Exposing the Hidden Costs of VoIP

www.sipmag.com Volume 1 • No.4 July 2006

TMC SIP MAGAZINE™

The Authority On Session Initiation Protocol

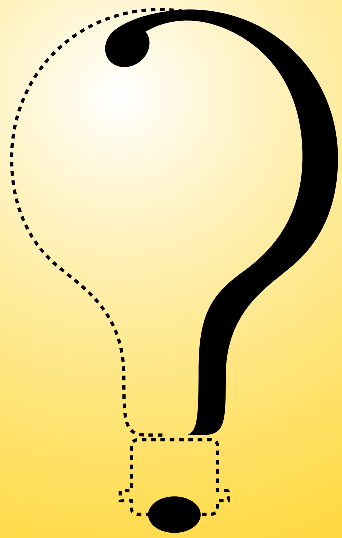
IM & the Case for Controlled Connectivity

The Need for High Availability

Is VoIP Secure?

To Be or Not To Be?

The Fate of Session Border Controllers



EDITOR'S NOTE

Microsoft SIPs Into Unified Communications

by
Greg Galitzine

SIP is increasingly making its way into enterprise communication solutions of all shapes and sizes and, frankly, who knows enterprises better than Microsoft?

Jeff Raikes, president of the Microsoft Business Division, (quote - news - alert) recently unveiled the company's vision, technology road map, and partner framework for unified communications.

"Unified communications (news - alert) will drive the next major advancement in individual, team, and organizational productivity in today's 24x7, always-connected and increasingly mobile work environment," Raikes said. "We believe that through software, we can transform business communications (bringing down both its cost and complexity) by now integrating voice communications with the familiar and powerful communications and collaboration experiences provided by Microsoft."

Now, I've been hearing about the fact that Unified Messaging (later expanded to Unified Communications) is the "next big thing" for nearly 10 years straight. I'm not exaggerating! If you've been following the industry at all, you know it's true. Maybe SIP is finally robust enough and ubiquitous enough to make this long-standing potential a reality. Time will tell.

At the heart of Microsoft's vision is the Microsoft Office Communications Server 2007, a SIP-based real-time communication platform designed to enable presence-based VoIP call management; audio, video, and Web conferencing; and instant messaging communication within and across existing software applications, services, and devices.

Microsoft Office Communicator 2007 is a unified communications client that works in tandem with Office Communications Server 2007 designed to deliver a presence-based, enterprise VoIP softphone as well as secure, enterprise-grade instant messaging that allows for inter-company federation and connectivity to public instant messaging networks such as MSN, AOL, and Yahoo!

The new initiative also calls for partnering. The reliance on SIP enables a whole universe of innovative devices and opens the door for Microsoft to engage in partnerships with a range of companies such as GN Netcom, Logitech, Motorola, Plantronics, Samsung, and Tatung for PC peripheral devices, such as USB handsets, wireless USB headsets, USB Webcams, and PC monitors with built-in audio and video components. Other partners will include Polycom, LG-Nortel, and Thomson Telecom.

Microsoft also announced new business alliances with HP, Motorola, and Siemens to deliver on its vision for unified communications. HP will provide hardware devices and systems integration services. Motorola will deliver mobile devices and network hardware. Siemens will advance the transformation of telephony, audio, video and Web conferencing, instant messaging, and e-mail into a single unified communications platform.

Microsoft also announced it had selected Quintum Technologies as a hardware partner. The Quintum Tenor line of VoIP switches and gateways support the voice, presence awareness, and instant messaging capabilities of Office Communications Server 2007 and allow it to be integrated into the existing PSTN/PBX voice network infrastructure.

While it will take some time for all of these initiatives to become real products, (expect general availability of most of the portfolio by mid 2007) the fact is that standards-based solutions, primarily based on SIP, are leading the wave of products that will be adopted by enterprises going forward. And when a company of Microsoft's stature makes this kind of a bet on SIP, it can only be seen as a positive sign.

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*

EXECUTIVE OFFICERS
Nadji Tehrani, *Chairman and CEO*
Rich Tehrani, *President*
Dave Rodriguez, *VP of Publications, Conferences & Online Media*
Kevin J. Noonan, *VP of Business Development*
Michael Genaro, *VP of Marketing*

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**

About *SIP Magazine*®
SIP is a fundamental building block at the center of the telecom transformation taking place all around us. SIP creates unprecedented opportunities for resellers, developers, and service providers alike. *SIP Magazine*® focuses on providing readers with the information necessary to learn about and purchase the equipment, software, and services necessary to take advantage of this technology. *SIP Magazine*® readers include resellers, developers, MIS/networking departments, telecom departments, datacom departments, telcos/LECs, wireless/PCS providers, ISPs, and cable companies.

Subscriptions
Circulation Director, Shirley Russo, ext. 157 (srusso@tmcnet.com)
SIP Magazine® is published bi-monthly by Technology Marketing Corp. Annual digital subscriptions; Free to qualifying U.S., Canada and foreign subscribers. Annual price subscriptions; Free to qualifying U.S. subscribers; $25 U.S. nonqualifying, $35 Canada, $50 foreign qualifying and nonqualifying. All orders are payable in advance U.S. dollars drawn against a U.S. bank. Connecticut residents add applicable sales tax.

Identification Statement
SIP Magazine® is published bi-monthly by Technology Marketing Corporation, 1 Technology Plaza, Norwalk, CT 06854 USA. Annual digital subscriptions; Free to qualifying U.S., Canada and foreign subscribers. Annual print subscriptions; Free to qualifying U.S. subscribers; $25 U.S. nonqualifying, $35 Canada, $50 foreign qualifying and nonqualifying.

Postmaster: Send address changes to: *SIP Magazine*®, Technology Marketing Corporation, 1 Technology Plaza, Norwalk, CT 06854

A Technology Marketing Publication,
One Technology Plaza, Norwalk, CT 06854 U.S.A.
Phone: (203) 852-6800; Fax: (203) 866-3326

SIP MAGAZINE™

Volume 1/Number 4 July 2006

CONTENTS

feature articles

Go To Table of Contents | Go To Ad Index

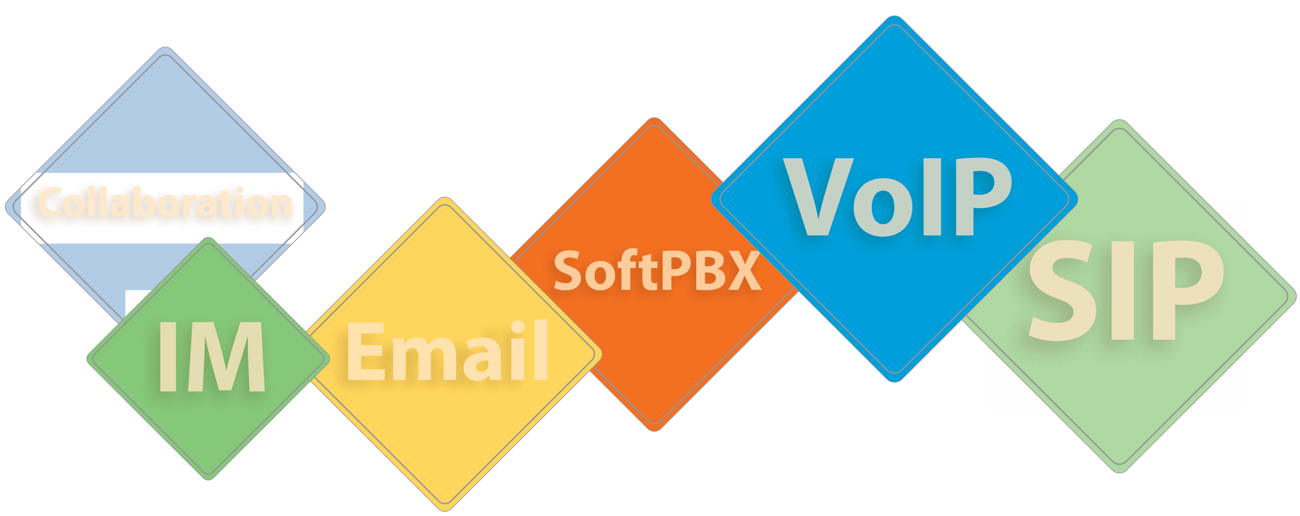
www.communigate.com

CommuniGate SYSTEMS

Trade In your
legacy mail systems

Trade Up to

Next Generation IP Communications



CommuniGate Systems is offering a Trade-In / Trade-Up program for legacy email systems. Stop paying outlandish fees for outdated technology. Trade up your systems to our world record holding, carrier-class technology and five nines uptime with CommuniGate Pro Dynamic Cluster architecture. To learn more about the program call 800.262.4722 or visit:

http://www.communigate.com/ads/tradeInTradeUp.html

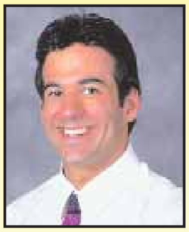
Download Free Trial at:
https://www.communigate.com/referral/leadform.html

2005 INTERNET TELEPHONY PRODUCT OF THE YEAR


UNDER NEW
MANAGEMENT
(AGAIN)!

PUBLISHER'S OUTLOOK

Supercalifragilisticexpi-SIP-idocious



by
Rich Tehrani

The proliferation of contact numbers and addresses is staggering. I have tried to keep my e-mail accounts to a minimum and I still have five. I have five different contact phone numbers and then there are fax lines for the various locations. On top of that I have addresses for Skype (news - alert) and other VoIP services. For all the advancement we are seeing in the world of communications we are just adding more and more confusion to our personal and business lives. What we need to do is simplify the world of communications.

The people at CommuniGate (news - alert) may disagree. Their goal is to SIPify communications, and they have solutions that allow an e-mail address to be the single address for users regardless of whether they want to communicate via voice, video, or text messaging. They are driving the world towards an e-mail-based communications model allowing users to focus less on devices and more on a single address for their regular communications.

This is not a new concept and others have lamented the concept of VoIP islands that don't speak with on another. If e-mail worked the way VoIP does today then when I e-mail from a Hotmail to an MSN account, someone would have to read the Hotmail e-mail and retype it on the MSN system. In other words, calls from one provider to another have to go over the PSTN. The concept of VoIP peering certainly can reduce PSTN (define - news - alert) usage but this line of reasoning will be taken at a later time.

Getting back to CommuniGate, they have recently taken their CommuniGate Pro v5.1 software and added an XML API and a Flash client for subscriber mobility. Their goal is to be the middleware that 'SIP enables' the two billion e-mail addresses in the world. Can you imagine if every user was on a common communications system that allowed free communications? What an amazing place the world would be.

If we can get the ILECs to generate revenue from this transformation, somehow even they might embrace the concept. Then there is that whole USF issue. Again, we will cross that bridge when we come to it.

If we can get the ILECs to generate revenue from this transformation, somehow even they might embrace the concept.

The new software comes with a client called Communify that has a Flash-based client and uses XIMSS, an XML API for messaging, scheduling, and signaling. The company believes this is the way to drive a full IMS strategy and, as you guessed, service providers are their target market. Well their product is touted to scale from the size of an SMB all the way up to the largest of carriers so they have a pretty large audience to address.

