IP Communications Comes of Age

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TMC

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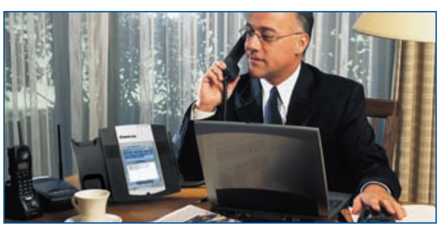

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EDITOR'S NOTE

Enterprise SIP Comes of Age

by
Greg Galitzine

Late last month, analyst firm Frost & Sullivan proclaimed that SIP is expected "…to replace the traditional modes of communication, and create an alternative communication industry reducing network elements to mere call-forwarding devices."

Although SIP (define - news - alert) is considered by many to be the de facto industry standard for call and session control, Frost analyst Shomik Banerjee pointed out that collaboration between and amongst vendors is still necessary to facilitate the standardization of the protocol.

"Vendors will need to collaborate and test their solutions extensively," he said. "Certification testing will also prove beneficial in assisting mainstream adoption of SIP-based applications and services."

In the enterprise communications space, work is ongoing, courtesy of the SIP-B (SIP for business calling features) initiative spurred primarily by a number of industry vendors, including Sylantro (news - alert), Siemens (news - alert), Polycom (news - alert), and others.

This issue of *SIP Magazine* features several excellent articles dealing with SIP in the enterprise and contact center space.

In his excellent article *The Voice of Business*, which appears on page 28, Hal Clark of BlueNote Networks explains how deploying a SIP-based enterprise communication platform today allows for use of existing telephone system investment while enabling process transactions to fully exploit communications for new business opportunities unencumbered by current limitations.

On page 32 Robert Winder of Genesys Telecommunications (news - alert) Labs talks about how deploying new SIP-based services in the contact center requires significantly lower investments in time and money than in traditional TDM systems.

Another element of SIP that's being talked about in enterprise circles today has to do with SIP trunking. SIP trunking enables an organization to converge voice and data onto a pure-IP pipe, allowing applications and services to remain "on net" out into the service provider's cloud. Among other benefits, this does away with the need for enterprises to deploy an IP/TDM gateway at the edge of their network. Not only does this save them the capital expense of deploying yet another piece of network equipment, it allows for the use of higher-quality codecs for voice transmission, resulting in CD (or better) quality conversations.

One player you need to check out in the SIP trunking space is Sphere Communications. The company's recently released Sphericall 5.0 is designed to enable service providers to deliver a new hybrid of hosted/premise-based IP PBX solutions, which gives enterprises a high-level of control over their communications environment as well as the ability to integrate on-premise IP communications with mission-critical business applications.

SIP is increasingly being deployed in the enterprise. And, as the research from firms such as Frost and Sullivan underscores, SIP is becoming an increasingly vital part of the global enterprise communications landscape.

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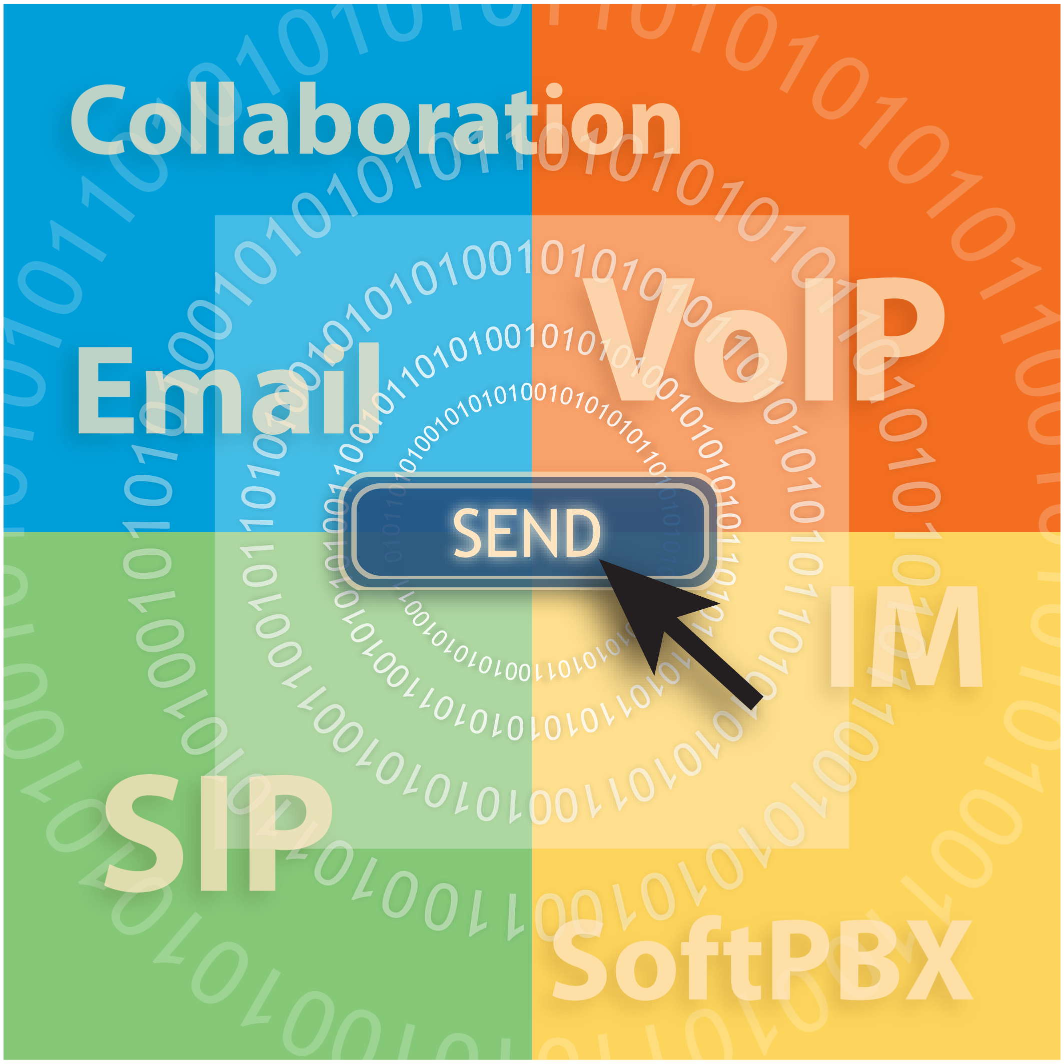
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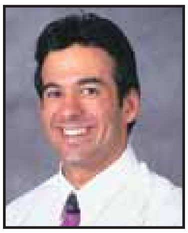
PUBLISHER'S OUTLOOK

Shifting Focus



by
Rich Tehrani

The whole premise of launching SIP Magazine was to develop a place where the community of people involved in SIP development and deployment can go to learn what is happening in the world of SIP. The purpose of this magazine is to provide analysis and perspective on the market while keeping you up to date on the latest news and announcements.

I get the feeling the timing of this publication was perfect, as the last month has seen an explosion in SIP announcements and the excitement in the market is beyond my belief. I spent a good deal of time speaking of SIP with industry icon Lawrence Byrd, Director of IP Telephony and Mobility Solutions for Avaya. (quote - news - alert) The company is devoting significant resources to SIP and has a
number of announcements related to session initiation protocol.

For example, the New Jersey Based communications solutions company announced it will support a greater number of SIP endpoints. Avaya supports interoperability with solutions from Juniper, Tello, and Cisco. Yes, you read that right, Avaya now supports Cisco phones. To me, this is akin to Larry Ellison inviting Bill Gates onto his yacht.

If you go back in the communications business long enough, you remember when the phones made a lot of money for PBX vendors companies. Even though the SIP standard has been around for years, not all phones are fully interoperable with PBXs from other companies. Well, it seems that Avaya is listening to customers craving interoperability and giving them what they want. This is pretty big news, in my opinion, and shows us that Avaya believes that money will be made on applications and a solutions ecosystem rather than just hardware.

The whole concept of Intelligent Communications — a term Avaya has been pushing as of late — is to sell people on solutions, not simple PBXs with dumb devices hanging off the end. Byrd stated to me that mid-sized and large organizations don't want cheap phones that break as soon as they drop. Avaya continues to upgrade firmware and adds features such as encryption to their devices. They feel they can compete aggressively on functionality and price.

The company is further committed to SIP trunking and they are working with a number of companies, such as AGN (news - alert) and AT&T (news - alert), to ensure that companies can directly connect SIP trunks to Avaya PBXs and take advantage of having the service provider terminate calls off net.

Avaya is also focusing on SOA and has worked with SAP to telephony enable mySAP CRM solutions. For more on SAP, check out my High Priority column in the March 2006 issue of TMC sister publication Customer Interaction Solutions.

The company has also upgraded Modular Messaging to version 3.0, providing unified access to voicemails, e-mails, and fax messages via phone or PC, as well as integrating with SIP networking.

Perhaps the biggest announcement Avaya has made recently is actually its smallest. It isn't a box or software but a simple telephone. If you have been following my articles over the past years you have no doubt heard me espouse the virtues of enterprise p2p solutions and Nimcat Networks solutions, in particular. I reported that the company was purchased by Avaya months ago and I've been waiting patiently for the fruits of this acquisition.

The point of p2p technology in the enterprise is that you don't need a PBX, just a phone that has all the functionality of a PBX built in. As you plug in more phones, the devices auto-discover each other and simulate a PBX, regardless of location.

In my previous writings I have mentioned that computer makers were eyeing this market carefully. Imagine a mail order computer vendor that offers such phones. They could offer a computer/phone bundle so that small companies can get their computing and telephony at once. I further mentioned this was a competitive threat to the incumbent PBX vendors.

A number of months later, Avaya purchased Nimcat Networks. This was a brilliant move on their part.

This technology has been a part of the new line of products form Avaya called one-X Quick Edition. Avaya says this solution is perfect for offices of 10-20 people and can expand to 100-150 people.

