community, naturally focusing on the telecommunications needs of the school.

Northwest Nazarene happens to be the first organization to have deployed Inter-Tel's newest solution, the Inter-Tel 7000, together with Inter-Tel's 8660 phones.

In an interview with TMC earlier this year, Inter-Tel CEO Craig Rauchle described his company's latest offering as a ". . .platform that will interoperate with other standards-based devices and applications. Over the next few years, this will provide businesses with enormous flexibility in their ability to integrate third-party solutions into their network."

The Inter-Tel 7000 can scale up to 2,500 users per site and is designed as a pure, standards-based communications platform. Using SIP at its core, the system is designed to be redundant and secure and offers an easy-to-use interface for remote management and configuration.

The new platform also offers full PBX-style features based on Inter-Tel's decades-old legacy in telecom, as well as a number of enhanced features like embedded presence management and mobility.

I asked Kellerer about NNU's needs, and how they came to consider a new phone system for the University and why they considered Inter-Tel.

"We had a [system from a competing manufacturer] since about 1994, and I think our motivation for looking around was two-fold; for one, our voice mail was just in poor shape, and we were having problems with voice mail delivery for sometimes as much as 24 hours," Kellerer told me.

"The biggest motivator was customer care," says Kellerer. "We wanted to be able to care for individuals. That's the focus of this institution: We're very personal and very care oriented about people. We weren't handling calls well. Calls would get routed and dropped, and we just needed to be more intentional about getting calls and routing them correctly and making sure they got to the right spots."

NNU serves a number of different constituents including prospective students; current students; alumni; donors; and vendors within the community.

Kellerer recounted the tale of a donor who experienced some difficulty in making a sizable donation to NNU. "One of the things that escalated our concern was a donor who decided to surprise us one day and showed up at the airport to come to campus. He had with him a $200,000 check, and he got to the airport and called our development office and he got sent straight to voicemail, and that voicemail never got delivered for a whole day. So you can see we had some motivation to fix that problem right away."

The good news is the donor eventually made his way to campus. The better news is that while he was a bit upset, he still delivered the check.

Expanding on his needs, Kellerer continued, "From my department's perspective, we have a support desk that needs to get calls, and we need to respond, and we wanted to handle those calls better in more of a call center type of approach, and our existing system didn't handle that well. As the administrator, I also needed something that was easier to maintain, and take care of."

Scott Langdon and Rod Richardson, president and vice president of DataTel Communications, were present as well when I visited Nampa. DataTel is the Meridian, ID-based dealer who installed the system for NNU. They have been an exclusive Inter-Tel dealer for the past six years or so, having ". . .picked up Inter-Tel for their expertise and strength on the product development side." Said Langdon, "The Inter-Tel 7000 is a great example of forward thinking, and we're really pleased to see Inter-Tel come out with a product that is going to give us an edge on our competition when we go out to sell products."

He continued, "Once we saw the 7000, we realized there was a huge fit for NNU and for the student body."

In the past DataTel would spend a lot of time assigning extensions and manually programming voice mailboxes, etc., that were static, and would remain associated with the dorm rooms at NNU. "For management, "Langdon explained, "we can cut down on the time it takes to assign, program and do all the manual things that we had to do with the old system."

The change has been dramatic. According to Langdon, "Now if I'm a first year student, and my extension is for example 1000, that extension stays with me for the whole transition until I graduate, no matter where I move on the campus."

Taking advantage of the 7000's capabilities, "I can make that extension ring wherever I want it to ring, which is a big benefit for the students and one that gives them flexibility."

I asked Kellerer about the transition. "I'll be honest, when we first said this was going to be a beta, I had several people call me crazy. 'A telephone system is way too important to be testing on-the-fly,' they said. And frankly, I was concerned with being down for a period of time. But Inter-Tel came in with their team, all the homework had been done in advance, DataTel knew our network, made sure that was all right, and they came in and got the system up and running in a very short period of time.

"Our offices were down for maybe ten minutes in the transition. The students, who all have analog lines, were down for maybe a couple of hours, because actual physical wires had to be punched down. I've been very impressed with how smoothly it's gone."

Among the more useful capabilities is the ability for the phone system to serve different constituents differently, by integrating the various business processes of varying departments into the communications system. "Admissions is its own business," Langdon said. "They want their phones to function one way, and student enrollment wants it to function another way, and on down the chain, and the 7000 is able to accommodate everyone flexibly."

Kellerer added, "When we looked at some other products, the business processes were built into the product, and you needed to mold your business to meet the phone. And that doesn't work on a campus. We have so many different business processes, and we need to be flexible regarding business process, and Inter-Tel was able to show me how that worked on their system."