There is also support for XMPP or the Jabber protocol, which means CommuniGate users can now connect with users of other IM systems such as Google Talk. It is worth pointing out that SIP phones will also work with the system.

What excites me about companies like CommuniGate is that they are changing the way we all communicate as they simplify, err, make that SIPify the world of communications. It is natural to believe that one day soon we will all have a single identifier and some computer aided logic behind the scenes deciding what calls to take and when. It is also logical to believe that this identifier will be presence-enabled and likely use SIP or some future version of SIP.

If the future will be as I describe above, then surely the first step to getting there is by embracing some of the concepts laid out on this page. I am not sure if everyone agrees, so feel free to just send me a message to the as yet un-SIPified address rtehrani@tmcnet.com.

Stratus Rolls Out SIP-Based Mobile Call Convergence Solution

By Laura Stotler

Stratus Technologies (news - alert) has announced its Mobile Call Convergence (MCC) solution. The SIP-based software offers seamless handover of voice calls between GSM/CDMA cellular networks and 802.11 WiFi networks and enables a number of enhanced services.

The MCC helps achieve single-subscriber access to services and data using any device and over any network. It also enables new revenue-generating subscriber services while preparing networks for IMS capability. This opens the door for IP Voice and text messaging services, and service providers may now offer converged solutions that bridge VoIP and mobile networks and forward the goal of fixed/mobile convergence (FMC).

Providers who implement the MCC solution may offer subscriber access to the network using an array of mobile or IP-based devices via fixed residential/enterprise broadband or a WiFi connection. They may also offer converged subscriber phone numbers, in which one number identifies a subscriber (or multiple numbers based on preference). The subscriber may then receive calls over any network device when a calling party dials the number.

"MCC capabilities extend far beyond the limitations of UMA to enhance the service experience for subscribers, while giving providers immediate opportunity to increase revenue and reduce subscriber churn," said Ali Kafel, vice president of telecommunications at Stratus.

http://www.stratus.com

American Telecom Launches First Digital Clear DECT Internet Phone

American Telecom Services (ATS), (news - alert)a provider of both Digital Clear Internet phones, VoIP, and Pay 'n Talk pre-paid long distance communication services bundled with digital cordless multi-handset phones, announced the release of its first DECT 6.0 cordless multi-handset Internet phone. The release of the new home and small office phones represents a significant advancement in Internet based telephone technology and consumer ease of use.

Efficient design and production allows ATS to price the E6501 and E6502 in a range comparable to traditional phones and removes almost all of the barriers to entry for an Internet user's conversion to VoIP.

The E6501 includes ATS' patent-pending Digital Clear functionality and is expandable to up to five total handsets. The release of the E6501 is the first cordless phone to use DECT technology that integrates a router and a SIP-based VoIP platform in the charging base of the master cordless phone unit.

"Internet phone users no longer need to purchase a separate analog telephony adapter (ATA) and telephone or undergo the added effort and expense of re-wiring their home to use their service around their house because all the needed functionality is included in this single affordable device," said Adam Somer, Co-President of ATS.

360networks Selects Netrake for Voice Interconnect Service

By Johanne Torres

Netrake (news - alert) announced its nCite Session Border Controllers have been chosen by telecom broadband services provider 360networks (news - alert) to support deployment of the new SIPConnect Service, designed for cable operators (MSOs), ISPs, broadband service providers, and VoIP service providers.

The partnership calls for Netrake's nCite Session Border Controllers (SBCs) to be installed at interconnection and peering points in 360's network. The SBCs secure the network, ensure quality of service, and deliver SLA compliance. Netrake's system provides a comprehensive set of security functions for 360networks such as Denial of Service (DoS) protection, authentication, FW/NAT traversal, access control, and network topology hiding.

"Netrake's carrier-grade, fault-tolerant solution and performance are imperative for the high quality requirements of the 360networks' SIPConnect service," said Rick Coma, senior vice president at 360networks. "Additionally, Netrake's proven focus on security and scale made them a clear choice for our network."

360networks' SIPConnect service suite offers high quality voice services, comprising local telephone numbers, inbound and outbound local calling, domestic long distance, E911, local number portability, directory listing, operator services, and directory assistance. The service will integrate 360's CLEC status and robust fiber optic network with reach into tier 1, 2, 3, and 4 markets throughout the western half of the United States.

"We're thrilled to be working with 360networks to help expand and secure one of the largest long haul fiber networks in North America," said Netrake's vice president of sales, Terry Orosco. "The Netrake Session Border Controllers give 360networks the tools and scale required to offer its customers a secure suite of integrated next-generation IP services."

http://www.360.net
http://www.netrake.com

BandTel Introduces Itself to the VoIP Market

BandTel, (news - alert)a global provider of next-generation VoIP termination to the PSTN for high volume telecom users, such as call centers, enterprise users, teleconferencing companies and IVR users, announced it has received a second round of funding.

Seagrove, LLC, which specializes in financing small-to-middle market companies, lead the Series B investment. BandTel will use the funds to further expand its sales and marketing team, invest in marketing and business development initiatives and support the company's working capital needs.

BandTel's VoIP termination to the PSTN solutions provide users with a fault-tolerant, scalable VoIP architecture, which allows for continuous service and investment protection on legacy devices. Its SIP Softswitch technology allows users to receive service from one of two BandTel switching centers at any given time. The company's unique ability to connect call systems via a trunk-to-trunk transfer combined with the ability to cost-effectively terminate calls to and from any location in the world instantly unites global businesses with multiple users in various locations.

http://www.bandtel.com

Nokia's business SIP solution

Finnish mobile telecommunications equipment maker Nokia (news - alert) announced the launch of the Nokia Business Communication Solution. The new product is a hosted end-to-end SIP-based solution that helps operators and other service providers tap into the enterprise voice market by hosting advanced IP-based services for businesses.

The solution includes both network equipment and phones, such as Nokia Eseries business optimized devices, for empowering an enterprise's workforce with voice services on the company's cellular and multi-radio mobile devices or fixed IP phones.

The solution is made up of the Nokia Business Communication Application Server, Nokia handsets and client software, and a Web-based tool that allows enterprises to manage their services themselves. It also includes a comprehensive range of system integration services to streamline interworking with an enterprise"s existing private business exchange, or PBX, systems.

The solution can be an integral part of an advanced IP Multimedia Subsystem environment, such as the Nokia IMS for fixed and mobile.

In addition, the Nokia MSC Server System mobile softswitch brings additional benefits to operators who deploy the Nokia Business Communication Solution.

BellSouth Provides Carrier-Class Telecom Switch for Georgia Tech

BellSouth (news - alert) announced that it will provide a fully redundant, carrier-class Nortel telecommunications switch for Georgia Institute of Technology to support the communication needs of students, faculty and staff.

The advanced switch will support up to 50,000 phone lines and connects 113 classroom and administration buildings, dormitories, research laboratories, and sports facilities to friends, family, and colleagues around the world. BellSouth will begin implementation at the Atlanta campus this summer. Additional phases of the project will support other Georgia Tech sites throughout the state, including a research facility in Cobb County.

The Nortel CS 2100 solution for Georgia Tech includes a geographically split core design. If a portion of the switch fails, the other half will maintain uninterrupted service to Georgia Tech's campus. The switch is managed and maintained by dedicated BellSouth technicians and connects directly to BellSouth's self-healing, fully redundant SONET ring, which consists of two diverse paths of fiber that connect both distributed switch core locations on the Georgia Tech campus to each other and the BellSouth network. If one fiber path is cut, the signal will travel over the second fiber route. If both fiber paths are cut, the signal reverses and travels around the remaining arc of the loop.

In addition to the switch's survivability, Georgia Tech selected the solution because it supports new and emerging protocols, such as Session Initiation Protocol (SIP) for VoIP and dual mode wireless technologies.

http://www.bellsouth.com

Peer Telecom Taps TelTel to Deploy VoIP Service

By Johanne Torres

SIP-based Internet telephony provider TelTel (news - alert)announced on Tuesday that Peer Telecom is using TelTel's SIP Virtual Network Operator (SVNO) program to deliver its Eggee VoIP telephony service in France.

The SVNO program will offer Peer Telecom a suite of products comprised of a managed network, branded hardware devices, and softphone client, in order to enable the telecom to rapidly deploy SIP-based Internet telephony.

The Eggee service seems to be a good choice for the SMB and SOHO markets offering its customers the ability to call more than 130 destinations for the cost of a local call. By partnering with TelTel, Peer Telecom will also be able to offer its users free access to over 1.5 million users of Public SIP Telephony Network (PsipTN), and to more than 30 million VoIP users of SIP. Peer Telecom can also use the PsipTN to deploy VoIP-based services, such as conference calling and desktop sharing.

"Peer to peer managed networks is the best alternative to Centrex solutions for the SMB and SOHO market," said Peer Telecom's founder Micha Benoliel. "TelTel's SVNO program allowed us to rapidly deploy the Eggee service and offer our customers a standards-based solution that cost-effectively connects them all over the world via the PsipTN."

http://www.peertelecom.com
http://www.teltel.com

Tekelec Announces TekMedia IMS-Ready SMS Solution

Tekelec, (news - alert) a developer of high-performance network applications for next-generation fixed, mobile and packet networks, unveiled the TekMedia Short Message Service (SMS) Solution that enables operators to deliver advanced messaging capabilities without costly network overhaul.

The TekMedia SMS Solution provides a secure, modular architecture that supports the evolution of SMS beyond simple text messaging to include mobile-to-application, application-to-mobile and application-to-application services, such as tele-voting and telemetry. The Tekelec solution also will support future multimedia instant messaging (MM-IM) applications using session initiation protocol (SIP) in Internet protocol multimedia subsystem (IMS) architectures, such as mobile IM services that link text with voice, video and pictures.

The short message service centers (SMSCs) currently deployed by many operators are designed to handle the relatively predictable traffic of simple text messaging but are not optimized to support the large spikes in signaling traffic created by SMS applications such as tele-voting. The TekMedia SMS Solution can replace or "cap and grow" SMSCs, allowing operators to increase capacity and capabilities incrementally or to create a completely new SMS solution.

The TekMedia SMS Solution is part of Tekelec's "IMS-Ready" portfolio for next-generation networks and applications. The solution is in trials with several operators and will be available commercially in third quarter 2006.

http://www.tekelec.com

SIP Expert Covergence Secures $15 Million in Series C Funding

By Erik Linask

Covergence (news - alert) offers solutions for scaling, securing, and controlling VoIP and other SIP-based real-time services. Lately, though, the company executives have focused their energies on securing a new round of Series C funding — which it has now secured in the form of a $15 million investment by Globespan Capital Partners, along with current investors Highland Capital Partners and North Bridge Venture Partners.

Covergence offers its customers confidence that their real-time communications have the same security, reliability, performance and quality to which they have become accustomed. During that past year — buttressed by the release of its flagship product, Eclipse, in October 2005 — that confidence has paved the way for the tremendous growth Covergence has experienced.

The company's single-minded focus on making real-time communication a reality for its customers has paid dividends in the way of both customer and partner growth, including customer wins like Vonage and New York Times Digital Media, and strategic partnerships and certification agreements with market leaders including McAfee, Centillium, NETGEAR, Broadsoft, and CounterPath.