This is a also a great product for remote offices and has the benefit of phones that can be configured over the Web, meaning no local IT people need to be on hand to install and configure the devices.

In addition the company has also launched the one-X Desktop Edition, which allows users to have access to the capabilities of their phone, using their computer. Think of this as MS Remote Desktop for phones.

The last piece of the one-X pie is the Mobile Edition of this product suite. This technology allows the Nokia S60 family of phones to have access to Avaya Communications Manager software enabling users to be accessible via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My take on Avaya's announcements is that SIP and VoIP have transformed the company and breathed new life into their products and services. We are entering an era where communications is not about phones but about applications and Avaya has certainly capitalized on this shift in the market. Phones have been commoditized for many years, but applications can transform organizations. Presence-laden solutions that allow workgroups and entire corporations to respond more rapidly are the future of communications and Avaya seems to be happy supplying us with a healthy does of such products.

TMC has launched a few new events this year and I would appreciate you marking your calendar to attend both. The first is VoIP Demo, August 8-10, 2006 in Santa Clara, CA. The event will be unlike anything you have ever seen, as it will consist primarily of demos from IP communications companies.

This will be the only place in the world where you can see demo after demo of VoIP companies and then visit them in an exhibit hall.

As the VoIP market gets more confusing, you need to come to VoIP Demo to make sense of all the products and services on the market. You can register now at http://www.voip-demo.com.

The world's first IMS Expo will be launched Oct 11-13, 2006 in San Diego, CA. The purpose of this event is to help the world understand what IMS is all about and to see IP Multimedia Subsystem solutions in action. We have chosen San Diego as a venue for this initial event because we realize it is the capital of wireless technology and, as the wireless and wireline worlds merge through IMS, the industry will converge to IMS Expo to find the solutions they need. Registration is now available at http://www.ims.expo.com.

COMPANIES WORTH WATCHING

Bluenote Networks (news - alert) continues to make noise. The company makes products that leverage p2p and SOA to enable companies to communicate more effectively. Soon they will open up an API allowing other developers to have access to their platform so they can augment it with new and exciting applications.

In addition, they will focus on enabling voice communities from the most basic click to talk applications on your Web site to enabling a site like ESPN to host conferences made up of basketball fans.

So far Bluenote, Sphere (news - alert) , and Avaya (news - alert) have been pushing SOA and communications integration and I think this concept has legs and will take off in 2007.

Spectralink (news - alert) is introducing DECT-based phones aimed and the SME market. DECT is a wireless phone standard that originated in Europe and is heading to America. The technology is ideally suited for environments where existing corporate WiFi telephony solutions might be too expensive — think a doctor's office or warehouse with few employees. You can use inexpensive access points to communicate with the DECT phones, making them extremely cost-effective.

Getronics (news - alert) is the largest company in technology you may have never heard of. They have 27,000 people, mostly in Europe, and focus on network security and VoIP security. They are using their network probes to try to identify attacks such as DDOS before they happen so you can do something about these potential problems before they happen. The company offers a holistic approach to building voice and data networks and focuses on companies that have 500 seats or more.

Moving to the call center space, Spanlink (news - alert) has developed a product that helps contact center managers record conversations but more cost-effectively. Integrating with Cisco solutions, they record the calls on the agents' computers and upload them to a central server at night, when bandwidth is more abundant.

There is an integrated scoring capability and screens can be recorded with the voice to see where the agents were when they made a crucial mistake or did something brilliant. The system is 100% IP-based and can even push out user training modules when agents don't measure up to standards.

Being that this is SIP magazine I thought I'd finish up with a company that has SIP in its name. Sipera (news - alert) is new and focuses on VoIP security. The company has some heavy-hitting voice experience in its management team and is attempting to anticipate VoIP attacks before they happen and defend against them.

Some of the management team comes from companies like Cisco (quote -news - alert) and NetScreen (news - alert) , where the focus was on VoIP and security respectively. They are trying to produce solutions that are proactive as opposed to reactive. Of course, they tell me they will react if there is an exploit they didn't anticipate.

The goal is to have a comprehensive security solution in one box.

My take on this is that security in the world of IP communications is crucial. We haven't seen too many attacks yet, but we will. We must be vigilant. VoIP security will be very hot.


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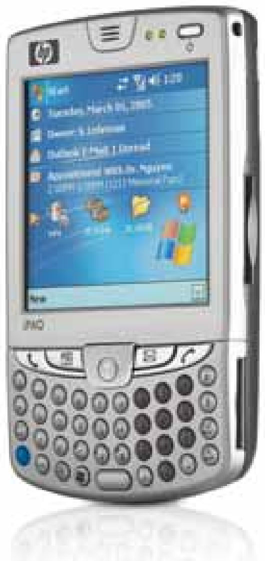
INDUSTRY NEWS

SIPquest Announces Enhanced Version of Mobile Console 2.0

(news - alert) Mobile Console, a software application residing in mobile handsets, delivers personal command and control of communications services over WiFi or WiFi and GSM or WiFi and CDMA network interfaces. Its single GUI and support for unified numbering consolidates both WiFi and cellular identities to provide a seamless end-user experience. The "Network Aware" feature detects and recommends network connectivity to allow users to place and receive a call over the best available network, optimizing for lowest cost, highest call quality or user preference.



The latest enhancements to Mobile Console include three major features: Instant Messaging, presence, corporate directory.

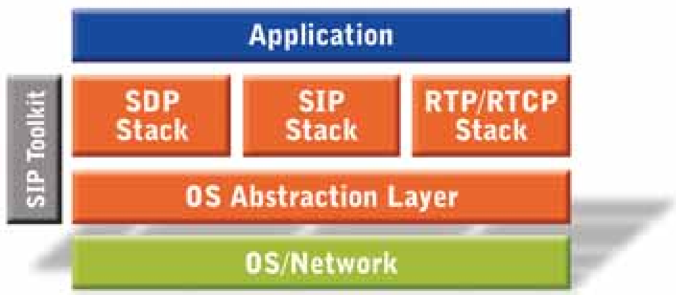
Enterprise workers can use these new features to check the availability of co-workers, sort through the corporate directory, enable three-way or four-way conference calls, sort through call logs of their desktop phone and have the IP PBX deliver calls to a temporary number.

According to David Hattey, president and CEO of SIPquest, "Mobile Console 2.0 is a major step toward the fulfillment of our vision of giving corporate user complete personal command and control of their communications capabilities - regardless of whether they are in the enterprise or a public setting. With these new features, enterprise users can expect enormous communication productivity gains anywhere, anytime."

http://www.sipquest.com

RADVISION Announces New Release of Its SIP Toolkit

RADVISION, (news - alert) a provider of multimedia conferencing and communications platforms, announced the immediate availability of version 4.0 of its SIP toolkit for real-time communication applications such as VoIP IM.



SIP Toolkit Architecture

"The entire telecommunications market is in the midst of a major migration from traditional TDM (time-division multiplexing) technology to SIP technology," said Adi Paz, Senior Director of Product Management and Marketing for RADVISION's Technology Business Unit. "RADVISION's SIP Toolkit 4.0 is a key enabler for this migration."

Specifically developed to address carrier-side applications, such as soft switches, gateways, and

application servers, the RADVISION SIP Toolkit 4.0 can support millions of busy hour call attempts (BHCAs) and has been enhanced with carrier-grade features including a reliable and effective Stream Control Transmission Protocol (SCTP) layer and automatic translation between phone numbers and SIP addresses (ENUM).

Using advanced design techniques, RADVISION has optimized the SIP Toolkit 4.0 for a small footprint, allowing it to serve efficiently in SIP terminals and IP handsets as well as network-side implementations.

http://www.radvision.com

Citel Unveils SIP-Based Handset Gateways

By Mae Kowalke

Citel (news - alert) released a new series of 12- and 24-port SIP-based handset gateways. The new products allow users to take advantage of feature-rich IP Centrex or SIP-based IP PBX services while still retaining their existing desktop handsets, wiring infrastructure, and data network.

Citel's new product line is designed to help companies of all sizes accelerate their migration to IP telephony services and applications, for both hosted and onsite configurations. The company's handset gateways connect directly to existing business-grade PBX handsets, which in turn connect to an Internet Telephony Service Provider or IP PBX over Ethernet using SIP.

"Some of the primary inhibitors to a forklift VoIP upgrade are business disruption and total deployment cost of new LAN infrastructure and IP phones," said Michael Robinson, CEO of Citel, in a press release. "Handset Gateways enable businesses of all sizes to migrate quickly to IP telephony and benefit from next generation feature functionality without buying or installing expensive new phones or performing costly LAN infrastructure upgrades."

http://www.citel.com

Covergence Delivers First Anti-Virus Solution for SIP-based Apps

Covergence, (news - alert) a provider of unified security and management solutions for applications and services based on the Session Initiation Protocol (SIP), announced that it has integrated technologies from McAfee, Inc. to ship the first complete anti-virus solution for SIP-based applications and services. McAfee's award-winning anti-virus solution is now tightly integrated into the Covergence Eclipse family of SIP application management products enabling enterprises to benefit from multi-modal, real-time collaboration while protecting them from the dangers of SIP-borne malware.

The proliferation of applications and services based on SIP and its extensions for instant messaging and presence (SIMPLE) represents a new challenge for enterprises concerned with protecting themselves against dangerous malware. One of SIP's greatest strengths is that it supports multi-modal collaboration applications that can support several different forms of real-time communication simultaneously. While this offers tremendous business benefit, it also creates serious security vulnerabilities.

McAfee Anti-Virus for Eclipse addresses these vulnerabilities and prevents the propagation of viruses, trojans, worms, and other forms of malicious software via SIP-based applications. The Covergence solution is unique in its ability to detect and to block malware embedded directly in the SIP signaling stream or in a SIP-associated content stream. The solution can also stop the spread of malware that attempts to propagate by embedding links to infected files in the bodies of instant messages.