In addition, the ability to integrate future business processes was critical to Kellerer's decision making process when choosing the 7000. "We have the architecture and the plan down; we just need to write the interfaces to new business applications, and we'll be ready to go."

> Using SIP at its core, the system is designed to be redundant and secure and offers an easy-to-use interface...

One of the obvious benefits of the new system is the control that people get in terms of call routing. Kellerer's staff can reach him any time, anywhere, because he sets his call routing profile to allow them to reach him when they need to. While Kellerer manages all the rules of how his calls are handled, he says the toughest thing is to manage how other employees use the system.

"We have some older staff, who may simply want to pick up the phone, whereas I want them to pick up the phone when they're there, but when they're not there, I want them to put their status on something else."

Langdon added, "I think that was a huge enhancement for NNU. The Personal Communicator on the 7000 has made it very easy for people to manipulate how they want their calls handled."

The Personal Communicator is the presence engine on the 7000.

Eric also touted the system's ability to help serve the International students as well as the professors who travel internationally. In fact every student has a cross-cultural element of their education at NNU, and as such they will all spend a significant amount of time traveling away from the University.

One such faculty member is Dr. Jennifer Chase, Professor of Biology, who spent time last semester teaching in the UK.

Dr. Chase was working in a lab in the UK, and with the good quality Internet connection there, and the XLite softphone (from Counterpath) deployed on her laptop, it was completely seamless for people to call her. "It was exceedingly helpful when I had research students here (in Nampa) to be easily in contact, just an extension call away."

"Certainly professors work at home frequently, so I can sit in my office at home correcting papers or what have you, but still be just as available as if I was in my office at the

University," she continued. "It's also been a good advantage to call prospective students from home, and have the caller ID show up as NNU when I dial them."

And when one considers that the University faculty often is charged with calling these potential students all over the country to aid in recruitment, from their home phones, you could see why they would be happy with any solution that enables them to avoid the high tolls associated with calling.

And managing the phone systems and user preferences have become a breeze with the 7000 as well. "It's very easy, when we can adjust all the settings on the telephone from a computer. It's all very intuitive."

In further discussing the ability to enable remote users, Kellerer mentioned the ability to load a SIP compatible softphone on the student or faculty member's laptop, and in fact he declared that another deciding factor on selecting the Inter-Tel 700 was the product's SIP core. "One of the things that was important to me was the adherence to standards. I didn't want another proprietary system," he said.

> ...softphones... integrate with the 7000's because of the built-in SIP capabilities.

It's important to note that the softphones are third-party products and they are easily deployed remotely and integrate with the 7000's because of the built-in SIP capabilities.

There are many advantages to moving to a standards-based system. Said Kellerer, "The SIP part of it is important to enable communication between people and devices. This year is not all I'm concerned about. I need to know that next year or next week, I can choose to get whatever I need to meet my requirements at that time."

According to Kellerer, one of the other exciting things was the response from the campus. We walked around the Nampa campus and dropped in on a number of people, who universally were thrilled to have this new phone system.

Take Marsha Rogers, who operates the switchboard in the Administration building. She is very satisfied with the new system, and among her favorite benefits is the ease with which she is able to train student helpers to work with the system. She uses the Inter-Tel attendant console, together with the on-screen software component of that solution to route calls throughout the University. "I've been here 14 years, and over that time, we've had three phone systems. This is by far the most useful one yet," she said.

Katie Salisbury wears several hats. In addition to her role in the office of the registrar, she works in several other offices as well. She particularly likes the ability to manage her messages, specifically the ability to pull messages out of her voicemail out of order. She also takes advantage of the presence management capabilities of the Inter-Tel 7000 to handle her call routing.

I spoke to an administrative assistant in the business office, who loved the conference call features as well as the use of caller ID to help properly deal with calls.

In the end, Kellerer is especially grateful for the system's ability to expand and grow with the University as needs evolve over time. His upcoming projects are integrating the phone system to enable unified messaging and see that synchronized with the University's Novell software, to have all the contacts in Groupwise be connected.

He's happy with the IM capabilities through Personal Communicator, based on the experience in his department, but he'd like to see that expand to include other groups on campus.

Kellerer believes that video too will have an impact on the system as adoption increases in the months and years to come. He's also looking for ways to integrate the phone system's Web interface into the campus' portal. "I don't want everyone to have to log in to 10 different interfaces. It would be great to have everything they need right there in on their portal."