Now, that success also evidences itself in investment growth as well. This most recent investment elevates the Company's total funding to $31M since inception and provides Covergence the resources it needs to develop new sales and marketing strategies and geographically expand further into international markets.

"When we founded Covergence, we knew that the key to unlocking the potential of VoIP and IP-based communications lay in securing SIP and the edge of the network. Our success demonstrates that we hit the VoIP security sweet spot. Our investors' confidence in our strategy and team is evident by their participation in our third round of funding," said Bob O'Neil, president and CEO of Covergence.

http://www.covergence.com

Motorola Wireless VoIP Gateway Broadens Home Connectivity Options

Building on its heritage of delivering innovative, multifunction devices that re-define home connectivity and communication, Motorola (news - alert)introduced a new wireless gateway that combines voice, data, and network functionality for DSL users. The Motorola Home Hub (HH) 1620 DSL VoIP Wireless Gateway enables carriers to offer high-speed data and VoIP services to consumers, while simplifying installation and enhancing security.

The Motorola HH1620 combines powerful communication features with advanced security and simplified installation. The product includes an 802.11b/g wireless access point, a four-port router, a USB 2.0 Host Connection, a USB 1.1 slave connection, as well as a Session Initiation Protocol (SIP)-based VoIP adapter.

This technology allows the HH1620 to run consumers' home networks — wired and wireless — and power standard telephones with VoIP service. The Motorola HH1620 uses industry standard SIP signaling protocols to provide an all-in-one solution — offering users advanced ADSL2/2+ technology for higher speeds and longer reach for their high-speed Internet access.

"Consumers today want a single device to integrate multiple functions — from telephony to Internet access — and they want it to be easy to install and use," said Charles Dougherty, Motorola corporate vice president and general manager, Connected Home Solutions. "This latest Motorola gateway enables carriers to offer their DSL customers a single device that addresses multiple communication requirements and eliminates the need for additional routers, hubs, and wireless access points."

http://www.motorola.com

Ubiquity Software Releases Version 7.0 of its SIP A/S

Ubiquity Software, (news - alert) creators of standards-based SIP (Session Initiation Protocol) deployment platforms, announced the release of version 7.0 of its SIP Application Server (SIP A/S). The newest release of the high-performance, carrier-grade platform offers significantly broader capabilities, including support for a new Service Oriented Architecture (SOA) that enables carriers to develop, deploy, and manage innovative converged services that blend voice, video, data, and presence.

Ubiquity's SIP A/S utilizes a new Service-Oriented Object Framework (SOOF), which Ubiquity has packaged as the Appcelerator SOOF 1.0 Feature Pack and Developer Kit. The SOOF architecture extends the capabilities of the Application Server to provide a more richly featured, structured and controlled environment for SIP application creation and deployment. SOOF defines a new concept for developing telecom applications that are strongly aligned with SOA, which eliminates the normal pain of application development and opens new marketplace opportunities for added competitive advantage and profitability.

Compliant with the latest IETF SIP specifications and key extensions, the Ubiquity SIP Application Server is a high performance SIP implementation and provides multiple API layers for optimal control and flexibility when developing and deploying innovative products and services. It is well suited for developing proxy servers, presence servers, general events servers, B2BUAs, registrars, redirect servers, soft switches, call session control functions (CSCFs), IP PBX, and application servers including conference servers, instant messaging chat rooms, third-party control servers.

Ubiquity's SIP A/S is an open, high-performance, standards-based service creation platform designed specifically for telecommunications applications to allow carriers to offer innovative voice, data, and video services. The SIP A/S offers a programmable, horizontal platform that makes it easy to add new features and applications to carrier networks.

http://ww.ubiquitysoftware.com

West Corporation Deploys Convedia Media Servers

Convedia Corporation (news - alert)announced a comprehensive supply agreement with West Corporation,(news - alert)a premier provider of outsourced communication solutions. Convedia's (news - alert)CMS-6000 and CMS-1000 Media Servers will be deployed by InterCall, a subsidiary of West Corporation and the largest conferencing service provider in the world, to deliver industry-leading economics in the IP media processing infrastructure essential for InterCall's VoIP audio conferencing and collaboration services, while providing a versatile, reusable platform for West Corporation's next-generation IP services.

Convedia Media Servers will deliver scalable, IP Multimedia Subsystem (IMS)-compliant media processing infrastructure for West/InterCall's IP-based services architecture. Advanced reservationless conferencing and collaboration services developed by West Corporation's application development teams will use Session Initiation Protocol (SIP) and Media Server Markup Language (MSML) protocols to integrate and control Convedia media servers, which will provide audio bridging, transcoding, and personalized multimedia mixing capabilities, with superior economics compared to existing TDM-based audio conferencing equipment.

"Large audio conferencing providers are increasingly turning to VoIP technology to drive service innovation and improved network economics, while still adhering to the highest levels of network reliability and scalability," said Marc Beattie, Partner & CSP Practice Manager, Wainhouse Research. "Companies like West/InterCall are capping their investment in traditional conferencing bridge technology, and are shifting the majority of their new capital purchases towards IMS-compliant technology and IP services. This announcement clearly validates this trend, and demonstrates Convedia's market leadership as the defacto choice for carrier-class IP media processing platforms."

http://www.west.com
http://www.intercall.com
http://www.convedia.com

Allied Telesis Enhances iMAP with SIP Telephony

Allied Telesis (formerly Allied Telesyn), (news - alert) a global provider of secure Ethernet/IP access solutions and IP Triple Play networks over copper and fiber infrastructure, announced SIP support for its iMAP integrated Multiservice Access Platform voice services. Allied Telesis continues to expand the iMAP's range of capabilities, most recently by adding DS3 and ESA (emergency standalone) service, and now with SIP POTS.

The new SIP-enabled POTS service module on the iMAP supports service providers as they move away from the original TDM-based voice network towards a faster, more scalable VoIP implementation. SIP has gained in popularity as a standard for real-time communications over IP due to its scalability, speed, and ease of implementation. Because SIP uses intelligence distributed throughout the network, it can more efficiently and accurately connect a multitude of calls, eliminating demanding processes that previously were done only at the network's core.

The iMAP offers system flexibility through the ability to mix and match different networking technologies within one platform. It accommodates both fiber and copper-based deployments and supports a wide range of services including SIP POTS, MGCP, xDSL, DS3, T1, Ethernet, active fiber, GEPON FTTx technologies, Emergency Stand Alone (ESA) and 10G backhaul. Ideal for new access deployments and upgrades, the iMAP features a modular chassis design that simultaneously supports different applications from access, transport, and network aggregation. The environmentally hardened iMAP is RUS-accepted and NEBS-certified.

http://www.alliedtelesis.com

Seiko Epson Calls on Mitel for IP Communications

Mitel, (news - alert)provider of IP communication applications and solutions for growing businesses and enterprises, has been selected by tech giant Seiko Epson to deliver networked IP business communications at three South American subsidiaries. Comprised of the flexible and highly scalable Mitel 3300 IP Communications Platform (ICP) and integrated business applications, the solution will be deployed in Argentina, Chile, and Peru.

Mitel's proven IP communications solutions were identified as the best fit for these branches of an organization dedicated to customer service and technological excellence. The units of Epson selected Mitel independently after a lengthy and careful analysis of several leading vendors before concluding that Mitel best met their current needs and offered the most forward looking technology solution.

A crucial consideration in the selection process was Mitel's advanced adoption of and commitment to Session Initiation Protocol (SIP). SIP is an open international communications standard that allows for interoperability and easy deployment of services in much the same way that hypertext mark-up language (HTML) redefined the Internet and helped create the World Wide Web. The 3300 ICP provides native support of SIP to enable users to leverage the power of the Internet to deliver a whole range of business processes and potential that delivers a host of benefits. Mitel phones support both SIP and Mitel IP (MiNET) protocols on a single hardware platform.

With this technology in place, Epson Argentina can not only link its exchanges locally to achieve seamless integration of all its communications, but can also extend this connectivity to other countries. Since Epson Chile chose the same vendor makes it easy to combine resources so that the two subsidiaries in different countries effectively share a single exchange.

http://www.mitel.com

ON THE EDGE

SBCs – The End is Near!

by
Erik Lagerway

Session Border Controllers (SBCs) are, to some, a saving grace and, to others, the devil's breath. The reason, I think, is because the SBC can solve quite a few problems, but, if not used appropriately, core network costs can easily double for a business. They also have a tendency to flood the network with requests when coming back to life after a failure, which causes a host of other problems.

Because I, myself, will ultimately have to answer for escalating network costs, I am inclined to do just about anything to stay away from SBCs. The sad part is that it is nearly impossible to get around using an SBC entirely in today's world of VoIP, Video, and IM — here's why.

NAT traversal is a nasty business and, unless your plans are to operate only in managed private IP space, you will need a mechanism to get off those pesky firewalls. For mobile users, this is a common issue and asking the network admin at any given hotel, for instance, to open up ports on his firewall is nothing short of a pipe dream, especially when you tell them that you plan on bypassing their telephone system and the 500% up-charge on long distance by making calls using your softphone over the broadband connection you paid $10 for. Yes, SBCs are useful here and there is nothing special the user has to do to take advantage of it; it's provided as an invisible service to the end user.

So, we need SBCs today to solve some basic NAT traversal problems, but should we be routing all traffic over SBCs? Is this necessary? No, in my opinion it is not, but, if you are short some good network security engineers and time, you may feel it's easier to go the SBC route. Hardening firewalls and SIPifying network security is not for the faint of heart. It takes experience, but if you have those resources, you can probably save yourself a wheelbarrow full of cash by not using an SBC for this piece.

What about SIP Peering? Many people say that, unless you are using SBCs, you will not likely gain access to many SIP trunk partners and carriers. I am not so sure about this. It's usually your own network with which you are most concerned and, as long as you are confident the holes are plugged, you should be able to peer with most providers offering the service. So, if the pitch is to use an SBC to provide more security for your core SIP network, I say this argument can be more of a resource driven issue.

In the end, it's really up to you, as the network architect, to decide if you are gaining enough for the 50-70% increase in cost, as opposed to what you would pay if you were to spend a few more dollars on the firewalls in your network.

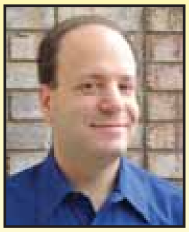
SBCs can easily cost around $20 per initial session — at a minimum of 1,000 sessions, you are looking at $200k. Multiply that for every 1,000 sessions you need to grow your network and the number gets very high very quickly. I can't speak for you, but I am confident I can put that money to better use elsewhere in my network.

For me, the SBC is a necessary evil, but I also have my ear pasted to the rails. The first commercial TURN server is right around the corner. Couple TURN & STUN servers with ICE, and the appeal of SBCs (and the Back-to-Back User Agent or B2BUA) quickly diminishes. Bring on open standards!

Erik Lagerway is CTO at Shift Networks. (news - alert) *For more information, please visit the company online at http://www.shiftnetworks.com.*

SPEAKING SIP

Is VoIP Secure?



by

JD Rosenberg

Undoubtedly, by the time you've read this, you would have heard about the VoIP "crime of the century." I'm referring to an incident in early June where a man was arrested for allegedly stealing minutes from numerous VoIP service providers over a period of a year and a half. The individual allegedly sold VoIP service to his customers and, rather than purchasing termination and origination services, found weaknesses in the networks of several VoIP providers and routed the calls illegally through them.