"The explosion of SIP-based applications and services has created a whole new set of security concerns for enterprises," said Bob O'Neil, president and CEO, Covergence. "With McAfee Anti-Virus for Eclipse, organizations can finally have the peace of mind that comes with knowing their SIP-based solutions are protected."

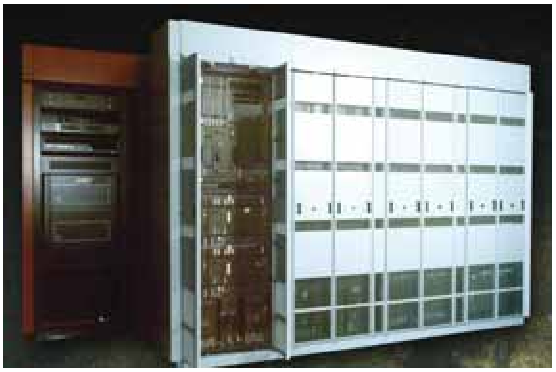
http://www.covergence.com

Nortel Enables New Rural Market SIP Access Systems

By Johanne Torres

Carriers looking into offering triple play services will have the option of tapping telecom technology provider Nortel to enable new SIP fiber to the home (FTTH) and converged IP access systems that will help them deliver broadband data, video and a set of voice features to residential and business customers in rural areas.

Nortel's (quote - news - alert) new systems are enabled by the company's DMS-10 softswitch interoperating with



products from Allied Telesyn and Pannaway Technologies. The bundle gives rural carriers access to a new portfolio of services such as VoIP with E911 lifeline calling, enhanced IP video and video on demand.

"As we evolve to VoIP, it is absolutely critical that we have the ability to provide advanced calling and safety features such as E911 lifeline support as well as new applications we can run on an application server. Interoperability between Nortel's DMS-10 and Pannaway's SCN will allow us to offer premier video and data services along with support for emerging applications such as HDTV and video on demand," Joseph Gottwald, CO engineer and ISP manager, Empire Telephone.

http://www.nortel.com

SIP Trunking Simplifies Conversion to VoIP, Saves Money

By Mae Kowalke

A recent survey by CTIA found that 60% of SMBs plan to increase their use of converged voice and data communication solutions during the next 18 months — an additional 13% had already implemented a converged solution.

One reason for the growth in converged communications is the advent of SIP trunking — a way for SMBs to cut down considerably on the startup and management costs of switching to VoIP phone service, according to Todd Landry, Senior Vice President at Sphere Communications. (news - alert)

Landry explained that until SIP-trunking entered the picture, using VoIP meant purchasing and managing an expensive and complicated system of gateways to convert voice signal from digital to analog and back again — often involving several such "jumps" during the call's journey from sender to receiver. "The jumps cost money, and they degrade the quality of the voice call," he noted.

SIP trunking eliminates the need for gateways by using software to manage a company's VoIP service, and by utilizing the carrier's already-existing network of gateways. When both caller and receiver are set up with internet telephony systems, SIP trunking is especially efficient, because it allows a call to travel the entire way as a digital signal.

http://www.spherecom.com

Sonus Networks Bridges Gap Between 2G and 3G Networks

Sonus Networks, Inc. (news - alert) announced that it has enhanced its IMS-ready SMARRT Wireless solution to include support for legacy wireless service protocols. The Sonus solution empowers wireless carriers with the ability to offer a bundle of services that include their existing services in combination with cutting-edge next-generation SIP (Session Initiation Protocol)-based services.

By providing support for key application protocols, such as Customized Applications for Mobile Network Enhanced Logic (CAMEL), into its Gateway Mobile Switching Center (GMSC) solution, Sonus enables wireless network operators to differentiate their service with new SIP-based applications while leveraging their existing CAMEL infrastructure for services such as prepaid calling plans. Sonus' solution provides an architecture capable of delivering legacy applications over a more cost-efficient IP-based network, while at the same time, providing a foundation for full scale migration to the IP Multimedia Subsystem (IMS).

Built on Sonus' award winning IMS-ready architecture, Sonus' SMARRT Wireless solution is a family of packet-based wireless switching solutions that allows carriers to cost-effectively deploy and operate wireless networks, increase network capacity to accommodate growing subscriber traffic, and build new revenue streams and customer loyalty through the delivery of packet-based enhanced services. In addition, Sonus' IMS-ready SMARRT Wireless solution provides a streamlined migration path towards Fixed-Mobile Convergence (FMC), the integration of wireline and wireless technologies and services to create a single telecommunications network foundation

http://www.sonusnet.com

SIPphone Announces Funding To Expand PC-Based VOIP Service to WiFi And Mobile Phones

SIPphone Inc, (news - alert) developers of the popular VOIP software Gizmo Project have secured $6.0 million in funding to expand their SIP standard service to non-PC devices, such as adapters, routers, WiFi handsets, and dual-mode mobile phones.

"The release of Gizmo Project has driven over 400,000 registered users to our PC-based service, but our goals go beyond free calls between PCs," says SIPphone CEO, Michael Robertson. "We want to seamlessly link the

internet and the traditional calling world of landlines and mobile phones and that's becoming possible as low cost SIP-based devices such as dual-mode mobile phones and WiFi phones become a reality."

SIPphone will use the investment to expand their VOIP platform to portable devices that do not require PCs to make or receive calls and promote adoption of the open standard auto-provisioning system plug-n-dial. The San Diego-based headquarters is expected to triple in size over the next year with hiring primarily in business development and engineering.

http://www.sipphone.com

BEA Named Spec Lead for JSR 289, SIP Servlets

BEA Systems (news - alert) has been named the specification lead for Java Specification Request (JSR) 289 for Session Initiation Protocol (SIP) servlets. BEA is continuing to help develop and evolve the Java-based SIP and Telecom Web Services technologies, which can help facilitate the rapid creation and deployment of next generation IP-based multimedia services.

BEA has been named the specification lead for the next release of the SIP Servlet API, version 1.1, as specified in Java Specification Request (JSR) 289. With SIP Servlet 1.1, BEA is set to lead the effort within the Java Community Process (JCP) — a standards organization for community development of Java technology specifications — to help build upon the foundation established in SIP Servlet API 1.0 (JSR 116), and enhance key areas of SIP application development and deployment. JSR 289 is slated to address enhanced control of application composition, invocation and convergence at runtime and deployment time. These enhancements are designed to help further accelerate the creation and deployment of highly rich, flexible, and dynamic IP-based real-time, interactive, multimedia services based on SIP.

"The SIP servlet specification is an important part of the overall SIP technology portfolio. Having APIs like SIP servlets is the key to bringing the web development community to telecom," said Jonathan Rosenberg, a Cisco Fellow at Cisco Systems. "That community is essential for building the kind of innovative services that SIP was designed to support."

http://www.bea.com

Ixia Tests Triple-Play Networks

By Johanne Torres

Calabasas, Calif.-based Ixia (news - alert) introduced its IxLoad 3.00, a test app for assessing the performance of triple play networks. Ixia's IxLoad triple play test system is capable of emulating a large number of subscribers, as well as the associated content servers. The system works in conjunction with Ixia's hardware platform to achieve the scale needed for real-world test scenarios.

The new IxLoad release includes features that enable realistic triple play testing. For example, IxLoad can accurately emulate millions of IPTV Broadcast Video and Video on Demand (VOD) subscribers and the millions of video streams they are viewing. Such large scale emulations enable the performance characterization of key IPTV elements, including video servers, multicast routers, and the video delivery network. The required protocols associated with these services, such as IGMP, RTSP and MPEG, are fully supported in IxLoad 3.00.

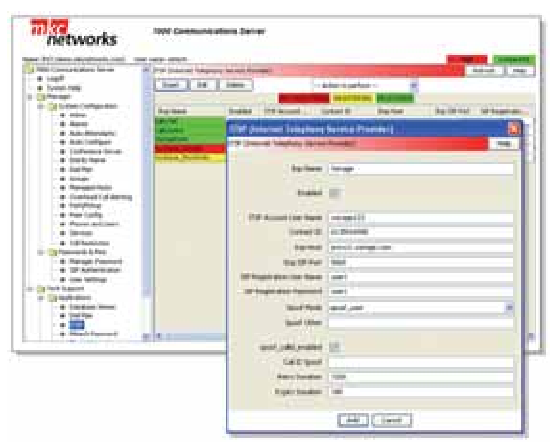
IxLoad tests the performance of Session Initiation Protocol (SIP)-based devices by emulating SIP callers and callees. The system also supports a set of Internet data protocols including HTTP, SSL, SMTP and POP, as well as Distributed Denial of Service (DDoS) attacks and vulnerability attacks.

Other new IxLoad 3.00 features include a built-in network impairment to emulate realistic network conditions; over 600 results metrics; capture and replay of custom protocol traffic; rapid adjustment of traffic and support for DHCP, LDAP, IMAP protocols; IPv6 support for HTTP and FTP protocols.

http://www.ixicom.com

More Than a PBX

MKC Networks' (news - alert) next generation 7000 Communication Server (CS) is a software-only Session Initiation Protocol (SIP) Application Server that pushes



the limits of conventional telephony. "Many have a hard time believing that standard computing platforms offer more value and performance than a purpose built product. There are significant benefits to this approach, including the speed of feature deployment, scalability,

cost and ease of use," notes Ben Morris, Sales and Product Marketing Manager of MKC Networks Corporation in Ontario, Canada.

The 7000 CS software on CD-ROM loads onto a customer-supplied hardware platform. The combination of open standards, a Linux operating system and SIP call control enables some exciting voice and data communications features such as desk and mobile phone "twinning," IP trunking directly to a carrier, phone relocation both internal and external to the corporate LAN, network alarms and flexible software-based licensing. SIP compatibility means the solution is not tied to a single vendor so that customers can choose from "best of breed" solutions, including phones (desktop and soft phones), unified messaging servers, call center applications, conference and collaboration solutions and vertical-specific applications.