Perhaps most of all he's looking forward to enabling lots of different services to work on the different devices that his constituents choose. "We have over 40 services that we're currently offering, and while not everyone subscribes to all of these services, we want all of these services to come to the device of their choice and not have to be logged in 10 different times."

Greg Galitizine is Group Editorial Director of TMC.

Session Border Control

By Richard "Zippy" Grigonis

SIP has gone through many changes since its humble beginnings. Still, SIP retains just enough of its simple former "endpoint-to-endpoint" characteristics to make it a bit difficult to pass across network borders and through firewalls. NAT (Network Address Translation) and PAT (Port Address Translation) ensure that both real IP addresses and port numbers are disguised behind a few public ones facing the Internet. This is essentially true of a "proxy" server that sits between a client application (e.g., web browser) and a real server (i.e., web server), grabbing all requests sent to the real server and fulfilling them itself.

In the enterprise, one solution is a "SIP-aware" firewall such as those sold by Intertex and Ingate; the other, more general solution is a Session Border Controller (SBC), which can "punch a hole" through network borders and NAT/PAT-friendly firewalls so that SIP-based VoIP devices can connect to each other. Some advanced SBCs also resolve peering, quality of service (QoS), and other issues, as we shall see.

Of course, an SBC should ideally support such protocols as TDP/UDP (Traffic Distribution Protocol/(User Datagram Protocol) and interwork with TLS (Transport Layer Security, the successor to SSL or the Secure Socket Layer) to successfully deal with SIP. These capabilities and more can be found in high performance SBCs such as the nCite by (news - alert) Netrake (http://www.netrake.com), now an AudioCodes (http://www.audiocodes.com) company. (news - alert) The nCite can do load balancing, offers H.323/SIP interworking, handles up to 21,000 concurrent sessions, achieves less than 31 microseconds of latency, offers a built-in protocol debugger and downstream failure detection, and even supports lawful intercept of calls via its CALEA compliance.

As IP Communications continues to expand globally, so will SBCs and the devices that harbor SBC functionality. A report entitled "Service Provider Next Gen Voice and IMS Equipment", by the market research firm Infonetics Research shows that up until the third quarter of 2006, the session border controller segment grew 8% QoQ and up 102% YoY. Infonetics projects that the SBC market will grow from $86 million in 2005 to $613 million in 2009, and worldwide service provider revenues for VoIP and IMS equipment will more than double between 2005 and 2009, from $2.5 billion to $5.8 billion.

Interestingly, (news - alert) Acme Packet (http://www.acmepacket.com) leads what Infonetics calls "the fast-growing session border controller (SBC) segment". That's not surprising, since Acme Packet's Net-Net family is used by over 240 service providers, including 21 of the top 25 wireline and wireless providers worldwide. Their success owes in part to the flexibility of their SBC in deployments ranging from VoIP trunking to hosted enterprise and residential services,
and support for such varied protocols as SIP, H.323, MGCP/NCS and H.248. Acme Packet SBCs can cross multiple border points — at the interconnect, access

network and data center.

Peering Edge & Access Edge SBCs

We've been speaking of SBCs as if they were a specific, well-defined item. Actually, we can now define two broad types of SBCs: The original "peering edge" and "access edge" SBCs.

Rod Hodgman is VP of Marketing at Covergence (news - alert) (http://www.covergence.com) which offers the Eclipse, a second-generation IMS-compliant, access edge session border controller that has all sorts of capabilities not found in earlier peering-session SBCs.

"We provide a specialized session border controller that's specifically designed for the access edge," says Hodgman. "To tie this to IMS, just look at what IMS and TISPAN does. Within that standards body they've defined both an access edge SBC and peering edge SBC. They did that because issues faced in those two environments are inherently different."

"To date, almost all SBC implementations have been peering edge SBCs," says Hodgman, "where you are setting up peering relationships among a few trusted providers. Access edge SBCs, on the other hand, are very different. Rather than a few trusted network interfaces, it must deal with hundreds of thousands or even millions of untrusted subscribers along with a diversity of endpoints, applications and interoperability issues. It's about a lot more than VoIP — it's about instant messaging, presence, find-me/follow-me, click-to-talk and all of these kinds of things. It must allow service providers and enterprises to use such non-VoIP services too. So it's a very different and difficult environment in the areas of scaling the access edge and in terms of providing the kind of comprehensive security necessary at the edge but without impacting performance or scalability."

"We definitely play in the access edge SBC segment," says Hodgman. "We see the market in 2007 recognizing these two different types of products and we think a lot of what happens in 2007 will center on connecting users to their services, rather than just connecting networks, which is what people have done for the last two or three years."