Certainly this is not the first time that there have been security weaknesses exposed in VoIP. (define - news - alert) In the summer of 2004, the big news was a bunch of folks that had used the Asterisk open source IP PBX to generate faked caller IDs, which were then routed into the Public Switched Telephone Network (PSTN).

Every time one of these incidents happens, there are invariably questions raised about the fundamental security of VoIP. Is VoIP secure? Is it as secure as the telephone network?

The short answer is that VoIP can be very secure — more secure than the PSTN, in fact. Unfortunately, the problem is the 'can be' part of that statement. The protocols behind VoIP, and SIP in particular, have been designed with extensive security measures, which have only improved over time. Unfortunately, only a small number of these have been implemented, and fewer still deployed. As such, weaknesses show up because vendors and providers choose not to implement the tools that can cover those weaknesses. Sometimes it takes attacks, like the ones we saw this past week, to drive providers to ask for these features, which in turn causes vendors to implement them, and ultimately results in deployment.

That said, what kind of security could SIP afford? Could it have prevented the theft of minutes and caller-ID spoofing that got press attention? The answer is yes. Absolutely.

The theft of minutes could have been prevented with SIP's mutual Transport Layer Security (TLS) authentication feature. This feature allows two proxies to establish a secure link between each other, and using cryptographic techniques, securely determine the identity of the other side. If a termination provider, such as the ones who had minutes stolen, only accepts or sends calls over SIP links that are secured with TLS, every single call can be securely traced to a particular customer. This feature is actually mandatory to implement in order to be formally compliant to the SIP RFC (RFC 3261).

What about caller ID spoofing? Can it be prevented with SIP? Interestingly, the answer is that it depends. SIP supports a broad range of identifiers for users. These include telephone numbers and e-mail-style identifiers (such as sip:user@example.com). When e-mail-style identifiers are used with SIP, it is possible to prevent users in one domain from spoofing calls from another. Consequently, a call from sip:joe@example.com could never be spoofed to look like it came from sip:george@whitehouse.gov. This is done using a SIP extension called "SIP Identity," which has been completed in IETF but has not yet been assigned its RFC number. The weakness in the mechanism, somewhat ironically, is the good old PSTN. With phone numbers, this mechanism doesn't work quite as well.

VoIP security is much more than preventing these two types of attacks, of course. There are numerous threats against VoIP systems, with fraud and faked caller ID being two of the most basic. Far more complex are denial-of-service attacks, and in particular, ones that actually try and use SIP to launch attacks. One of my favorites is an attack I call the "voice hammer." In this attack, an attacker can, by sending just a few dozen SIP requests to a server, cause any target host on the Internet to receive a flood of voice packets at hundreds of kilobits per second to megabits per second, depending on the voice codecs in use. This attack requires protocol extensions to fix. This attack in particular (which is possible with almost any VoIP protocol — it's not specific to SIP) is prevented using a mechanism called Interactive Connectivity Establishment, or ICE. Many may be familiar with ICE as a NAT traversal technique. It has an interesting side effect of preventing this particular attack, and this benefit is described in some detail in the ICE specification.

The lesson from all of this is that there are lots of potential threats out there. However, the good news is that many of these attacks, such as theft of service, are totally preventable as long as providers of SIP services actually use the tools that have been designed to deal with them. I'm almost happy that this recent security issue got so much press. Hopefully it means that people will think twice before running a service without these security features enabled.

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)*(http://www.cisco.com)*

Value-driven Communications Solutions

Inter-Tel provides converged voice and data business communications systems and applications for the small, medium and enterprise business markets.

- **Designs, engineers, sells and installs technologically advanced communications systems including SIP-based solutions**
- **Enables investment protection through a commitment to design architecture with open standards, scalable deployment options and migration opportunities, giving the customer choice**
- **Develops SIP-based applications designed to address operational performance, improve business processes and deliver ROI**
- **Provides a complete portfolio of SIP-based Presence Management solutions, and Collaboration and Messaging applications designed to link departmental resources into a single, cohesive, cost-effective organization**
- **Offers provisioning and facilities management, professional services, and custom development support through the Inter-Tel Managed Services program**

To locate an Inter-Tel Authorized Provider near you, visit **www.inter-tel.com**

FAST FACT...

Over 35 Years of Focused Commitment in Business Communications.

SPECIAL FOCUS

Speaking with Aculab's Ian Colville

Ian Colville

Recently, *SIP Magazine* had the opportunity to speak with Ian Colville, product manager at Aculab, about the future of SIP and, more specifically, Aculab's focus on SIP. Here are Ian's comments.

What does SIP mean for Aculab?

Aculab (news - alert)has always had a strong focus and expertise on communications protocols. Indeed, that has been a key part of its strength over the years. Now, when next-generation networks are a reality (perhaps we should call them 'current generation') and communications means far more than simple voice calls, we have a definite focus on SIP. Aculab offers SIP, under its standard cost free licence model, for use with the award winning Prosody X media processing resource card. GroomerII makes full use of SIP in its guise as a media gateway and SIP developments are ongoing, particularly in the areas of security and interoperability.

Please describe some of your more recent development efforts, with regard to SIP.

SIP is a fundamental component of Aculab's enabling technology solutions portfolio for developers and systems integrators. SIP Bridge developed the concept of separating media processing from session handling to present developers with many advantages. Having revolutionized telephony board design with Prosody X, we also got rid of the 1:1 allocation of media resources for each call (session) inherent in previous architectures. Now, there can be far more efficient allocation of those resources, which gives customers a great deal, in terms of cost effectiveness. Third-party call control (IETF RFC 3725) is also possible, with Aculab's integrated SIP stack being used effectively as a back-to-back user agent. The Aculab SIP Bridge becomes a powerful and cost-effective way to build complex IP contact center and IP PBX type products with rich media and call control features that were simply not possible with TDM trunks and protocols.

Capabilities introduced include: offer-answer model (IETF RFC 3264); fine control of SDP content; MRCPv2; API presentation of raw SDP; mid-call signalling (e.g., INFO, NOTIFY); blind transfer; custom headers and message bodies (e.g., ISUP, MIME) in setup and mid-call request; symmetric SIP; and synchronous RTP. Furthermore, current projects are underway to include SIP security, NAT traversal and resilience features.

What are some of the challenges facing the SIP community and, conversely what are some of the brighter opportunities (security, interoperability, call/voice quality...)?

I would suggest there isn't much of an argument against SIP becoming the de facto protocol for setting up and controlling voice and multimedia services across next-generation IP links. However, in that scenario, I'd guess the main challenge is living up to the promise of SIP — attaining the Promised Land of SIP, if you like. That is the concept of SIP as a single protocol for all communications between any and all classes and types of device. In other words, SIP becomes a single ubiquitous protocol for the entire network.

Part of the challenge is that the PSTN is still in existence. Despite the exponential rise in downloads of Skype and other clients, increases in users of services from the likes of Vonage, and new networks, such as Global Crossing and Level 3, the balance of teledensity is still tipped in favor of the PSTN. The number of users of next-generation communications will continue to proliferate, but even those of us making use of new ways to communicate still need to communicate via the PSTN. So, gateways and interoperability are a key part of the solution.

If SIP is going to replace the plethora of national protocols that exist, it's going to need all of the functionality of its predecessors. Everyone will expect SIP to do the job, the role, the task, that the legacy protocol did for them. This is part of the necessary evolution of SIP. However, even today, there are lots of PBX-type features that are available in a SIP environment.

Opportunely, SIP extends far beyond telephony, as voice is simply one of many applications. SIP enables a rich portfolio of communications services, such as presence, which provides for vastly increased efficiency, enhanced productivity, and time savings. It's been said before, but presence may well be the dial tone of the 21st Century.

Security is important, as are interoperability and the ability to work through firewalls and NAT scenarios. There are a number of extensions, such as STUN, TURN, and my personal favorite, ICE, which is really cool!

In your opinion, what does the future hold for the SIP standard?

Undoubtedly, the flexibility and extensibility of SIP is one of its main assets and there is so much that is possible with SIP. It is an enabler for new ways of communicating and we've only just scratched the surface. There are some competitors for SIP, of course, notably, H.323, which is still hanging in there, although I think advocates like Avaya and Cisco may be seeing the light, and the likes of Skype and IAX2. The difficulty with Skype is that it is proprietary and it does not look like it will be adopted for mass use at the telco level. Looking forward, there is a lot of force behind SIP and, in the UK, BT has announced that its 21st Century Network will run on SIP — using gateways to connect to legacy systems at the edge of the network, where needed. With an institution of this scale laying a SIP card on the table, we must surely accept that SIP looks like it is here for the long haul.

Three things you need to run a successful small business:

An original idea

Passionate commitment

A really great phone system

TALKSWITCH IS A FOUNDING MEMBER OF SIP CONNECT

THIS IS YOUR PHONE SYSTEM

Resellers love TalkSwitch because it's easy to sell and easy to install. Customers love TalkSwitch because it's easy to use and affordable.

TalkSwitch systems start as low as US $695 (msrp). VoIP-equipped systems start at US $1795 (msrp).

Call toll-free 1.888.332.9322 x301, or visit us at www.talkswitch.com to find out more about our systems or our reseller program.

talkswitch®
ANSWERS WITH INTELLIGENCE

THE AWARD-WINNING TELEPHONE SYSTEM FOR SMALL AND MULTI-LOCATION OFFICES

SPECIAL FOCUS

The Third Annual VoIP Developer Conference:

The Race Is On

By Erik Linask

VoIP (define - news - alert)is changing the way the world communicates and the world is taking note. If 2004 was the year that VoIP gained recognition as a viable alternative to circuit-switched telephony; and if 2005 was the year when deployments — by both consumers and businesses — experienced marked growth; then 2006 is the year when new applications will drive fence-sitting enterprises to choose VoIP over traditional telecom.

If you have any intention of seizing what is, undoubtedly, the most promising telecom opportunity of the past several decades, then the VoIP Developer Conference in Santa Clara, August 8-10, should be on your calendar.

The growth and progression of VoIP and IP communications markets over the past 24 months is creating a remarkable opportunity for companies and individuals designing and building applications for broadband telephony networks, both wireline and wireless. As deployments increase, service providers are in a race to offer the most attractive services. In order to do so, they must cultivate relationships with developers who can help them add the most compelling services to their networks.

The Third Annual VoIP Developer Conference is the only chance this year to learn, in a focused environment, how to become on of the first to quickly develop an entire spectrum of new VoIP applications and products that are in high demand: video over IP, SIP, cable telephony, wireless/WiFi telephony, software development, and hardware/device development, and more.

At the Conference, you will have the chance to learn how best to leverage the development tools available today. You will become educated about today's hottest topics, including VoIP software development, VoIP hardware development, VoIP security, wireless VoIP development, IMS development, SIP development, chip-level VoIP development, host media processing, and much more. And, you will have the opportunity to see live, hands-on demonstrations of products from industry leaders.