The cost-effective 7000 CS is designed for a variety of markets, including both small businesses and large carriers since it can be easily scaled with the addition of software licenses. The 7000 CS is available in various license packs (10, 20, 50, 100, and 1,000) on one server, or up to 5,000 users using several servers. MKC Networks reports one carrier in Toronto, Canada is running the 7000 CS software on a powerful Dual Xeon processor, hosting 5,000 users. The 7000 CS can be deployed in global network configurations via SIP-compliant carrier-class gateways.

http://www.mkcnetworks.com

iTalkBB Chooses Covergence

iTalk Broadband Corporation (iTalkBB) (news - alert) has chosen Covergence's Eclipse solution to address the challenge of reliably scaling thousands of concurrent VoIP connections at the subscriber edge.

The subscriber edge is the boundary point at which service providers connect subscribers to their service. At the SIP subscriber edge hundreds-of-thousands to millions of real-time connections need to be managed and secured as they traverse the service boundary. In order to continue to grow its business while maintaining its position as a tier one VoIP service provider, iTalkBB sought out Covergence to ensure service scalability and performance at the subscriber edge.

Eclipse is the first product specifically designed to secure and manage large numbers of active endpoints. The remarkable performance and capacity of Eclipse begins with its hardware and software architecture which were purpose-built for the demands of the SIP edge. Eclipse's flexible, distributed design supports integrated clustering and load balancing to enable administrators to scale performance to carrier-class levels simply by adding Eclipse appliances to the cluster. With Covergence's (news - alert) Eclipse(TM) solution, iTalkBB can now offer subscribers the predictable performance and quality they have come to expect from their existing PSTN service.

http://www.covergence.com
http://www.italkbb.com

Verso Rolls Out SIP Gateways with Integrated IP Routers

By Laura Stotler

Verso Technologies, Inc. (news - alert) has launched its Session Initiation Protocol (SIP) Gateways, which include a VoIP gateway as well as an integrated, full-featured IP router. The new models support between one and six T1/E1 digital interfaces and offer a call capacity between 15 and 120 voice channels.

The new gateways offer IP and TDM connectivity for carrier and enterprise markets, and provide advanced gateway technology with an integrated access router for seamless connectivity on wireless, wireline and IP networks. They also work on interconnect legacy systems and the PSTN without sacrificing QoS, reliability or performance.

"This is a strong niche solution that addresses the market demands of carriers and enterprises alike, designed to work seamlessly on wireless, wireline, satellite and IP networks to interconnect legacy systems and the PSTN," said Monty Bannerman, chief executive officer, Verso Technologies. "The demand and growth for SIP gateways is a thriving business for carriers and enterprises alike, it is one opportunity that Verso and our award winning NetPerformer solution are prepared to capture."

http://www.verso.com

Mitel Extends IP Interoperability with 3300 ICP

Mitel (news - alert) delivers native support of SIP and SIP trunking with its latest release of the Mitel 3300 IP Communications Platform (ICP). Release 7 offers a host of new benefits including reduced access costs; easier implementation and facility redundancy; duplicated facilities to multiple providers; enhanced ability to increase (or decrease) capacity and improved disaster recovery capability.

Mitel has completed or is in the process of completing SIP interoperability testing with a number of key industry softswitch vendors, including Asterix, Broadsoft, Cedarpoint, Cisco Call Manager, Cirpak, Huawei, Ingate, Newport Networks, Nortel, Radvision, Samsung, Siemens, ZTE, and more.

Mitel enables users to fully leverage the power and feature functionality of the 3300 ICP through generic SIP endpoints and innovative use of existing Group and HotDesking functionality. Additionally, SIP connectivity to Microsoft Communicator for softphone functionality and third party call control inter-working with Microsoft Live Communications Server is enabled using SIP / CSTA / XML.

Microsoft's interoperability with Mitel is based on SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE). The companies are using the SIP and SIMPLE protocols to enable the federation of presence information and instant messaging (IM) that can be encrypted and authenticated. SIP and SIMPLE provide a rich set of solutions for computer and telephone interaction.

http://www.mitel.com

ON THE EDGE

SIMPLE & MSRP: An Interview with Robert Sparks

by
Erik Lagerway

SIMPLE — SIP for Instant Messaging using Presence Leveraging Extensions. Whew! That was a mouthful. SIMPLE is an architecture for developing rich presence and messaging enabled applications. In other words you can build a skookum Instant Messaging application using this technology. This is true but it's also much more than that.

In an attempt to get an accurate understanding of the current state of SIMPLE, I interviewed Robert Sparks the President of SIPfoundry (news - alert) and a member of the board. Robert is currently VP of Research and Development at Estacado Systems, providing solutions in real-time IP communications. Prior to Estacado, Robert held the CTO position at Counterpath (formerly Xten) focusing on SIP-based voice, video, instant messaging, and presence communication software and currently serves on the Technical Advisory Board at Counterpath. Before that, Robert was a Principal Engineer at dynamicsoft, the company that pioneered commercial SIP solutions. Robert is a co-author of the core SIP specification, RFC3261, and several of its extensions. He co-chairs the IETF's SIMPLE working group. Robert has focused on improving the level of interoperability of SIP implementations by coordinating the SIPIt and SIMPLEt interoperability events and by contributing to the SIP Forum Interoperability Test Framework. Robert is currently on the board of directors of the SIP Forum, and is an active contributor to the reSIProcate project. Robert has a BSc in Computer Science and a MSC in Mathematics from Texas A&M University.

Robert addresses SIMPLE as an open standard and the fact that this is now essentially finished in the IETF (Internet Engineering Task Force) and is on its way to the IESG (Internet Engineering Steering Group). This means that SIMPLE is now so close to becoming an open standard it's hard to think of a reason it would not do so.

This 25-minute interview has been truncated in this article but you can listen to it in its entirety on the Web at: tmcnet.com/260.1

EL: *Robert, where are we with SIMPLE today, is it nearing a useful state?*

RjS: Definitely. The good news is that SIMPLE is primarily a done deal from a standards perspective. We have long finished the core parts of the protocol. The Working Group still exists and is active. With a couple of exceptions most of our work now is simply taking the last crumbs of the meal.

The exceptions are to do with some ongoing work in putting together the last of the framework that's needed for a user of SIMPLE to be able to tell a service provider what their preferences are on how people can reach them. Which are the last parts of the mechanism that let XCAP (XML Control Access Protocol) work.

The XCAP data formats are all well defined and the protocol for the most part is well-defined. The one corner that we are working out right now is the modification of partial documents in XCAP. For example, if you want to modify your buddy list or make small changes to the buddy list or presence state without resubmitting the entire list back to the server. This incremental change piece is still something that is still in flight. Fortunately it's almost done, in two to five months we should have that over that wall to the IESG.

The second price that is still in flight is messaging sessions based on MSRP (Message Session Relay Protocol) which is in the very final process of the IETF standardization process. The Working Group has approved the documents and we are reviewing feedback before submitting this to the IESG.

EL: *What will partial updates for XCAP mean to the users?*

RjS: The best way to look at that would be talk about a couple of use cases. We currently have the technology to reproduce a classic Instant Messenger not unlike Yahoo!, MSN, AOL and that has been in place for a while. The work that we are doing right now will allow the user to say additional things apart from the typical status messages like "At lunch" or "On the phone". Imagine this… I have two people on my buddy list, Ben and Adam. I want to show Ben my entire and private Presence state, which could easily include "I am on the phone with X customer." And I want Ben to be able to see that, all I want Adam to see is "I'm on the phone."

The work we are doing right now will easily allow users to put this rich presence to work for them in a standard way so that you are not locked into one client or the other.

EL: *So this is really empowering Presence now, allowing the users to leverage the benefits of rich presence, not just basic status messages which are static for all.*

RjS: Exactly.

EL: *This is a path to localized Presence and very specific presence meant for specific purposes that everyone can use.*

In the past we talked about MSRP, and the importance of MSRP if SIMPLE was to take off and actually get some traction in the market. MSRP is required in order to provide the conferencing aspect to IM using SIMPLE, correct?

RjS: Sure, but it's not just that. It allows us to treat Instant Messaging the same way that SIP treats voice. So anything we can do with Voice we can now do with an IM text stream.

EL: *So now have conferencing, three-way calling, transfer etc.*

RjS: Right, it also provides a much more robust way to introduce into your text stream like pictures, audio, video etc. An example would be that if we were in an IM session and I had a picture I wanted to show you. I could do that and we could share that image dynamically without interrupting the Text session plus associating that image with the session when we were done.

EL: *So we can transfer images back and forth now, what makes MSRP so much better for this?*

RjS: MSRP allows us to break the file down into chunks and sending these chunks over in multiple streams.

(For the remainder of this interview please visit http://tmcnet.com/260.1.)

Robert goes into detail about how MSRP works and what the implication sit will have in Firewall traversal for file transfer and more. Robert also talks about how MSRP abides by Fair Use policies and takes into consideration the fact that bandwidth is not limitless.

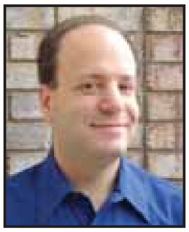
We talked about MSRP relay servers and the function of these servers in a SIMPLE network and who the vendors most likely to include this technology would be. Robert also mentioned that there are a few companies working with MSRP but none have released product yet, sounds like there could be a great opportunity here for a nimble software start-up! Robert sends a message to those working on MSRP to bring your wares to the next SIPit event in Tokyo in April 2006, find out more and register at http://SIPit.net or e-mail RjS@SIPit.net.

Robert closes with his positive and strong comments on the adoption of SIP by providers of business class VoIP and hosted IP Centrex. And that makes a great segue to my next article where I will be talking about SIP and VoIP in the enterprise and the powerful communications features that are now being deployed for business users at the edge of the network.

Erik Lagerway is an independent consultant and contributing writer for various publications, a full bio can be found here (http://sipthat.com). Contact Erik via e-mail: erik@sipthat.com.