"Our customers fall into two categories," says Hodgman.

The Market for Edge Session Border Controllers (SBCs)

Customer Type	Customer Needs	Competitive Position (e.g. Eclipse)	Market
Consumer Services	Needs to scale the access-edge	▸ Process registrations and floods ▸ Predictable scale active endpoints ▸ Troubleshoot and isolate faults	Established
Business Services	Needs "business-grade" VoIP to win enterprise customers	▸ Defend against attacks, outages and theft – w/out degrading performance ▸ Intelligent routing, failover, . . .	Established
	Needs IM, presence, video, conferencing, etc. to increase revenues and customers	▸ "Business-grade" IM, presence, video, push-to-talk, etc. - not just VOIP	Rapidly Emerging
Enterprise	Need "business-grade" real-time services	▸ Policy-based security and control ▸ Monitoring and alerting ▸ PSTN failover, local call completion, . . ▸ Deep packet inspection, validation, repair, mgmt., etc. ▸ Consistent policy across all applications	Emerging

Table courtesy of Covergence, Inc.

"They're either companies that have begun to scale their access edge and have found that a peering SBC doesn't support the number of endpoints, the registration rates, the kind of devices and applications they want to support, or the stateful connections and comprehensive security they need. For those customers in those categories, we come in and provide the access edge solution for them."

"For us, the access edge market really consists of three customers segments," says Hodgman. "First, companies providing consumer services that really need to scale the access edge; they have some experience with it, but they've run into some problems and they're out in the marketplace looking for a more optimized solution to that problem. Second are those providing services for business. They need to provide what we call 'business-grade' VoIP to be able to provide service to enterprise customers. By 'business-grade VoIP' I mean VoIP that is secure, reliable, available and of high quality. Generally, this comes down to providing the comprehensive security that's necessary to allow them to deploy the system. It passes the internal requirements of their security teams' policies. There are companies that need to or want to provide services beyond VoIP. That would be instant messaging, presence, video, conferencing, click-to-call, find-me/follow-me, and those sort of features so that they can not only sell business-grade services to the enterprise but sell advanced services and therefore get more revenue per customer out of the enterprise. This is a rapidly emerging market. The third category is the enterprise customer, who by definition requires a business-grade communications and collaboration systems for them to be able to deploy either VoIP or real-time collaboration. This is an early-stage, emerging market, in which we also participate."

"Our customers don't need SBCs," says Hodgman. "They need results. If they're service providers, they need to increase high-value enterprise customers; to do that, they need business-grade services and they need to increase revenues through the provisioning of additional services such as multimedia services beyond VoIP. They need to reduce their costs and improve customers satisfaction by improved management — I can't emphasize this enough. Our product has some really sophisticated and advanced management tools in it, and one of the things we've spent a lot of time doing, when we visit customers, is cleaning up what is essentially a rats' nest of these malconfigured TAs and devices that are themselves sending up registration storms and causing a lot of mayhem on the network."

"For the larger providers, clearly migrating to IMS is a big, important factor for them," says Hodgman. "At the enterprise, it's all about reducing costs and streamlining business processes. They can do that via broad deployment of real-time communications systems, either VoIP or collaboration as long as it needs there internal security policies and external regulatory compliance."

Integrated versus Stand-alone SBCs

With the flexibility and miniaturization afforded by today's technology, we no longer have to talk about SBCs as if each were a discrete "box". We're really talking about SBC functionality, and that can be situated anywhere and in any form. Gateways can be combined with session border control functionality — indeed, you may find SBC functionality embedded or integrated into other network elements, as opposed to existing as a separate SBC box.

For example, take Cisco's 7600 Series Routers, said to be the industry's first comprehensive Carrier Ethernet service edge platform for converged IP video, voice and data offerings with mobility. You can find an integrated SBC on the Cisco 7600 Series Routers, which provides per-session control and management of IP multimedia traffic based on SIP and H.323. Such a unified SBC implementation obviates the need for additional appliances and overlay networks, enabling multi-service scalability with lower costs. The Cisco SBC is also integrated with the Cisco XR 12000 Multiservice Edge platform for business services.

SBCs on the Move

Even when SBC functionality is concentrated in a single box, it's a good idea to have a system flexible enough so that it can be situated anywhere in the network. In this vein, (news - alert) Tekelec (http://www.tekelec.com) a developer of high-performance network applications for next-gen fixed, mobile and packet networks, offers the Tekelec 6000 VoIP Application Server that includes automatic disaster recovery features. The Tekelec 6000 can re-route all IP phone calls automatically to a wireline or mobile phone if an operator's network is disrupted by a natural catastrophe such as wind, or flooding. The architecture of the Tekelec 6000 thus allows operators to situate system servers and session border controllers in geographically diverse locations, ensuring no single points of failure and continued operation because of extreme weather conditions.