Education

As always, VoIP Developer Conference boasts a full schedule of informative conference sessions. New to this year's Conference curriculum are tracks focusing on Security and IMS (IP Multimedia Subsystem) Development. The agenda also includes four of the most popular educational tracks from past events — Hardware, Software, SIP Development, and a special track on Wireless applications — each featuring new and expanded content.

As with all TMC events, presenters in sessions are strictly forbidden from delivering company pitches or 'industry perspectives.' This ensures you get an unbiased education about emerging trends and markets, like IMS, SIP, open source, new consumer electronics applications and devices, and other technologies creating huge development opportunities. What you will not get is a vendor-focused sales pitch. Instead, you will get the maximum amount of information to help your company forge ahead.

"The strength of this event is, and always will be, the in-depth, cutting edge educational content," said Rich Tehrani, TMC President and chairman of VoIP Developer Conference. "Our team of editors spent months assembling the most relevant curriculum that will be taught by the highest caliber faculty. I honestly believe VoIP Developer's educational content is the best available and is the strength of this one of a kind event."

Indeed, each session topic and presenter is hand selected by TMC, from literally hundreds of submissions. Only the most relevant sessions presented by speakers that are certain to deliver critical information in an intelligible manner make it on to the program.

Keynotes

Attendees also will be treated to keynote addresses delivered by industry veterans from companies that are firmly entrenched in the VoIP industry. These speakers each have extensive experience in the industry and are uniquely qualified to impart their thoughts to VoIP Developer Conference attendees on why this opportunity is far greater than any in the telecom industry's century-long history.

This year's featured keynoters include:

Ben Rabinowitz — Vice President of Marketing and Sales, AudioCodes

Fred Zimmerman — Executive Director, Customer Premises Solutions Packet Voice and Video Business Unit, Texas Instruments

Lawrence Byrd — Director of IP Telephony and Mobility Solutions, Avaya

David Mandelstam — President/CEO, Sangoma Technologies

Michael Stanford — Director of VoIP Strategy, Digital Enterprise Group, Intel

Kevin P. Fleming — Senior Software Engineer, Digium

Vendors

In addition to the terrific educational sessions, the program for VoIP Developer Conference includes opportunities to see, test, compare, and select today's most exciting IP Communications building blocks. You'll be able to see these components in action, and get answers to your questions about them.

Day Two of the Conference includes a series of concurrent Product Showcase sessions, where individual vendors will have 20 minutes to explain how their building blocks will help bring your next application/product to market more quickly. This is the one chance outside the exhibit hall vendors have to convey the specific benefits of their own development tools.

Speaking of the exhibit hall, what would the VoIP Developer Conference be without them? Between sessions, you can meet with industry leading vendors and your potential partner, each of whom is ready to help you build successful VoIP applications.

Throughout the event, you can visit the exhibit area, where vendors will have their tools on display and are prepared to describe the merits of their products and solutions and answer any questions you may have. There will be ample time to visit each vendor, ask questions, view demos, and gather the information you need to intelligently choose the correct tools.

Networking

Also, since this is the only conference in 2006 focused solely on VoIP development, it is, likewise, your only opportunity to meet and talk with other developers to share ideas, exchange business cards, and discuss the virtues of one vendor's development tools over another's. This is you unmatched opportunity to form valuable partnerships with OEMs, service providers, and your peers in the VoIP development community.

The Place to Be

VoIP is clearly the world's most important communications technology right now. VoIP applications are revolutionizing the way businesses, consumers, and government agencies communicate on a daily basis — and the best part is, this is only at the beginning. With SIP and open source, communications solutions never been so open, so flexible, and so customizable. The opportunities IP telephony brings — for the end user, the service provider, and the developer — continue to multiply as VoIP applications pervade service provider networks, offices, mobile devices, and homes at an astounding rate.

"The VoIP industry continues to explode as enterprise and residential users recognize the benefits of IP telephony as more than just cost savings over traditional phones, "said Fred Zimmerman, senior director, Customer Premises Solutions, in Texas Instruments' Communications Infrastructure and Voice Business. "TMC's VoIP Developer Conference provides a unique opportunity to bring together many of the companies and developers behind the VoIP industry to meet with and consult each other about the requirements of this explosive growth, as well as foster new applications."

The bottom line is that VoIP Developer Conference is the only event focusing on the needs of both companies and individuals creating new applications for broadband telephony networks. With three days of educational sessions, keynote speakers, networking receptions, and an exhibit hall full of exhibiting companies, it is your best opportunity to meet with colleagues from across the industry to learn about and discuss the present and the future of the VoIP industry.

This is the perfect venue to compare developer programs. Serious developers must come to this show to see what sorts of developer programs exhibitors are pushing and what incentives there are to develop on various platforms. This event is the perfect place to evaluate the tremendous opportunities available today.

It is also where the race to develop the next great consumer application can be won. WiFi telephony and dual mode phones are becoming ubiquitous. Here you will be able to gather the tools to be the first to develop mobile applications that enterprise customers and consumers alike will spend good money for.

If you intend on competing in that race, don't miss the biggest and most important VoIP Developer event of the year. Getting to the VoIP Developer Conference is easy: The Hyatt Regency Santa Clara is convenient and easily accessible, just a few miles from San Jose International Airport, in the heart of Silicon Valley.

For more information or to register for the third annual VoIP Developer Conference, please visit http://www.voipdeveloper.com. If your company is interested in exhibiting at the show, please contact Dave Rodriguez: drodriguez@tmcnet.com or 203.852.6800 x146.

Exposing the Hidden Costs and Gotcha's of VoIP

By Jon R. Doyle

Why is it that companies often rush into VoIP, only to be disappointed by the results? Perhaps, it's because no one can agree on what VoIP really is. Is it a consumer-grade, closed network like Skype? Is it low-cost access to the PSTN network, so that you can save a dime on a call to Milan from San Francisco? Or, is it best defined as the use of digitized signalling and voice packets and the applications that make it work, like SIP proxies or voicemail?

This confusion appears to be not as prevalent outside the United States. France, for example, is a heavy adopter of VoIP, where nearly 33% of all subscribers use some form of digital voice. The phone system has been digital for over 15 years and the quality is crystal clear and extremely reliable.

That is a far cry from the cheap PSTN-SIP gateways, poorly designed soft clients, and low quality SIP proxies and services we still see in the U.S. and Latin America. What is it going to take to really get us to the consistent levels of service seen in countries like France? Maybe we will leapfrog Europe when the carriers and enterprises get serious and stop playing around with consumer products.

Meanwhile, these differing perceptions of VoIP and the "raw" state of the VoIP offerings available today are causing confusion and dissatisfaction among buyers. When CommuniGate Systems conducted a survey of CIOs and CTOs at enterprises and carriers, the findings revealed that 60% believe it is difficult to evaluate the true costs of VoIP and that the returns on the investment are unknown. It seems many people have no idea what they are implementing and are often taken by surprise when the hidden costs of a VoIP deployment offset the potential savings or revenues. This is clearly unacceptable in a world where accounting for IT spending is of utmost importance and where IT budgets are slashed routinely.

It is the intention of this article to bring some clarity by helping organizations planning a VoIP implementation to navigate their way through the most common hidden costs and pitfalls of VoIP.

Pitfall #1: What's on your feature 'wish list'?

What do you really want? Quality? Wideband? Secure access from everywhere? Do you lease a line and have QoS to your gateway? If it's only to call a friend in Jerusalem a couple times a month, get one of those providers that are all over the ads and doing IPOs these days. If you want business-class telephony, add that to the top of your requirements list, as it will affect all of your decisions, in terms of purchasing and configuration of the architecture. In fact, you might think to outsource it all, because design of a strong and high-quality VoIP infrastructure is not a simple matter.

Many of the features VoIP promises, such as voice mail integrated into email and conferencing, are not included with standard VoIP systems. Many small business offerings don't even provide voice mail as part of the standard cost.

The most productive features of VoIP are the "touch points" to all forms of IP-based communications, where voice and video touch scheduling, email, and IM. Imagine a system that calls you for conferences instead of relying on PIN codes and calendar reminders. Imagine being in a hotel and arranging a wake-up call by sending a text message to the server with a time and address to call.

To avoid pitfall number one, consider exactly which of these unified communications or messaging features your business, or your customers, need. Look closely at what your vendor means by unified messaging, then factor in any additional costs for features not included.

Pitfall #2: The true cost of security

Security is a must with VoIP implementations and it's often not included as part of an installation, since VoIP system providers tend not to be in the security business. It should be included, though — security can add an extra 40% to the total cost, just in session border controllers for NAT traversal and flow control, as well as securing the media channel. The other major "gotcha" is that many of the systems being designed today are offerings from legacy telephony vendors that have little expertise with DDoS or other techniques, which can cripple systems exposed to the Internet.

Pitfall #3: Ensuring your VoIP implementation

Hardware for backup or failover is also required, but shouldn't result in doubling of hardware costs. Hardware needs to be deployed intelligently, so it's as cost-effective as possible. Voice communications need to retain tone quality and only carrier-grade solutions offer this. Products also need to be specifically designed for clustering or redundancy.

Be sure that the system designed for your VoIP deployment has capabilities for SIP load balancing and that you can add or remove hardware or update software without taking the entire system offline.

Pitfall #4: Keep an eye on the big 'I's': Implementation and Integration

Naturally, speed of implementation is important, and it doesn't have to be a lengthy process. A good-sized VoIP system can be deployed in as little as a week, providing appropriate evaluation is done in advance. Inquire about and ensure that contingency plans are in place for bad migrations.

Consider how the solution integrates with other external systems. Does it have open, standards-based APIs? Are there any development tools or capabilities to customize the product to suit your needs? Don't get trapped into a closed system or one that requires the vendor to charge $400 an hour to add an extension or greeting to your IP PBX.

Pitfall #5: The domino effect

It's also important to consider how other systems will be affected by the VoIP implementation. Will your legacy systems need upgrading or replacing? More significantly, can cost efficiencies be gained by integrating the new voice system with the existing collaboration or messaging platform — combining the three functions into one communications platform?

Pitfall #6: Underestimating scalability requirements

Insufficient scalability can drive up costs of hardware, power, and IT real estate to unexpected heights as organizations roll out VoIP to larger numbers of workers. Many VoIP systems have not yet been proven in large-scale deployments, and some initial implementations have required 80% more servers than a data network supporting the same user base. The key to smart business growth is to add system and storage resources in conjunction with revenue growth and user demands.

Conclusion

Many of these considerations are routine for any technology evaluation. But the fact is, with different vendors focusing on different parts of the whole VoIP picture — and with little incentive for them to coordinate efforts — it is exceptionally tricky for organizations to build a true assessment of the real cost of VoIP and, consequently, for them to truly appreciate the potential savings or revenues.

By asking the right questions of vendors and learning about the real benefits of productivity in VoIP application servers, CIOs should be able to cut through the confusion and root out any nasty surprises before it's too late.

Fortunately, VoIP is starting to get a makeover and we can expect more focus on reliable, secure, business-class VoIP in the months ahead. Internet-based VoIP will replace the current telephony business model — and organizations will have the metrics to deploy VoIP successfully. It won't be long before we catch up with, or overtake, European standards. Building a globally distributed VoIP network using standards is possible. I just wish that we had the DSL they have in France right now, so we could receive voice and video calls through our TVs or stream multi-party video to every room in the house.