SPEAKING SIP

SIP Peering: What It Does and Doesn't Mean



by JD Rosenberg

Changes are afoot in the Internet Engineering Task Force (IETF) efforts around voice over IP (VoIP). March 2006 will probably be noted as the beginning of two important new activities there. First, the IETF has created an entire new area, called Real-Time Applications and Infrastructure (RAI), which collects all the groups working on VoIP and VoIP-related technologies under a single umbrella. The second event is the formation of a new working group within that area, Session Peering for Multimedia Interconnect (SPEERMINT). As the name implies, SPEERMINT is going to be looking at SIP peering between providers.

Many Benefits

SIP peering has been a topic of conversation for some time now. There are many motivations to deploy it. The obvious one is cost reduction. When SIP peering is not used, calls between providers get routed over the public switched telephone network (PSTN), which usually incurs a cost for both providers. This cost includes the direct fees paid to the PSTN provider for interconnection, but also includes the capital and operational costs of running gateways to provide the interconnect. Doing a direct connection using SIP eliminates the need for the PSTN connectivity, immediately saving companies operating costs.

SIP peering can also greatly improve voice quality. Without SIP peering, calls between two SIP providers will require transcoding to G.711 going into the PSTN, and transcoding back coming out. This is not an issue if the transmission was using G.711 to begin with, but if it wasn"t, transcoding greatly reduces the voice quality. Regardless of transcoding, the gateways are likely to introduce delay into the system, as they'll need to provide an additional set of jitter buffers.

Worse yet, PSTN-based interconnects make it impossible to use wideband speech codecs. These codecs provide better-than-PSTN voice quality, and they are the source of the much-praised voice quality of Skype. With SIP peering, these codecs can be used for inter-provider calls. Without SIP peering, they can't. The difference between wideband and narrowband speech is huge — it's like the difference between HDTV and TV. Wideband speech has the potential to be a significant differentiator for VoIP, and it cannot be realized without SIP peering.

SIP peering also allows for services beyond voice to work between providers. Indeed, the whole promise of SIP is to bring new services to telecommunications. Video, presence, and instant messaging, for example, are all possible with SIP, but none of them work when two SIP providers interconnect via the PSTN.

No Extensions Necessary

But what does SIP peering mean? Is there some new special protocol required for SIP peering? What problems does peering introduce that aren't already covered by the existing specifications and products?

Those are good questions. First and foremost, SIP peering does not require any new extensions to SIP. The ability to interconnect provider networks is built into the SIP protocol itself. There is a common misconception that SIP peering requires some kind of special profiling of SIP in order to provide interoperability. That is simply not true. SIP was designed to interoperate, even among implementations that support different extensions and capabilities. SIP networks are interconnecting today without additional extensions. SIP has built-in negotiation capabilities that allow fallback to a common baseline set of capabilities when there is mismatch between sides.

As an example, SIP has an extension for preconditions (RFC 3312), which makes sure that a call proceeds only if a quality of service (QoS) reservation exists between the endpoints. What happens if only one side supports the extension? If implementations follow the specifications, they will correctly fall back to baseline operation without this feature. Now, some will argue that this is a problem. "We need this feature to always be used between our networks!" they'll say. The interesting thing is, the extension is implemented at the endpoints, not in the network servers. Thus, a SIP "profile" that mandates usage of the extension could not be applied to the SIP servers doing the interconnection.

Fortunately, the SPEERMINT working group has recognized that SIP peering is not about SIP profiling. Its charter explicitly rules profiling as out of scope, in fact. So, if SIP peering is not about a SIP profile, what is it about?

Improved Procedures

For the most part, SIP peering is about putting in place a set of best practices and operational procedures for the interconnection. Examples might include the following:

- For call routing — using carrier telephone number mapping (ENUM) to find the terminating provider.
- For accounting — keeping tabs on the calls between providers so they can reconcile bills, if needed.
- For security — using existing SIP security mechanisms, such as transparent LAN services (TLS) and SIP Identity.

Those practices don't manifest themselves as new protocol extensions, but rather, as capabilities in the SIP proxies that sit at the edges of the two networks and peer with each other.

That's not to say that new extensions won't be required to solve new problems. One such example is diagnostics, the ability to trace calls to figure out which network is causing the problem. It's important to note that SIP extensions for troubleshooting are not specific to inter-provider operation; they would be equally applicable to intra-provider troubleshooting. Indeed, drafts have been brought to the IETF on several occasions proposing such extensions. SIP peering only amplifies the need.

I, for one, am excited about this new area of work in the IETF. It's the first IETF group that is operationally focused, and its establishment signifies a maturation in SIP technology. As technologies move through the adoption curve and into the mainstream, the problems change from the basics of just making it work to the challenges of making it work reliably in a worldwide deployment. That's where we are now with SIP.

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

CASE STUDY

Leveraging Open Source SIP to Deliver Cost Effective VoIP

Founded in 1929, Sterling National Bank provides a wide range of products and services, including commercial lending, asset-based financing, factoring/accounts receivable management, international trade financing, commercial and residential mortgage lending, equipment leasing, trust and estate administration, and investment management services. So, when Eliot Robinson, CIO of Sterling National Bank, wanted a more cost effective alternative to its legacy Western Electric PBX and Centrex solutions to communicate between branch offices, the back office operations center, and the disaster recovery office, they looked for something reliable.

"The industry has standardized on SIP for IP communications, so we knew we needed a standards-based SIP solution to achieve our goals," said Robinson. "We looked at a variety of systems and there were no pure SIP solutions – phone, server and gateway. Pingtel's SIP-based solution was the obvious choice as the linchpin of our new system as it provided the most options for end points given its native SIP architecture and interoperability focus."

The other main driver was economics as Sterling's recognized a significant and rapid return on their capital investment using the Pingtel solution. Robinson estimates that including service provider access charges, Sterling will be able to lower their average monthly costs for telephony services by two-thirds once it replaces all the legacy phones.

Sterling uses Cisco SIP phones, connected to an IBM server (with Red Hat Linux) running Pingtel's SIPxchange PBX, and a Cisco media gateway to connect to the PSTN. Sterling expects to expand its production environment to 700 seats quickly as the economics clearly favor this approach. "One huge advantage to the Pingtel solution is that it is compatible with a variety of phones – Polycom, Cisco, GrandStream, and more, so Sterling has the ability to use whatever SIP phone that meets SIP standards and has been approved by Pingtel to work with Pingtel's SIPxchange. "

An open source business model means Sterling does not have to worry about whether a particular software vendor is going to be around to service the source code.

Once Sterling has the system fully implemented, Robinson likes the fact that their options, in terms of endpoints, go up dramatically. In tests done to date, he's found consistent voice quality no matter the endpoint. In the future, Sterling plans to integrate the system with more of its computer systems. For example, if a customer went to Sterling's Web site, they could use the system to set up a voice chat (IP call) to answer questions directly from their computer.

Why Open Source?

An open source business model means Sterling does not have to worry about whether a particular software vendor is going to be around to service the source code. For Robinson, it's kind of insurance policy. "Pingtel is important because they supply support, configuration tools and training. We want to have the flexibility to add phones as we see fit and we don't have to worry about the licensing every time we add another phone as we do with other business models," he added.

Partnering for Success

Given Sterling's business, the bank had specific needs outside a typical configuration so having a partner that could provide the requisite professional services was critical to Robinson and his team at Sterling. "We asked Pingtel to provide guidance in how to set up a back up server in case we had a hardware failure and the professional services team more than exceeded our expectations," said Robinson. Pingtel's professional services team also helped Sterling set up a sophisticated ACD-like functionality for some of the bank's help desks.

The Bottom Line

According to Robinson, Pingtel combined the three things Sterling needed to move forward with this type of solution: a low-risk, easy-to-use SIP-based system, a fundamentally better economical model for Sterling's telephony needs and a technically savvy professional services team that helped tailor the application to meet its specific needs.

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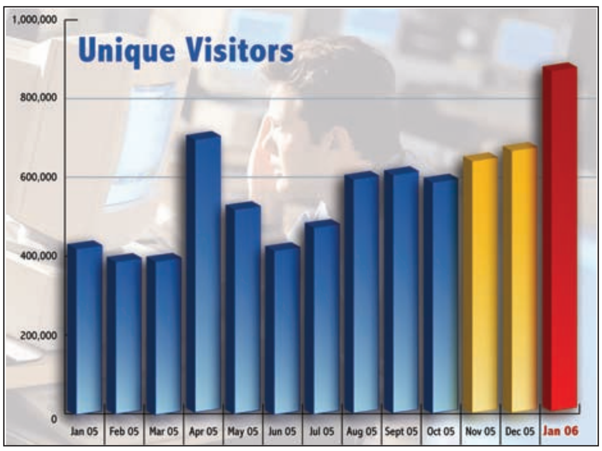

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Source: Web Trends February, 2006

**Source: Alexa.com ranks Web sites by traffic. The number indicates a site's proximity to being the number one most visited Web site. Date: 3/6/06
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TMCnet.com Traffic vs. Technology/IT Web Sites

Web Site	Alexa Site Rank
TMCnet.com	**1,647**
eWeek.com	1,942
Computerworld	2,940
InfoWorld	3,980
Network World	6,195
Light Reading	12,471
Wireless Week	42,137
Pulver.com	43,326
Telephony Online	52,860
Destination CRM	56,033
VoIP News	68,122
Telecomweb	191,198
Telephony World	229,201
Call Center Magazine	234,504
CommWeb	243,661
America's Network	251,349
Wireless Review	604,039
Communications News	673,705

TMCnet.com Traffic vs. Business Magazine Web Sites

Web Site	Alexa Site Rank
TMCnet.com	**1,647**
Smart Money	2,992
Fast Company	3,748
Red Herring	4,370
Inc. Magazine	6,008
Barron's Online	6,551
Technology Review	7,382
Weekly Standard	8,342
CIO Magazine	12,663
Fortune Magazine	13,063
BtoB Online	25,004
Worth Magazine Online	217,538

TMCnet.com Traffic vs. Fortune 500 Company Web Sites

Web Site	Alexa Site Rank
TMCnet.com	**1,647**
GE	3,209
Ford	3,329
General Motors	3,797
Coca-Cola	5,718
State Farm Insurance	6,513
DuPont	17,070
Kroger	22,656
AIG	29,404
Chevron Texaco	34,412
Exxon Mobil	34,607
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TMCnet Traffic Analysis

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SIP: A Key to Convergence in the Enterprise

By Tarun Kapoor

In today's increasingly connected world, many employees of an enterprise are computer literate enough to use the latest communications technologies, such as an Instant Message (IM) application or an IP phone. Yet they still complain that they need multiple applications to reach their colleagues or customers; one application for sending and receiving Instant Messages, one application or device for telephone and/or video calls, and another one for collaboration tools such as a whiteboard. Occasionally, they even need to use one application just to set up another application. It is even worse for system administrators who have to manage multiple, disparate systems. However, there is an answer: the "Converged Platform" where multiple features and functionalities are all combined into one client or device; a "one-stop" concept, making it a potential candidate for the next killer app. The increasing need for scalable and flexible converged communications mandates SIP as the protocol that serves as the backbone of the platform.