Over at (news - alert) NexTone (http://www.nextone.com), their IntelliConnect™ System has earned acceptance from the Rural Utilities Service (RUS) branch of the United States Department of Agriculture (USDA). Therefore, it's now easier for the Independent Operating Companies (IOCs) servicing rural markets to extend VoIP services to all of their wireline subscribers via RUS financing to purchase and deploy NexTone's IntelliConnect System, which includes its Session Border Controller (SBC) for securing bilateral interconnects and the Multiprotocol Session Exchange (MSX), a platform for interconnecting SIP and H.323 networks.

But wherever an SBC happens to be, it must be subject to periodic testing. Although VoIP quality steadily improves, it's still possible for a phone call to suffer problems as jitter buffers in session border controllers and media gateways overflow and begin discarding voice packets. Now, however, service providers can remotely test to a customer's multimedia telephone adapter (MTA) or SBC to monitor and troubleshoot VoIP quality using (news - alert) Tektronix' (http://www.tektronix.com) loopback testing offering. Over 50 service-quality metrics can be measured on-demand including speech quality (MOS, echo), call connectivity (PDD, CCR), DTMF transmission, packet-loss, jitter and delay.

Attacks on SBCs

As both workers and their devices become increasing mobile under IMS (IP Multimedia Subsystem) and FMC (Fixed/Mobile Convergence), the concept of a defensible perimeter around an organization lorded over by conventional security devices and software (firewalls, session border controllers, intrusion detection/prevention systems, etc.) is starting to evaporate. Since every device has an independent interface and vulnerable internal workings, every device now essentially has its own perimeter that is what really must be protected by attacks by hackers or crackers, such as SIP-specific denial-of-service (DoS) attacks.

Fortunately, SBCs have become more secure at a pace slightly ahead of the hackers. NexTone's Multiprotocol Session Exchange (MSX), for example, successfully passed the most call-intensive attack test plan executed against any SBC tested by Chris Bajorek's CT Labs, a testing facility that's now part of (news - alert) Empirix (http://www.empirix.com). In the course of the test, NexTone's MSX successfully processed over 18.7 million legitimate SIP calls in a 62-hour period while rejecting a SIP-specific DoS attack.

Future SBCs

As the network expands in size and bandwidth, and as more processing-intensive applications make their appearance, SBC hardware will have to expand in power too. New form factors such as AdvancedTCA (ATCA), MicroTCA and ATCA Mezzanine Cards (AMCs) will come into play. (news - alert) RadiSys (http://www.radisys.com), for example, recently expanded its ATCA and AMC family with two new products based on the Cavium Networks' latest OCTEON™ processors: The Promentum ATCA-72xx, a high performance, modular Gigabit Ethernet Line card for ATCA systems offering 4, 8, 12 or 16 Gigabit Ethernet interfaces, and the RadiSys AMC-7211 to provide power efficient, packet and security processing for customers requiring AMC modules for their ATCA and uTCA platforms. These new ATCA and AMC solutions provide high density Gigabit Ethernet interfaces with sophisticated dataplane hardware acceleration, and will power the next generation of high performance products such as SBCs, Media Gateways, Edge Routers, and Security Gateways.

But if you want to know what will definitely happen to the future of many SBC companies, at least in the short term, then look no further than Netrake Corporation, a leading provider of SBCs, media gateways and security gateways, which has been acquired by AudioCodes, makers of voice network products such as their superlative packetization boards. There has been a profusion of SBC makers in the past, and one suspects that some shake-out/mergers and acquisition goings-on will continue for quite a while.

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

SIMPLE in the Enterprise

By Richard “Zippy” Grigonis

If and when IM (Instant Messaging) and presence eventually become an IETF standard, the SIMPLE protocol will undoubtedly be the principal component, thanks in part to its ability to integrate IM/presence with voice, video, data-sharing, and other elements of conferencing and real-time collaboration. Every major telecom software and hardware vendor now has support for SIP/SIMPLE, or soon will.