Jon R. Doyle is Vice President of Business Development at CommuniGate Systems. (news - alert) *For more information, please visit the company online at http://www.communigate.com.*

Controlled Connectivity: Interfacing Enterprise IM Systems with External Domains

By Christian Stegh
& Alan Johnston

Some 75% of companies use instant messaging, according to Nemertes Research. IM's popularity is understandable: presence-enabled real-time communication with associates and friends is efficient and easy to use. As its popularity exploded in the consumer market, workers began bringing the technology to the office on a multitude of IM clients. IM's wide use within enterprises comes in spite of the lack of a unifying standard and cross-client interoperability. New approaches to standards and federations between enterprises and outside domains seek to solve these challenges, and are the focus of this column.

The Case for Controlled Connectivity:

IM in the enterprise is typically supplied using one of two models:

- Users log on to a public IM service (like AIM, MSN). In this case, the IT team has little to do with supporting the service, aside for allowing it to pass through the network.
- IT provides an internal service of some sort (SameTime, Jabber). In this model, an IM system is deployed within the intranet, and a corresponding client is deployed on users' desktops.

The first approach typically means that the IT department either doesn't mind or doesn't control user access, and that there is more or less wide open access. This approach has its pitfalls; security risks and lack of IT control are at the forefront. Client vulnerabilities, an alternate conduit for spam, viruses, identity theft and loss of privacy are most common concerns. And with good reason, considering that December 2005's instant message exploits jumped 826 percent over December, 2004 according to IMlogic's Threat Center. As such, enterprises are getting more serious about banning the use of public IM services. Yet attempts to lock down IM are difficult, since many common IM platforms use open ports (like port 80) if their default ports are blocked at the firewall.

The second approach, in which IT has control of an internal IM system, has its benefits:

- Reduced virus risks
- Tracking and logging
- Encryption
- Control of use/access

But it's isolated – each enterprise is an island unto itself with limited or no external IM connections to partners, clients, and the rest of the outside world, isolating a company and its employees.

The Best of Both Worlds

Enterprises can leverage the best of both worlds — controlled connectivity — by systematically interconnecting their internal IM system with other

·VONEXUS·

Microsoft®-based IP Communications

It's time to reach beyond business as usual.

Make your enterprise an interaction center.

VONEXUS | Microsoft®-based IP Communications

Vonexus offers the only complete Microsoft-based IP Telephony solution for business. An IP PBX and more for converged communications to grow your business. One reliable solution. From the leader in Microsoft-based IP telephony. Vonexus.

www.vonexus.com

Don't miss
New Rules for Business Communications Luncheon
Wed., Jan 25, 12:00 pm
At INTERNET TELEPHONY Conference & EXPO
Visit www.vonexus.com for more information

enterprises and public IM services. While bilateral peering (one company to another) is possible, it is not scalable. If a health care insurance company wanted to peer with all of its providers, for instance, a team of engineers would be needed to manage the policy and technology. So, what's the happy medium? Something to efficiently interconnect the domains is needed.

Enter "federated services," a way for enterprises that need to work closely with partners, hosted suppliers, and customers, to interconnect their enterprise VoIP networks, federate enterprise IM systems, and interoperate with the public IM networks. This is an emerging business segment in the convergence landscape. A service provider works with the enterprise to develop a peering policy, then handles the complex policy implementation, process work, and protocol exchange. The result is a collaborative, yet secure environment, since enterprises can still control with whom they want to federate. Of course, to federate, service providers must handle multiple IM protocols. A brief explanation of the common protocols used by public and enterprise IM systems is provided next.

Instant Messaging Standards

There are two primary approaches to Instant Messaging. One is based on a client-server model, the second based on SIP. While both strive to meet the requirements of IETF Request For Comment 2779 (an informational IETF document on IM/Presence Protocol requirements), they're different in philosophy and practice and are not natively compatible. Most public IM services use older proprietary protocols.

XMPP (Extensible Messaging and Presence Protocol), the first standard IM protocol to become popular, was born out of the instant messaging world. After being formed as the Jabber protocol in 1999, the base Jabber/XMPP protocols have since been approved by the Internet Engineering Task Force (IETF). XMPP is an XML streaming technology using client/server architecture. Like email, XMPP clients connect to servers and servers can connect to each other. Unlike email, though, XMPP doesn't have multiple hops between servers and has native security mechanisms, such as channel encryption and authentication. IM systems from Apple (in its Tiger OS), GAIM, Google, and Sun, to name a few, use XMPP.

SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) is the IETF working group responsible for one of the first major extensions to the core SIP standard. Since SIP inherently supports multiple media types, IM is simply another form of communication supported by the protocol, along with voice, video, gaming, and others. SIMPLE's base protocols have been approved by the IETF. Since SIP is inherently a peer-to-peer protocol, IM can be P2P, not passing through servers. Being a part of the SIP stack enables presence across other devices, like SIP phones, so a buddy list can include on-hook/off-hook status in addition to basic IM/desktop status. SIMPLE's inherent scalability and modularity within the broader SIP stack made it the choice of enterprise players like Avaya, IBM, and Microsoft.

Current State of the Standards

Both the core XMPP and SIP specifications are complete and published in the IETF as RFCs. However, significant work is being done on extensions to the base standards.

Extensions to XMPP are not being done in the IETF. Instead, they are being handled in the Jabber Software Foundation's JEP process and will never be published as RFCs. There are well over 100 JEPs listed at http://www.jabber.org/jeps/jeplist.shtml. XMPP extensions include the initial documentation for a set of multimedia extensions, called Jingle, released in late 2005.

The extensions to SIP for IM and Presence are not yet complete either; they are being developed in the IETF's SIMPLE Working Group. Many have been published as RFCs with others in various stages as Internet Drafts. The SIMPLE WG charter page lists about 20 Internet Drafts that will eventually be published as RFCs.

Interworking Between SIMPLE and XMPP

SIMPLE and XMPP systems have been deployed in enterprises and both will continue to be deployed in the near future. Gateway products have emerged out of the need to interwork between different IM systems. Gateways mainly transpose IM and desktop presence protocols, so that users with one client can see buddies from other IM systems. Behind the scenes, gateways manage protocol conversion, security, policy rules, and privileges, usually all on the same platform. When connected to SIP-enabled IP PBXs, some gateways can transpose the on-hook/off-hook presence of phones as well. In most cases, gateways are custom developed, often involving APIs, to interwork with each IM system.

Fortunately, basic interworking between the protocols is possible in a standardized way, by directly mapping addresses and presence subscriptions/notifications from one protocol to another for use in such gateways.

However, the different security models used by the two protocols have proven more difficult to merge. For example, Secure SIP requires the use of TLS transport end to end — the session must fail if this cannot be provided. It is unclear how to interwork a Secure SIP IM session with XMPP, which has no equivalent security mode. As a result, it is likely that federation services providing both protocol interworking and security and policy enforcement will be a viable service.

Advanced Features Call for Advanced Services

While the challenge of the base protocols working together might be solved by standards in the short term, more advanced features, security, and management will be value-added services of the federated model for the longer term.

For instance, the more complicated mapping of text chat rooms and conferencing has not yet been described in published documents. Privacy and manageability policies across borders will take significant care and feeding. While enterprise IM systems have some policy control built into them, in their current versions, they lack robust features to handle complex cross-company requests. While enterprise systems may be able to allow/deny based on general characteristics, it would be a coarse, rather than granular, set of options. For instance, Company A could peer with Company B, but, if A's recruiting department shouldn't be able to see the presence of B's coveted IT staff, today's enterprise systems might not block their view. Another example: two companies that are merging might want to allow IM only from executives and merger analysts, but not between line workers.

What about a Standard for Federation Itself?

Currently, no discussion is underway about standardizing how enterprises peer with their federated IM service providers, but there is good reason to begin them. Like SIP itself, there are options within SIMPLE that network engineers may employ differently, such as the duration of the subscription before a watcher marks a user "away" or "offline." Even mapping of graphical "emoticons" between IM systems needs to be defined. The SIP Forum's IP PBX / Service Provider Interop Task Group provides a precedent for creating such recommendations, since early deployments of SIP VoIP to service providers experienced implementation complexities. That said, the IM federation model is less complicated and should take less time, since there are fewer players in the enterprise IM space than in the VoIP space, and the interconnection requirements are more straightforward. Nonetheless, standards outlining authentication requirements, TLS rules, and so forth would speed adoption.

Moreover, agreements between federation providers would make a large-scale federated IM model more feasible. If Company A federates with Provider 1, Company B federates with Provider 2, and the two companies want to share IM and presence, then both would benefit if Providers 1 and 2 were federated. Otherwise, enterprises may end up peering with one another as the standards mature and their security and management tools improve.

Summary

Enterprises should consider the risks versus rewards of using public IM services and move towards a standards-based enterprise IM system as required by security and corporate policy. Using a system based on SIMPLE provides other integration benefits, like being able to view the on-hook/off-hook phone status presence of buddies using the enterprise IP PBX and making an informed decision on whether to call or IM.

While IM and its underlying standards are mature, the options for controlling connectivity between enterprises and the outside world are new. Federated service providers are leveraging standards to provide enterprises with options to securely interconnect with other enterprises, partners, and public IM networks. Instead of peering with one another, enterprises peer with the service provider, who handles any protocol conversion, implements the enterprise's desired policy, and manages security rules. This model should speed the time to IM federation; soon enterprises will benefit from exposing their IM systems to the outside world for customer contact center, supply chain, and other creative communication applications.

Christian Stegh co-authored early SIP protocols and is currently IP Telephony Practice Leader, Avaya, (quote - news - alert)North America. Alan Johnston is Chief Technical Advisor, Tello Corporation. (news - alert) For more information, please visit the companies online at http://www.avaya.com or http://www.tello.com.

SIP-Controlled Communication Services for SOAs

By Alan Rosenberg

Voice interactions are a common element of almost any business process. Whether designing a product, executing a financial transaction, or fulfilling a sales order, workers have a natural need to speak with associates, partners, and customers. Yet, most automated business processes offer little or no voice integration.

Suppose an aeronautics firm is working with a number of outside component suppliers in the design of a new aircraft. An extranet-based CAD program enables the manufacturer and its suppliers to work in concert. An aeronautical design engineer can review a component blueprint from a supplier, add comments, and reply to the supplier with the click of a mouse. But, if she wants to discuss some details of the design she has to look up the supplier's phone number, call the supplier, and risk a potential extended game of phone tag.

What if voice and presence capabilities were integrated into the CAD program? What if, while reviewing the blueprint, the engineer could determine if the supplier is online and available to speak, and then initiate a conversation with the supplier directly from the CAD program with the click of a button. If the supplier is not available, the application might have the capability to automatically connect the two parties when both are available.

The supplier agrees to make some changes to the component and submits the modified design after business hours. The engineer, now at home, receives an automated call to her mobile phone indicating the modified design is available for review. She goes online and approves the changes in advance of the next morning's project review meeting.

The Interactive Communication Platform

Science fiction? Not really. A new class of software product, known as the Interactive Communication Platform (ICP), is fulfilling this next logical phase in the evolution of converged voice and data services — today.