But before we can begin to decipher why SIP is so important to converged communications, we must first redefine the term enterprise convergence. Today, enterprise convergence is narrowly defined as the merger of voice and data networks in an IP environment. However, this limited definition distracts from the real goal of enterprise convergence.

To truly embrace network convergence, an organization must consider voice communications as simply another application. Convergence therefore should be viewed as the sharing of IP network resources and services among different applications. These applications can provide features such as voice, data, video, etc. What if you could provide all of these features on a single platform for enhanced communications, increased productivity, and a complete converged experience? This is how convergence should be viewed; the concurrent delivery of various media types to a single application (Figure 1).

Leveraging the right protocol for convergence is crucial and no protocol to date can be compared to SIP; it is the key to convergence in the enterprise. SIP already has built-in support for many convergence features and functionality such as setup and negotiations of multimedia sessions and parameters, support for Instant Messaging (IM) and Presence,

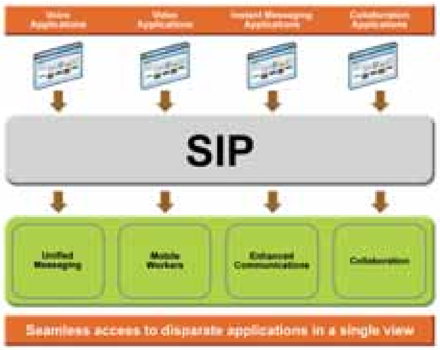

Figure 1

and many extensions (e.g., UPDATE for conferencing) to provide other advanced features of converged communications. This eliminates the need for multiple servers supporting multiple features and allows for easier administration of both users and features. Network service providers can benefit from having one server for all features, a "one-stop" server, to provide all the converged features, even providing them through NATs and firewalls by utilizing a persistent TCP connection to the server.

SIP was initially designed to be a call setup protocol, though now it has matured to offer many different services. The main function of SIP was to set up a multimedia session between two endpoints, and even now this is still one of the strong points of SIP. The two endpoints can negotiate multimedia parameters before the session setup, during the session setup, or change the multimedia parameters in mid-session. Imagine how convenient it is to be in the middle of a voice session, then change to a video session, and finally add a whiteboard to the video session, yet the user does not lose the current multimedia session if the remote endpoint cannot support a desired multimedia type. Using SIP, all this can be automated; for instance, suppose you want to change an existing voice session to a video session. You initiate the video session from your User Interface (UI), yet the two endpoints automatically determine that one endpoint does not have video capabilities and then you get notified with this information without the remote user ever realizing that you wanted to change to a video session.

With its scalable and flexible architecture, SIP easily incorporates innovative and unique features for enterprise convergence. For example, a SIP server can invite users to persistent multimedia conferences such as a "push to talk" persistent audio conference. As the server also "speaks" presence (i.e., a presence agent) by sending presence updates, it can notify existing members of the conference of the addition of new members, while simultaneously negotiating the multimedia session parameters. This powerful functionality is excellent for supporting multiple conferences and presence updates per user.

As users recognize the benefits of converged communications within the enterprise, they are very likely to demand more advanced services to augment their converged experiences, such as event notifications. The event mechanism for event notifications is one of the key aspects of SIP. The events can be generated (and subscribed to) by either the server or the client. One client can notify another client about a client-specific event (e.g., when they are on the phone with a specific customer) and/or all clients can receive notifications about system-wide events (e.g., scheduled system maintenance). Since any device on the converged platform (phone, handheld device, or soft client) can subscribe to and receive different types of SIP event notifications, it can integrate the event notification with an existing feature (such as automatically invite other users to a multi-party conference) or even third-party applications such as a scheduler or automatic dialer (with additional XML support).

By supporting Internet-based standards such as XML, a converged platform for the enterprise can easily integrate with various other Internet-based applications and protocols. For example, with XML support, a SIP-based converged application can integrate with public IM clients such as Yahoo! Messenger or MSN Messenger and possibly even enterprise IM clients. Web services, XCAP-based configuration servers, even real-time streaming of live corporate video feeds… these are some of the potential services of the enterprise converged platform; they are feasible due to the inherent tying into technologies that many applications and services are using today and which users are accustomed to. Keeping in mind that SIP was designed to be a modular protocol and that it was based on the HTTP protocol, this is a logical consequence of those decisions. The initial goal of SIP was to provide some core features and functionalities for multimedia session setup that could be leveraged by applications or devices in coordination with other protocols to provide a rich user experience (e.g., convergence). The fact that SIP has added many extensions to provide additional functionality does not alter the original concept of modularity.

Yet modularity is only one key concept which differentiates SIP from other signaling protocols. For enterprise convergence, the real benefit of SIP compared to other protocols is where the intelligence for a service resides. There exists no notion of a client-server architecture, or dumb endpoint versus intelligent network concept. Depending on the type of service, the server makes the intelligent decisions (e.g., routing decisions, processing decisions, network decisions, etc.) or the client implementing the service communicates the information to the server, or it can even be two clients using a peer-to-peer mode to provide the service. This is ideal as each feature of the converged platform can be used in different styles based on the end user expectations and network infrastructure available. For example, based on changing network conditions or endpoint mobility, either the server or the client can renegotiate the multimedia parameters in an existing multimedia session.

Of course, as SIP is currently considered the foundation for the IP Multimedia System (IMS) industry, of which converged communications is a subset, it only seems logical that SIP would be used as the backbone protocol for enterprise convergence. Venture Development Corporation (VDC) estimates worldwide markets for SIP infrastructure and software to exceed $5.5 billion by 2007. This represents a compound annual growth rate of 36.1 percent between 2003 and 2007.

According to Chad Hart, VDC's Telecom/Datacom Practice Director, IMS adoption will drive SIP growth, with initial trials through 2006 and more significant deployments in 2007. "The acceptance of IMS by both wireline and wireless carriers solidifies SIP's role as the protocol of choice for all network operators. Most carrier-focused vendors will repackage their existing SIP offerings to cater to this rapidly emerging market."

As more and more vendors and service providers are moving towards SIP-based services and offerings, many of the drawbacks and missing pieces of SIP with regards to enterprise convergence will be resolved with extensions to the base protocol (that is not to say that there will be a day with no shortcomings in SIP).

As more vendors implement converged solutions for the enterprise, they will need to provide more enhanced features and services to differentiate themselves in the marketplace. Understanding the advantages and benefits of using SIP as their foundation will have a profound impact on their ability to provide these new services to their customers.

Tarun Kapoor is the CEO of Pangean Technologies (news - alert). For more information please visit this company online at http://www.pangeantech.com.

Using SIP & SS7 to Deliver Seamless Voice Services Over Fixed & Mobile Networks

By Andre Moskal

Service providers worldwide are under tremendous pressure to deliver meaningful services that not only retain customers, but also attract high-value business subscribers. The competition from emerging VoIP service providers, cable operators, and other competitive network operators has led to increased customer churn, which in consequence increases costs and lowers overall profitability of existing services.

Remote mobile, home, and business locations represent important communication connectivity points for most people. However, each one of these connectivity points has its own network characteristics, requiring the user to be familiar with multiple communication devices and different user interfaces. The ability to deliver services that enable mobility, feature-consistency and an "always-connected" paradigm in a multi-access environment would have tremendous value for service providers, and encourage customer loyalty in an increasingly competitive marketplace.

But this opportunity is not without significant challenges. Service providers must find a way to deploy services not only across their own disparate networks, but also across their partners' and customers' various access networks. Hence, service providers must make a tough decision on how to balance future-facing offerings of value-added services focused on a specific set of devices, such as emerging multimodal handsets, against the exploitation of the deployed base of fixed and mobile devices.

Developing a Relationship

Perhaps one of the most significant areas of recent market change is in the relationship between fixed and mobile voice services. Initially service providers viewed these two service types as independent, each serving their own respective market, with mobile services specializing in addressing users demanding communication on-the-go for business or social reasons. As the price of wireless voice minutes fell during the 1990s, and as the mobile data services continued to struggle with mass market acceptance, mobile operators started targeting wireline subscribers as a source of their continuous growth, creating in turn a fixed-to-mobile-substitution trend. Likewise, the appearance of unlicensed spectrum technologies, such as WiFi, and the tremendous growth of WLANs and Hot Spots created opportunities for the wireline service providers to offer mobility at an attractive cost to their own customer base. Hence, two types of service providers are driven by their own competitive pressures to collaborate in order to supplement their most valuable assets such as broadband access and ubiquitous licensed spectrum radio coverage.