SIP is a control protocol for initiating, modifying and terminating IP sessions such as multimedia conferences and IP telephony calls. Its functions thus parallel those of SS7 in the PSTN (Public Switched Telephone Network). SIP simply makes communication possible. SIP works with other IETF (Internet Engineering Task Force) protocols to actually achieve communications. Given its humble beginnings, SIP wasn’t originally designed to centrally concentrate all call intelligence, or provide network usage information for billing purposes, or handle all facets of security, or support overlapping dialing, or completely replace the PSTN (the original NNI, or network-to-network interface approach, presumes a closed network like an SS7 network). This is why SIP has undergone some extensions in recent years and/or relies on other protocols (e.g. XCON for centralized conferencing), which in turn is why SIP is popping up everywhere.

SIP can deal with multimedia multi-party sessions; the use of SIP for two-party multimedia telephony was first suggested by Scott Petrack of eDial (later Divisional CEO of Alcatel), because two-party audio-only calls seemed to him to be a trivial limiting case of collaboration. The varied nature of multimedia suggests conferencing and collaboration applications, and for conferencing and collaboration to be truly effective, IM (instant messaging) and presence-awareness (or “availability”) signaling must be supplied either by adding additional signaling extensions to SIP, or else developing something different, such as the IETF’s XMPP (Extensible Messaging and Presence Protocol) suite, found in the Jabber IM system.

In the case of SIP, an IETF working group extended SIP functions and came up with SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) — Jonathan Rosenberg, formerly of dynamicsoft, now of Cisco, co-author of the original SIP specification, is also the principal figure in SIMPLE’s development.

Don’t let SIMPLE’s name fool you. As the Wikipedia says, “Despite its name, SIMPLE is not simple. It is defined by about 30 documents or more than 1000 pages (7 times more than HTTP 1.1, 15 times more than SMTP and IRC).”

Even so, SIP/SIMPLE definitely overshadows competing protocols. Most non-SIP IM protocols are relegated to solely handling text communications. This compels their users to resort to other approaches when using voice and video — namely, SIP. The SIP extensions in SIMPLE are a comprehensive standards family for events, presence and IM, so non-SIP IM will probably fade over time, since most public and enterprise services are deploying SIMPLE or migrating to it. Major IT companies such as Microsoft and IBM support SIMPLE and SIP/SIMPLE is the official 3G wireless multimedia protocol and a cornerstone of IMS (IP Multimedia Subsystem), a universal service architecture for the worlds of both wireline and wireless.

Microsoft Windows Messenger soon adopted SIMPLE, as did the Microsoft Office Live Communications Server (LCS), starting in 2003. In 2005, some improvements were made. Microsoft Office Communicator 2005 provides integrated communications capabilities that include IM, extendable presence with Microsoft Exchange Server calendar information, PC based voice/video, and VoIP telephony. Live Communications Server enables users to determine the status of other users (e.g., away, buys, idle, do not disturb) and whether they are offline or unavailable. Microsoft Office Communicator 2005 can display offline presence details even if a user is not logged in — you just right-click on the user name, and the system will tell you whether the person in question can be reached via other forms of communication such as email or a telephone. The platform can also serve as the presence engine for team sites and portals, thus making available presence and IM access from within Windows SharePoint Services and SharePoint Portal Server sites. Live Communications Server 2005 (with Service Pack 1) allows security enhanced IM and presence information to travel between public IM service providers so that enterprises can collaborate with business partners just like co-workers, while keeping sensitive business information encrypted and logged. This can be taken to its most sophisticated incarnation, federation, which enables an organization to establish trusted relationships with other organizations, allowing users to initiate and share IM sessions and subscribe to user presence across organizational and network boundaries without a VPN connection.

SIMPLE's use will continue on into Microsoft's successor platform, Microsoft Office Communications Server 2007 (OCS), currently in beta, to be released in the second quarter of 2007. ("Live" was nixed from the product's name to avoid confusion with Microsoft's hosted Office Live and Windows Live.) Deployed with Microsoft Exchange Server 2007, the business-oriented OCS will help make Microsoft into a major player in the unified communications market. OCS will also integrate its functionality into Microsoft's Office products — particularly the 2007 Microsoft Office applications. Clicking on a fellow worker's name in Office Outlook 2007, Office Word 2007, or the Office Communicator client will initiate a call. A revision to the desktop Office Communicator software offers VoIP capabilities along with various telephony capabilities (users can enjoy such features as call hold, call forward, and call transfer), web conferencing (previously offered via hosting in Live Meeting), incremental chat, corporate instant messaging and richer presence capabilities. Microsoft has also forged alliances with partners to help facilitate the development of sophisticated unified communications platforms, such as the Innovative Communications Alliance (ICA) with Nortel. This will be a user-centric software-based approach enabling a user to maintain a single identity across email, voice mail, VoIP, call processing, IM and video, as well as embedding communications functionality into the Microsoft Office system and third-party software applications.