ICPs improve communications and productivity by delivering integrated voice and video capabilities to automated business processes, Web sites, and other distributed software applications.

Until now, most organizations have treated communication services differently than information services. Voice services have been deployed on stand-alone PBX systems with proprietary provisioning and management systems. Information services have been hosted on separate computing platforms governed by a different set of management applications, directory services, and user authentication and authorization systems. Tying the two worlds together has been costly, cumbersome and, in many cases, not feasible.

Computer Telephony interfaces, such as TAPI, attempted to bridge these gaps, but enjoyed only limited success because they required extensive programming as well as an intimate knowledge of each PBX vendor's specific implementation and call control and signaling nuances. Furthermore, these solutions were designed around static, device-centric models with centralized PBXs and stationary telephones and weren't well-suited for modern mobile and Internet-based applications.

ADVERTISING INDEX

ICPs deliver SIP-controlled voice and video capabilities as services that can be invoked by any software application in an emerging distributed enterprise architecture known as a Services Oriented Architecture (SOA). ICPs provide a platform-independent, language-neutral set of development tools for fast, flexible, and reusable application development.

Unlike conventional Computer Telephony interfaces, ICPs provide an abstraction layer that shields software developers from the complexities of the underlying communication infrastructure. ICPs require no prior familiarity with SIP or other communications protocols, so software developers can focus on business-specific features and functions, rather than generic communication services. Furthermore, ICPs are based on distributed, user-centric models and are well-suited for mobile and Internet-based scenarios.

ICPs allow communication services to be treated similarly to other IT services, enabling enterprises to leverage existing IT infrastructure including network facilities, directory and AAA services, business process logic, and registry and messaging infrastructure.

Traditional PBX with CTIs versus Next Generation Web Services ICP

Traditional PBX	Next Generation ICP
Centralized, phone centric architecture — designed for traditional static office environments	Distributed, user-centric architecture — well suited for mobile and Internet applications
Limited vendor-specific APIs — requires detailed understanding of communications protocols	Web Services APIs — abstract interface requires no familiarity with SIP or other communications protocols
Hardware-based solution — costly, vendor lock-ins	Software-based solution — exploits general purpose computing platform economics
Disjointed subscriber and policy management systems	Unified adds/moves/changes via RADIUS, LDAP or Active Directory
Proprietary phone instruments — limited choices, expensive	Standards-based SIP clients — choice of hardware or software based solutions to meet feature, form, and cost needs

Interactive Communications for Web sites and Business Processes

ICPs deliver myriad voice and video capabilities to a variety of software applications, in virtually every industry or market segment.

Click-to-Talk Functionality enables Web site visitors to communicate with enterprise employees, call center agents, or other community of interest members using voice, video, or chat.

Business Process Integration adds presence and voice/video services to CRM, ERP, SCM, or other business applications for integrated communications and point-and-click session establishment.

Business Logic Integration allows automated call establishment, based on business rules. For example, an application can notify a financial client when a stock hits a pre-defined price or connect a customer to a reservations agent when a flight is cancelled.

Session Correlation capabilities tie session information to business applications. Customer records and transaction histories can be linked to callers to better serve customers and prevent them from repeating transaction, service history, or other account information.

Interactive Voice Response (IVR) empowers callers to perform business transactions, access automated systems, or connect to departments or individuals through touchtone or speech, reducing transaction costs and improving the customer experience.

Call Recording permits callers or agent supervisors to record conversations for recordkeeping, quality control, or compliance purposes.

Speech and Video Recording and Playback enables subscribers to embed audio or video recordings into personal Web pages, such as social Web sites.

Software developers can leverage these features to embed communication services into enterprise software architectures and to tightly integrate voice and video with business processes.

Potential Applications for Interactive Communication Platforms

ICPs are applicable to virtually any industry or market segment. Candidate applications include:

- Customer Relationship Management
- E-commerce
- Enterprise Resource Planning
- Customer Care Solutions
- Human Resource Systems
- Supply Chain Management
- Social Web Sites
- Business and Personal Matching Services
- Web-based Advertising and Auctioning Services
- Internet Gaming Sites
- Industry-Specific Applications (i.e., Personalized Concierge Services for the Hospitality Industry).

Conclusion

Interactive Communication Platforms improve communications and collaboration between employees, affiliates, and customers. Software developers, solution providers, and Web site designers can leverage ICPs to add SIP-controlled voice and video services to Web sites, commercial or custom software applications, or internal business processes with lower development costs and shorter development cycles compared to traditional approaches. Enterprises can reduce administrative overhead and operations costs by unifying management systems and by consolidating operations and support functions. Forward looking organizations will enjoy increased customer satisfaction, greater employee productivity, and improved economics by leveraging ICPs to weave communication services into their business processes.

Alan Rosenberg (arosenberg@bluenotenetworks.com) is director of Product Management at BlueNote Networks. (news - alert) *For more information, please visit the company online at* http://www.bluenotenteworks.com.

Ensuring IP Voice Service Stability

By Ken Osowski

Identifying and Eliminating Common Single Points of Failure in Voice Services Architectures

The telecom industry continues its steady, impressive progress towards achieving the full potential of IP services deployment and adoption, but is it in danger of becoming a victim of its own success? With Tier 1s and Mezzanine providers increasingly scaling and evolving IP services, the goal of achieving TDM-like resiliency and recovery speeds is becoming an ever more urgent one to reach. Highly scaled, geographically disperse deployments are on the rise and providers are looking to price these services in competitive parity with other, more proven telco-class services. But, doing so means first rationalizing exposure to service outages and resulting SLA risks, a goal that hasn't yet been realized. Until now, even mundane and comparatively common IP network component failures disrupt services and trigger penalties; and recovery from a major disruption event, such as the failure of an entire service point of presence (POP), triggers a recovery cycle of unacceptable duration.

Entry-level table stakes for IP voice services scalability is, of course, redundancy. Physical redundancy of servers and N+1 configurations have been dominant characteristics of TDM architectures since the world was discovered to be round. But, services are now abstracted out and physical redundancies aren't sufficient. SIP enables an architecture that is flexible enough, because all the control logic is separate from the logic, so that availability of service features and preservation of call states can be separately programmed.

There is, of course, light years of difference between "can be" and "are." There is no guarantee that service creation environment and IP voice services developers have implemented HA features to leverage this model. In fact, you may find out what's missing in high availability capabilities only by finding out what's not getting to your customers: their services. For most network operators, that's a far too costly discovery process.

Whatever else it was, TDM was safe. If a TDM architecture had the necessary redundant components and failover capabilities, it was readily apparent. From a network operator's viewpoint and the PSTN standpoint, it was closed and secure.

In contrast, the stability of IP voice service applications — and the services they drive — depends on the particulars of a given IP network configuration, which may or may not be secure, performance-optimized, or failure-resilient; individual architectures may or may not themselves be market-hardened. An individual IP service architecture's potential service capability provides no indications as to whether it also has it has market-hardened failure recovery procedures and use cases that one would encounter.

For example, a single-site deployment may want to protect against an application server session failure. Class 5 switches have redundant line card capabilities in which service providers may choose to invest. That level of redundancy is also required to ensure that subscribers can maintain server interaction in the event of any failure at the application service network level and is especially important for collaborative applications where the network must maintain multiple simultaneous subscriber session states to stay intact, such as conferencing or prepaid calls, once the subscriber is speaking to the called party. What is the true cost of allowing these sessions to terminate because the application server has failed?

The best IP call session recovery model is based on a single IP voice session strategy to protect all services. Once a services broker or any intermediary is introduced, state problems can arise between applications and fundamental, expensive service integrity interruptions if the service broker itself fails.

Separating Service Architectures: Is Session Integrity Lost in the Stovepipe?

Proposed architectures regularly feature multiple applications servers, running separate applications, controlled by a service broker or proxy. In principle, this model seems to be a good fit for best of breed solutions, but it doesn't embody a unified, high-availability architecture. Heartbeat protocols between service brokers, application brokers, and databases maintained for single-subscriber models also aren't addressed in this architecture. Most importantly, separate service stovepipes architectures don't incorporate essential cross-over reporting and recovery capabilities among service applications and network elements for one reason: They can't.

Call state preservation is lost: Ensuring that SIP call signaling remains intact during an application server failure requires the duplication of call sessions and implementation of basic — but, until now, missing — notification and failover procedures in real time.

Unified awareness of network status is absent: In multi-site networks — a common scenario for Tier 1 providers — there must be full application awareness of any failure, such as RTP streams, participants, call legs, which audio channels on the media server, who's the moderator, etc.

Ongoing network integrity: Interoperability among various applications servers, applications, and proxies is an objective, but not yet a given reality. Is the SIP protocol sufficiently standardized to guarantee this interoperability? Not yet. Are standards bodies and competing vendor bureaucracies up to the task of solving and guaranteeing interoperability? Good questions, for which every vendor has standard assurances. But, why risk it, when a unified multi-service architecture enables both economical business model expansion and ongoing service stability and integrity?

So far, only the first tentative steps of vendor interoperability have been proven, but it's too soon to become excited. For an example of how interoperability may progress, look at the evolution of database technology. Before the SQL standard drove various vendors' access to data, every vendor had its own methods for accessing databases and maintaining transaction-level data integrity. More than two decades after the first major initiatives and shakeouts, interoperability at the SQL level is all that's been achieved. History teaches us that we're likely to achieve interoperability at the application level itself to enable issuance of service invites, but reliability and recovery are likely to remain unsolved for quite a while. This is why many established carriers have continued to balk at offering IP-based conferencing services, despite their obvious economic, flexibility, and innovation advantages.

What's needed is 100% software-based high availability features for service applications and their underlying architecture, which will allow new, more intelligent management and resiliency across multiple, geographically dispersed service points of presence simultaneously. This service application model must:

- Load balance service-enabling resources, including application server and media server resources, across multiple geographically-dispersed service POPs;
- Re-route call traffic across multiple sites around media gateway, application server, and media server failures;
- Provide intelligent, automatic failover of database processing to alternative service POPs; and
- Re-route all call processing to alternative service POPs in the event of primary site failure.

While this may seem ambitious, IP voice service developers are achieving these levels of service-embedded intelligence, enabling service providers to sustain both overall service call capacity and individual call states in the event of a myriad of common and extraordinary IP component/network failure conditions. This smarter SIP service architecture also features several side benefits, such as allowing service providers to replace complex, delay-centric system management intervention with automated, real-time call re-routing It also enables the optimization of ongoing utilization of all service-enabling resources and investments.

Such smarter, unified SIP services architectures are counteracting IP's inherent single points of voice service failure and enabling large carriers to eliminate their most likely sources of potential call state disruption.

Ken Osowski is VP of Product Management & Marketing at Pactolus Communications Software. (news - alert)*For more information, please visit Pactolus online at http://www.pactolus.com.*

Q & A

Speaking with Pannaway's Michael Skubisz


Michael Skubisz

I recently had the chance to ask Pannaway Technologies' CTO Michael Skubisz several questions regarding that company's position on SIP. Michael's comments follow.

GG: Why did you decide to develop your broadband access framework around SIP?