Operators that can supply fixed and mobile services can get some relief from voice call price commoditization, but it is clear that the emerging VoIP services will impact not only wireline voice pricing, but also wireless voice. The primary driver for VoIP is the deployment of low-cost broadband service to consumers and businesses, driven by the desire of emerging voice services providers to create another revenue source through Internet access. This same desire exists in the mobile space, and the result will be a reduction in the cost of mobile broadband capacity, eventually pushing down the price of voice calls.

The Internet has already provided the world with the first broadly accepted mass-market model for non-voice services, and IP data is the largest driver of change in the network today. In fact, providers worldwide are migrating all their services to IP backbones, both to lower overall cost through the elimination of multiple service technologies, and to prepare for a future in which all services will be delivered in IP form as part of the fixed mobile convergence (FMC) trend. The interest of wireless and wireline service providers in IP multimedia subsystem (IMS), licensed and unlicensed wireless spectrum, as well as the increased number of multi-mode handsets, are all proof points that FMC is a real and current phenomenon. There is no alternative for any voice provider other than to embrace FMC, and find a way to profit from it.

Creating Linkages

FMC is a strategy that creates linkages between wireless and wireline voice services, meaning features that either migrate between the two services, or work by coordinating user behavior across both services. There are a number of applications that can be based on FMC, including:

There is no alternative for any voice provider other than to embrace FMC, and find a way to profit from it.

- Call routing and disposition in the context of network presence, where an FMC user can select which service is to be used to complete calls, or have the call direction default based on the user's presence within a multi-access environment.
- Multimodal handsets, offering both WiFi and cellular capability that can be used as extensions of a VoIP service on home/enterprise WLAN networks or as standard wireless handsets when no private network is within range.
- Device changes during an active call, including cellular to wireline (VoIP or TDM) or vice versa; used either for consumer convenience or to provide a backup mechanism for cellular services, should those services be unreliable due to the poor in-building coverage.

The industry has taken a number of different approaches to FMC. They include PBX-based adjuncts, cellular network-based emulation, and IMS network-based solutions as described in this article.

For some time now enterprises have been demanding effective solutions from the IP PBX vendors to reduce the expenses related to mobility and roaming charges. Major IP PBX vendors already provide "on-net" wireline services for optimal call routing and are in the process of introducing solutions in partnership with handset and WLAN equipment manufactures to address these requirements for mobile users. Although in high demand, the existing solutions fall short in addressing truly seamless handoff between enterprise WLAN and cellular networks, and require two-stage dialing to access IP PBX dial tone for voice supplementary services. Of course, the advantage of this type of solution is the ability to use dual-mode handsets as extensions of the IP-PBX while within the enterprise WLAN coverage.

The dominant cellular emulation-based solution is unlicensed mobile access (UMA) delivering GSM voice and GPRS mobile data services over WiFi. The solution is based on emulation of the base station controller (BSC) that is managing a WLAN access network and interconnects to the core GSM/GPRS networks as a typical BSC. Because of the soft handover capability between the BSCs, UMA solutions offer seamless handover and consistent user experience regardless of the cellular or WLAN access. So far, UMA has been the only FMC technology standardized as part of 3GPP activities. Of course, there are a few disadvantages to UMA. Namely, it is difficult to integrate UMA within the enterprise environment due to the lack of straightforward integration with SIP. The current viewpoint among service providers and vendors alike is that UMA has a limited life span as an interim solution, and that in the long term it will be replaced by the IMS-based approach.

To support consistently new value-added services in a core IP network, a new standard architecture called IMS has been defined by the 3GPP and adopted by the 3GPP2 organization. IMS also facilitates service convergence in the fixed line networks and it has been defined as open, multi-media service architecture for mobile and fixed networks using SIP as a unifying signaling protocol for voice and multimedia sessions. Although current IMS standards are still in the early stages when it comes to defining interaction between fixed network services (such as IP PBXs) and mobile services, the IMS framework greatly reduces the complexity of integration.

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The Voice of Business

Deploying SIP-Based Enterprise Communication Platforms

By Hal Clark

Simplifying the creation and execution of business automation processes is at the heart of IT systems migration to a Service Oriented Architecture (SOA). The personality of business transactions with a specific enterprise is enhanced through the use of voice and video interaction –– but full exploitation of these personal connections cannot be achieved using today's telephony systems. Enterprise communication platforms, architected from the ground up around SIP, operate as applications within the IT infrastructure used today to execute data-oriented process automation. SIP connects interactive voice and video elements serving business transactions in a defined process to diverse applications and user devices. Deploying a SIP-based enterprise communication platform today allows for use of existing telephone system investment while enabling process transactions to fully exploit communications for new business opportunities unencumbered by current limitations.

Personal Connections Capture Customers

Throughout history, exchanges of services and products for negotiated value have been conducted person-to-person. In spite of revolutions in communications technologies, personal interaction remains significant in the development and maintenance of strong relationships in our social culture. Businesses can exploit voice and video interaction for stronger customer relationships that differentiate their enterprise from their competitors.

Much marketing effectiveness has been achieved by the appearance of Web interfaces to automate business transactions at the convenience of the customer. However, convenience is only one dimension in attracting a sales transaction. The customer is not always attracted to this "self-help" paradigm and will ultimately be alienated by the isolation or lack of a personal connection to the business. "Personal touch" is one more method in the business arsenal to capture as many customer transactions as possible, whether these transactions are recurring for each customer or are attracting new customers through positive referrals.

Opportunities & Current Limitations

"Personal touch" places conflicting demands on today's business operations environment. Practical financial considerations drive staffing towards producing and delivering products. In many cases, the people designing or handling products may be the most knowledgeable resources to assist customers in a way that is most productive to the enterprise. For instance, a person working at a retail location stocks the floor and assists customers in the store but may also be inactive at times when on-line customers could use personal assistance. Duplicating the knowledge to assist customers in a store and over the phone spreads the investment out to support both stores and call-centers separately. This thinning of "customer touch" usually results in customer dissatisfaction with the level of service. A more productive approach may be to have more enterprise employees producing or delivering products as well as being accessible to customers online. In the retail example, more staff could work in a store location and also have the experience to assist customers by phone when not assisting customers on the floor. An additional benefit is that more employees have experience with customer issues to apply in product development or delivery – another way to improve enterprise value.

This nirvana of business productivity is difficult to achieve with existing communication solutions: specifically phone and video conferencing systems. While data application environments have advanced significantly in both user-interface and process-flow automation, voice and video communications remain locked in a separate environment through continuing use of historical PBX and conferencing systems that evolved with no concept of future transaction or financial considerations. Computer Telephony Integration (CTI) went part of the way by connecting telephone and computer environments for better coordination of transactions – better "personal touch," but neither customer satisfaction nor business productivity were optimized. In fact, these complex, expensive implementations are proprietary and require extensive work to implement and maintain as business processes change.

> While data application environments have advanced significantly, voice and video communications remain locked in a separate environment.

Migration

Simultaneously attracting customers and achieving financial operating objectives require a synergistic combination of voice, video, and data applications into an automated transaction process flow using a unified environment. The advanced development of data application platforms and operation management systems suggests this environment as the place to add voice and video interaction elements to business transaction automation.

New and emerging Web technologies can simplify the definition of automated processes. Building blocks for data application interfaces include an extensible "http" protocol and "XML" information format that enable a fast, customized creation of "Web Services." SIP is conceptually equivalent and focused on enabling similar "Services" for person-to-person voice and video conversations. With these common family attributes, why not combine them in a common services automation environment?

A SIP enterprise communication platform is exactly what is required to accomplish this optimized operating environment. Taken from the perspective of a "Web" service, the SIP enterprise communication platform is implemented as a software application of voice and video services that resides in the same operating environment as "data" applications and uses inter-communications more native to the "data" application. For instance, a process can invoke a connection between a customer and a "knowledge resource" using a "Web Service" interface. The SIP enterprise communication platform manages the specific requirements of the voice and video connection in conjunction with the "data" application. Both "Services" operate concurrently in a coordinated manner to conduct the single transaction.

Once the process application determines that human intervention is required to proceed with the business transaction, locating an appropriate "knowledge resource" is the next decision. In the retail example mentioned earlier, a SIP enterprise communication platform can be extended to understand the "knowledge base" of users on the system as well as making the connection to a user that is not occupied by other tasks. Take the case of a customer trying to determine color matching during an online purchase of pants and jacket. Items viewed by the customer during the session, along with supporting queries, identify an area of interest that can be mapped to the employees that can assist the customer's decision. Further examination of the knowledge group determines who is at work in a store location and not occupied by another customer. The SIP enterprise communication platform makes the connection directly between the employee and the customer wherever they are located. This personal interaction quickly determines the additional information needed to successfully complete the order. Available resources were used to facilitate the customer order without requiring a separate contact center.

Note that while knowledge resources can be reached via existing PBXs, complications arise for environments in which employees do not stay at the same desk all the time or are not dedicated to "contact center" seats. Finding the appropriate person to respond to customer issues requires association of skills to individuals in the "resource" pool along with their location and availability. A SIP enterprise communication platform determines the "availability" of individuals through their "presence" on the system along with skills inventories and call policies rather than the 'line'-based definitions used by legacy PBX systems and call centers.

Different "phone" sets may also be used by employees depending on whether they are at a desk, in transit, or out of the office. Indeed, an employee may sit at different desks as they proceed through their work activities. With an end-user, identity-oriented perspective of the communications environment, the SIP enterprise communication platform enables mobility across a range of "phone" set types based on the "presence" of the user in the system. Implementing a SIP enterprise communication platform provides backwards compatibility with installed legacy equipment with the native ability to interface directly with alternate user-interfaces as the communications environment migrates to newer applications.

A SIP enterprise communication platform enables you to move towards future operations productivity with improved customer loyalty but without requiring replacement of installed communications equipment in the initial phases of migration.

Service Elements

Today's data application and telephony systems cannot accomplish the goal of a unified business-process environment for flexible operating scenarios. The guiding principle of process implementation embodied in Service Oriented Architecture (SOA) treats applications interacting with users as services, including voice and video communications. Implementing functions that find users and manage communication sessions as services in an SOA environment removes a requirement for each application to provide the specialized functions needed to communicate directly with a customer. While the application interface implementing a verbal communication session may be similar to the data interfaces used for the "Web" presentation, the functions necessary to locate users and manage the media requirements can be reused by many applications.