Microsoft is considering adding support for the XMPP protocol, but their commitment to SIP and SIMPLE is such that you shouldn't expect anything any time soon. IBM's Sametime IM and web conferencing platform supports both XMPP and SIP/SIMPLE. IBM Sametime is interoperable with AIM, and IBM recently announced upcoming interoperability among Sametime, Google Talk and Yahoo IM networks. A feature in the new Lotus Sametime 7.5 is location awareness, which informs you not just if your chat partner is available, but whether he or she is at a desk, in a home office, working in a hotel room, or elsewhere. The person can leave a detailed message, such as "I'm in a meeting with a vendor".

SIP/SIMPLE Meets Jabber/XMPP

SIMPLE and XMPP have been positioned as competing technologies. XMPP is basically a data transport protocol for streaming XML elements — called "stanzas" — between any two network endpoints. Message and presence stanzas are both defined as core data elements in XMPP and are generally used to exchange instant messages and presence information between IM users. XML is extensible and applications can use the general semantics of these stanza types for other purposes, but XMPP focuses on IM and presence. SIMPLE, on the other hand, is more general in nature, making it suitable for not just presence and IM, but voice, video, push-to-talk and other types of communications. SIP has even been used to add voice to XMPP-based IM systems.

There are ways for SIMPLE and XMPP can work together, however. SIMPLE is basically a paging protocol meant to perform signaling via SIP methods but not actually carry anything else. For SIP/SIMPLE, the signaling occurs using basic SIP methods and the media data (e.g. voice and video) is exchanged using dedicated data transport mechanisms such as RTP (Real-Time Transport Protocol). Since SIP can rely on various media types and media transports, it can use XMPP as just another one of these media transports, a peer to RTP. SIP can then be used to set up and manage XMPP IM sessions.

Even so, XMPP is still slowly gaining in popularity. Sun's Java System Instant Messaging offers presence by using XMPP in an integration of Sun's Messaging, Portal and Calendar servers. In January 2006 Google made Google Talk into an open federation product so that you can now chat with users on other XMPP services. Shortly thereafter, Jive Software released its XMPP-based Wildfire 3.1 enterprise IM server, that allows for communication with users on proprietary IM networks, such as AOL.

SIMPLE still has an edge, however, and will continue to worm its way into everyday business and consumer applications.

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

Q & A

60 Seconds with Timothy Winters, Software Management Engineer

Timothy Winters is Software Management Engineer at the University of New Hampshire InterOperability Laboratory (UNH-IOL). Most of the lab's work staff are students. Indeed, Winters started out as a computer science undergrad (1999-2004) and became a staff member. His communications background is in IP. He started in the Multicast IP Group and then moved to the IPv6 Group, where he has worked for the past two years. Now, he's moved into the VoIP/IPv6 area. Founded in 1988, the lab is owned by the University but is supported commercially by the companies whose products they test. Historically the lab has focused more on IP than telecom, and it has a very strong base in testing all 'flavors' of Ethernet.

Richard "Zippy" Grigonis recently spoke with Timothy Winters about their testing of SIP-enabled products.

RG: Do you charge a straight fee for your work?

TW: We prefer a collaborative model. We're officially a nonprofit even though we operate like a company, but we like to position ourselves as a neutral party because we don't have shareholders who are waiting for us to turn a profit. We provide a neutral ground for companies to collaborate with each other and to promote conformance to standards so we verify that products perform to ITU and IEEE standards. We also test interoperability between multiple vendors' products.

For example, let's say that a company comes to us and wants to join or VoIP Group. We consider it a membership rather than us providing services, so they become a 'member' of what we call a 'consortium'. They join the VoIP Consortium for an annual fee. They also provide one piece of equipment into a shared testbed to which all of the members of that consortium have equal access. For that fee we also provide them with as much testing as they can get within a fairness algorithm for everybody. We schedule the testing in one-week blocks. What that actually means is that, unlike other labs that charge per test or just give you a one-shot deal where you either pass or you don't, we tend to work much more collaboratively with our members. The company can even come back with other products or with the same product to schedule another week of testing. Basically, a company can use as an extension of their R&D department. They can tweak the code as they go and leave the lab with a product that's now a lot better than when they first arrived.

RG: How do you deal with confidentiality?

TW: We also provide confidential reports to each of these companies which are very technically detailed, describing exactly what we did and how the product fared. All of our member companies are listed on our website (http://www.iol.unh.edu) and we have about 20 different technology groups covering things such as IPv6, storage, VoIP, WiFi and what-not. We serve 150 member companies, and we work with the various industry forums.