MS: Pannaway's (news - alert) Service Convergence Network (SCN) represents both a product solution and a vision for the evolution of broadband service delivery. This vision is based on several core principals that work in tandem to form an IP, Ethernet, and SIP-based network that can support multiple services and applications. The three principals are:

Deliver a fully integrated and remotely managed product line.

As the number of new communications and entertainment services emerge, the importance of a tightly integrated solution becomes essential. These services, many of which will be delivered in real time, will depend on a consistent end-to-end quality of service scheme, which we've developed within our SCN framework. Also, by integrating multiple Ethernet and SIP-based network elements within a single system, deployment, configuration, provisioning, and ongoing serviceability can be simplified and streamlined.

Deliver a Primary Line VoIP solution that will allow VoIP to be used to deliver a regulatory friendly telephone service.

While VoIP is a hot topic today, boasting over 25 million lines according to reports, four years ago, when we began development of our SCN solution, it was quite a different picture. VoIP solutions weren't being used in primary line scenarios; instead, they were marketed as low-cost consumer products or as enterprise PBX replacements. Our goal, with SCN, was to provide telcos with a solution that could leverage all of the operational and feature benefits of VoIP without sacrificing the reliability and safety aspects that have made the PSTN successful for 100 years.

Our SCN provides guaranteed automatic failover in the event of a power outage to a remotely powered back up VoIP-enabled digital loop carrier. This failover mechanism ensures that Lifeline calling and E 911 services are always available without the need for battery backup or multiple phone numbers to the home. SIP was a key enabler that allowed us to accomplish this task.

Deliver a durable product set that would evolve over time, allowing telcos to introduce new, revenue generating technologies in a non-disruptive manner.

The old adage is that the only constant is change, but how do you account for change in a complex network environment? Our technology is extremely modular in design and, due to our use of IP, Ethernet, and SIP, software upgrades can be automated. While many traditional DSLAM designs are chassis-based and leverage ATM or TDM, Pannaway chose to develop a series of self-contained units, which enable various technology choices, including VoIP-enabled POTS only, ADSL2+, full-blown triple play, FTTH, and more. These standalone units share common connectivity in the form of Gigabit Ethernet (up to 10Gbps) allowing them to be interconnected in a myriad of ways. Best of all, they can all be remotely managed and provisioned via a single management platform. Additionally, this modularity and innovative use of SIP technology allows for a limitless number of network design options.

GG: Why did Pannaway decide to include SIP in its SCN and how does it benefit telcos?

MS: During the initial design phase of our SCN architecture, it was critical that our solution be extremely flexible and scalable, not just on paper, but in practice. In that context, we examined the VoIP options that were in play and it became clear to us that SIP was the up and coming protocol of choice, primarily based on its flexibility and use of intelligent endpoints. As we began to build our VoIP product line, we believed that SIP was going to be very important to the marketplace, but we also understood that it would place an entirely different set of requirements on our products. Access products from other manufacturers, at the time, were designed to support MGCP or other legacy protocols and simply didn't possess the horsepower to handle the requirements of SIP. We recognized this short coming and designed our switching and routing access products to be SIP-compatible from the very beginning.

The importance of SIP is continually increasing in the telco marketplace. SIP is much more than a VoIP protocol; it is a multimedia session management framework and, while the early telco deployments have focused on the delivery of traditional telephony services, its potential is far greater. Our telco customers are already using SIP to deliver truly converged triple play services with advanced features, such as caller ID TV screen pops and time-of-day call forwarding. Moving forward, SIP will be used to enable services like video conferencing, inexpensive voice conferencing systems, gaming, IMS, and much more. The true appeal of SIP is that it is the only protocol in the VoIP space with the potential to enable services beyond POTS emulation.

TMC
14th Global VoIP Convention!
INTERNET TELEPHONY®
CONFERENCE & EXPO
The VoIP Authority Since 1998
San Diego Convention Center
San Diego, CA
October 10-13, 2006
www.itexpo.com
VoIP 2.0
Join Over 9,000 VoIP Professionals Coming to the Largest VoIP Conference Ever!
Educational Tracks Include:
- Enterprise/Government
- Service Provider Solutions
- Open Source Summit
- SIP Workshop
- IMS
- VoIP Security Summit
- Conferencing/Collaboration
- WiFi Telephony Summit
- VoIP Peering
- IPTV
Diamond Sponsors:
voipsupply.com
Platinum Sponsors:
IWATSU VOICE NETWORKS
AGN business class voip
NORTEL
CITRIX
INTER-TEL
AudioCodes
sansay
VONEXUS
DITECH COMMUNICATIONS
aculab
FORUM
netcentrex

Q & A

GG: Is SIP deployment becoming more prevalent across the industry?

MS: While very few access equipment providers are shipping SIP-enabled products today, most have announced their intentions to provide SIP support in future revisions or releases. Most of the softswitch vendors have already released SIP support on their platforms and the majority of today's CPE devices with POTS interfaces already support SIP. The reason that the access equipment vendors are lagging is that, for many years, the manufacturers of DSLAMs considered POTS support to be either something outside of the DSLAM or backward compatible with existing TDM networks. Simply stated, we believe that many access providers misjudged the importance that SIP would play in the delivery of broadband services along with the rapidity with which the SIP wave would hit the telco marketplace.

GG: Does SIP improve VoIP delivery?

MS: Absolutely. SIP-enabled VoIP networks deliver advantages that other systems simply aren't capable of. SIP is fundamentally an enabling technology with an architecture that readily accommodates new features and services in a highly extensible manner. Core facilities, like presence and events, differentiate SIP from other VoIP technologies. These distinctions enable support of VoIP integration with video conferencing, instant messaging, find me/followme across different access technologies, etc. SIP also provides a means to introduce higher fidelity audio capabilities and, of course, is very Internet friendly.

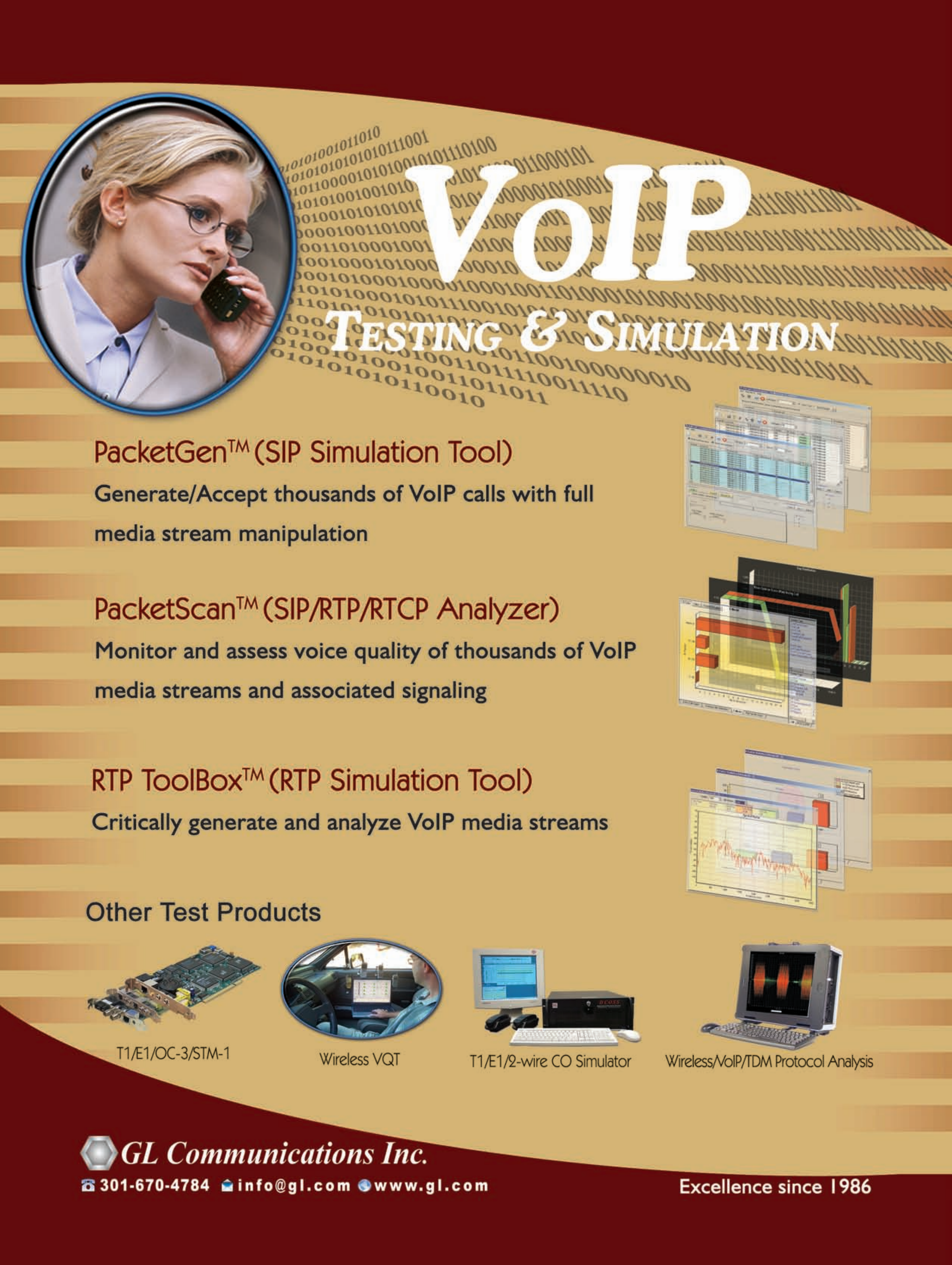
GG: Are there advantages to SIP outside of a VoIP environment that our readers may not be aware of?

MS: One unique advantage that SIP enables us to deliver is the ability to extend signal distance and to eliminate video corruption commonly caused by Ring Trip in a converged ADSL voice, data, and IPTV environment.

When ADSL and analog voice run simultaneously over the same line, the ringer voltage will often times cause unacceptable video interference. This phenomenon is called Ring Trip. The most common method, due to technological shortcomings and limited use of SIP, for lessening the effects of Ring Trip and correcting video errors is to reduce the ADSL train rate. That, in turn, shortens a telco's rate/reach capability and reduces the quality and level of billable services that can be offered.

Pannaway's SCN supports Primary Line VoIP enabling voice to be sourced at both the customer premise and at the remote terminal (RT) or central office (CO). In this model, the signal between the RT/CO and the customer premise is converted to all-digital IP Ethernet, eliminating the need for analog voice to run concurrently with ADSL video and data. This innovative technology, which uses SIP as its underlying engine removes the need for telcos to unnecessarily decrease train rates and shorten loop lengths.

ADVERTISING INDEX


VoIP
Testing & Simulation
PacketGen™ (SIP Simulation Tool)
Generate/Accept thousands of VoIP calls with full media stream manipulation
PacketScan™ (SIP/RTP/RTCP Analyzer)
Monitor and assess voice quality of thousands of VoIP media streams and associated signaling
RTP ToolBox™ (RTP Simulation Tool)
Critically generate and analyze VoIP media streams
Other Test Products
T1/E1/OC-3/STM-1
Wireless VQT
T1/E1/2-wire CO Simulator
Wireless/VoIP/TDM Protocol Analysis
GL Communications Inc.
301-670-4784 info@gl.com www.gl.com
Excellence since 1986