A SIP enterprise communication platform supports mobile availability of users and flexible interworking with processes defined in the SOA that are so important to future productivity. Essential elements for business process also include stored voice and video for applications like voice mail, unified messaging, auto-attendant and automated voice-response systems. A SIP enterprise communication platform orchestrates resources — implementing these functions more globally and flexibly than legacy systems and making them available in a more intelligent manner to SOA-defined processes. A customer will not be dumped into a voice mail dungeon and left there unless they agree that is what they want.

And not to be left unnoticed, the SIP enterprise communication platform is also serving simple service requests, known popularly as "telephone calls." The lower-level applications use the same resources used by an SOA process definition, and vice-versa, the SOA depends on the same functions of presence and connection-management provided by the SIP enterprise communication platform to make the important connections with customers.

A customer will not be dumped into a voice mail dungeon and left there unless they agree that is what they want.

Conclusion

As enterprises seek competitive differentiation, Service Oriented Architectures and Web Services enable flexible packaging of service elements to drive a vision of business personality for each enterprise. Personal connections with customers need to be natively integrated into these processes to yield increasing loyalty, and employees need to be used productively to increase business value. Both customers and employees are increasingly mobile in terms of their location and the types of devices they use to communicate at different times. Implementing a SIP enterprise communication platform early in the evolution to SOA enables backwards compatibility to installed systems — with a smooth migration to future applications with powerful customer attraction.

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SIP: Delivering the Promises of Integrated Communications

By Robert Winder

The move to IP-based contact centers continues to gain momentum as organizations recognize the inherent benefits that can be gained by consolidating multiple contact channels into a single communication infrastructure. Moving from two separate networks to a single converged network provides immediate cost reductions in both staffing and maintenance of hardware and software. Also, deploying new services requires significantly lower investments in time and money than in traditional time division multiplexing (TDM) systems.

The virtualization capabilities of IP have resulted in significant cost savings and eliminated geographic limitations by allowing agents to sit at any PC desktop anywhere in the world. This location independence dramatically alters the contact center landscape. Above all else, however, an IP-based contact center facilitates the implementation of new cutting-edge applications that can deliver significant business gains in the areas of productivity, efficiency and customer satisfaction.

Building the Framework

Key to the successful implementation of any IP telephony solution is a call control protocol that is widely accepted, easy to use and provides all the user features demanded of today's telecommunications services. Emerging as a leading contender to fit this bill is the Session Initiation Protocol (SIP), an application-layer protocol for initiating, managing, and erminating interactive sessions between one or more users on the Internet.

> Each organization needs to develop a migration strategy that maps closely to their overall goals and current infrastructure needs.

A key advantage of SIP is that it enables you to build new applications that use standard component-based equipment, which is less expensive and offers greater flexibility for scaling services compared to the TDM environment. Another SIP benefit — interoperability — is derived from the architecture of the Internet that has intelligence located at the edge of the network and devices communicating using standard protocols.

Organizations are beginning to recognize the inherent flexibility of SIP — particularly its ability to integrate well with the Web, e-mail, streaming media applications, and existing protocols used on the Internet. Such powerful integration enables rapid new service deployment while providing organizations with better continuity and higher productivity than has ever been possible before.

Easily supporting a wide array of endpoint devices and configurations, SIP helps transform the communication paradigm from a device-centric to a user-centric model. Because of its flexible user location and name mapping features, SIP plays an important role in the delivery of rich mobility services such as personal mobility, call screening, forward and transfer, and multiparty calls.

Among the capabilities available in a SIP-enabled infrastructure include:

- **Unified communications** — A SIP session can contain any combination of media (voice, data, video, etc.) and use different communication devices to help people communicate at any time and under their control.

- **IP-PBX functionality** — A SIP-compliant IP-PBX can be used in a single office or multiple office environments, offering flexibility for future expansions.

- **Web IVR** — SIP supports IVR features within a Web page environment, guiding users through options and providing auto-responses to common requests. Web IVR combines the rapid communications of voice with the ease of navigation of the Web to allow users to quickly scan the available links at any level.

- **Instant Messaging (IM)** — With support for instant messaging built into the SIP protocol, SIP makes it possible to promote an IM session to a telephone call or video session at the click of a button. With integrated IM, agents can communicate with supervisors and experts while remaining on the call with the customer. This helps reduce talk times and allows agents to serve customers with greater speed, accuracy, and efficiency.

- **Presence** — The value of presence to a contact center is clear, and when coupled with voice, becomes even more compelling. Rather than presence being a simple “on” or “off,” indicating the ability to receive instant messages, it can reflect a person’s readiness and ability to communicate using a variety of different means.

- **Mobile Phones and PDAs** — Lightweight SIP client software can be embedded in mobile phones and PDAs so that new and existing services can cross all platforms. By extending intelligence out to devices like PDAs and cell phones, organizations will be able to create a virtual contact center that can effectively leverage a company’s resources to their fullest.

The SIP standard only defines the signaling protocol for establishing and controlling sessions; it does not define how applications or features should be built, or how they should be delivered. This opens up the application development process, allowing applications from independent software vendors with expertise in specific vertical markets, spurring greater innovation.

Maximizing the Capabilities of IP

One of the most significant challenges facing organizations considering the move to IP is how to leverage existing systems. More specifically, they must be convinced that they will not only gain new features and functionality, but they will not lose the old applications contact center users enjoy with TDM and advanced CTI routing, including reporting and agent productivity applications. One way to overcome this challenge is to migrate to IP slowly.

The idea of location-independent services in the network means that a company can use a building-block approach for migrating its communications to IP on a site-by-site, group-by-group, or application-by-application basis. Because IP runs on a standards-based SIP infrastructure, it can be seamlessly integrated into the contact center, and organizations can easily make the transition in a phased approach, bringing value to the customer at each new phase.

> As an open protocol, SIP promotes interaction and involvement between individuals and companies in shaping how IP communications can grow and evolve.

Key to a successful phased roll-out is identifying those users who can show a near-term benefit in terms of improving business processes and targeting them for early implementation. This enables you to roll out new capabilities first to those people that will see the most immediate benefit, and migrate to others over time. For example, opening a new center in a low-cost region might present a more compelling proposition with IP technologies.

Each organization needs to develop a migration strategy that maps closely to their overall goals and current infrastructure needs — one that can create a best-of-breed communication solution with reduced total cost of ownership and without sacrificing existing capabilities. The migration process can be initiated at any time and transitioned at whatever pace is desired. The good news is that the benefits of SIP can be leveraged even with a small initial deployment.

A key point to keep in mind when making a decision around IP technology is that it’s not an all or nothing proposition — that is, there’s no need to migrate your entire environment to a single-vendor IP solution to begin leveraging the advantages of IP. The beauty is that by choosing vendors that support an open standards-based approach, an enterprise can deploy multiple technologies from multiple vendors in different places and use SIP to enable all of these pieces to work together.

By designing a strategy around business needs and taking a phased approach, you can ensure that existing investments are not compromised, and that the implementation will not disrupt current business processes. SIP provides the foundation that allows you to do everything on SIP that you can do on TDM, while establishing the framework to build next-generation applications.

As an open protocol, SIP promotes interaction and involvement between individuals and companies in shaping how IP communications can grow and evolve. While many companies are looking at IP telephony as simply a cost-effective replacement to what they already have, its true value lies in the integration of IP technologies like Web, e-mail, and instant messaging and allowing these to become integral components of new voice services.

This openness of the protocol is expected to continue to drive development of new and innovative products and services. As long as applications that provide real business benefits running on SIP are available now, the technology has the potential to fulfill the vision that integrated communications has long promised.

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Q & A

60 Seconds with Ken Osowski

Ken Osowski

I had the chance to speak with Ken Osowski, VP of Product Management and Marketing at Pactolus regarding the recent Frost & Sullivan statement about SIP becoming a de facto standard for VoIP.

That statement:

According to a recent Frost & Sullivan report, SIP is fast becoming the widely accepted standard for Voice over IP that allows convergent multimedia/multi-modal communication. The author of the report says, "SIP is anticipated to replace the traditional modes of communication, transforming IP communications, creating an alternate communication industry and reducing network elements to mere call-forwarding devices."

I asked Ken to share his thoughts and his company's position regarding those comments, and if he agreed or not. And of course, the editor's favorite question: Why?

Ken: We agree. So far, the industry has focused on voice service delivery via SIP, the consensus – increasingly backed by investments — is that multimedia content will increasingly complement voice with video, and that IM and other real-time communications will be increasingly SIP-based. There are several factors driving SIP-ification, one of the most important being its agility. SIP's light weight and easy to integrate into client software and devices, and at the same time allows sophisticated services to be brought into the core of the network, so it readily scales up and down.

Greg: So, as SIP is more widely embraced, what are some of the pitfalls that might stand in the way of its proliferation?

Ken: One potential obstacle to SIP's widespread integration is proprietary peer-to-peer voice services, which today don't implement it and can potentially impact its acceptance. Applications such as Skype and others use their own signaling mechanisms. In fact Skype embeds their own voice codecs, which eliminates licensing royalties and gives them the flexibility to deploy free peer-to-peer VoIP. I believe that the industry will recognize that widely available free or inexpensive codecs is a good idea and probably here to stay. But instead of throwing the baby out with the bath water, the industry should strive to drive the uniform adoption of a standard signaling protocol. In essence, SIP should be the common denominator for all usage models, sort of a lingua franca. Otherwise, incompatibilities will quickly degrade the advantages of both proprietary peer-to-peer and other VoIP communications. IMS will also drive SIP into wireless handsets, and mobility is incredibly compelling to consumers. In many ways, SIP gives peer-to-peer services the power to evolve to all networks.

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