RG: Where does SIP come in, exactly?

TW: SIP testing is part of our VoIP Consortium. We currently offer SIP conformance testing. We have a SIP Interop/'torture test' suite we currently offer to our VoIP Consortium members. That basically tests an item against the standards, or, in the interoperability case, it takes specific test cases and companies get to use our testbed. Part of the agreement that member vendors have when they come to work with us is that they leave a platform here in the testbed and the equipment usually stays. So that makes for very large testbeds.

RG: With so much equipoment, do you hold 'plugfests'?

TW: Oh yes, we'll periodically have a week of open testing where we create a big network and anybody can come and throw things at it and we see what they do. As a neutral test lab, we provide a "DMZ" for industry plugfests for such organizations as the SIP Forum. In January 2007 we're hosting the IMS Forum's first interoperability test event. Still, a great deal of our work involves testing individual companies' products for standards-compliance and multi-vendor interoperability. Our VoIP Consortium right now is populated by 3Com, BlueNote, Empirix, netopia and Sonus — for whom we provide interop qualification testing.

In short, our work results in a tremendous, unbiased, vendor-neutral knowledge base.

We like to think that our set-up here at the UNH-IOL practically guarantees a 'win-win' situation for both students and the companies whose products we test. As time goes on, I think you'll see other examples of such alliances forged between academia and the corporate world. Students get valuable hands-on technical and testing experience, and companies end up with better products without spending all of their money on R&D.

ADVERTISING INDEX

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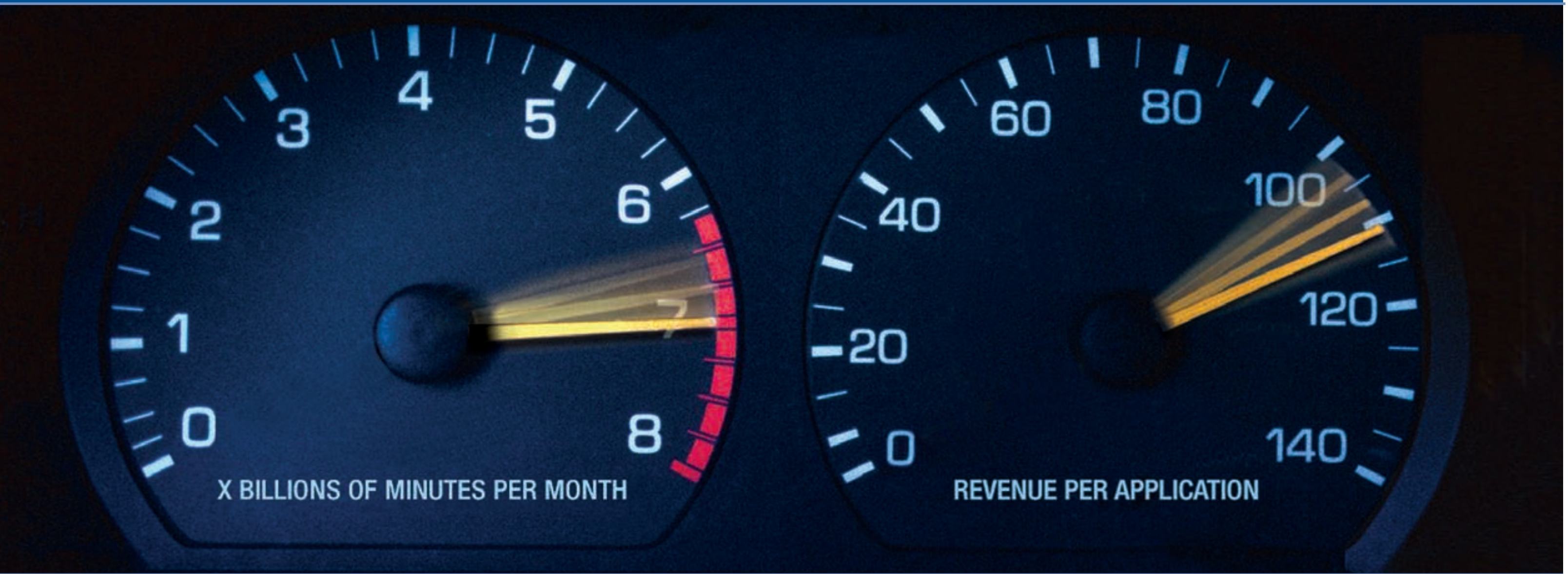

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