SIP — Winning The Battle For VoIP

www.sipmag.com Volume 1 • No.1 January 2006

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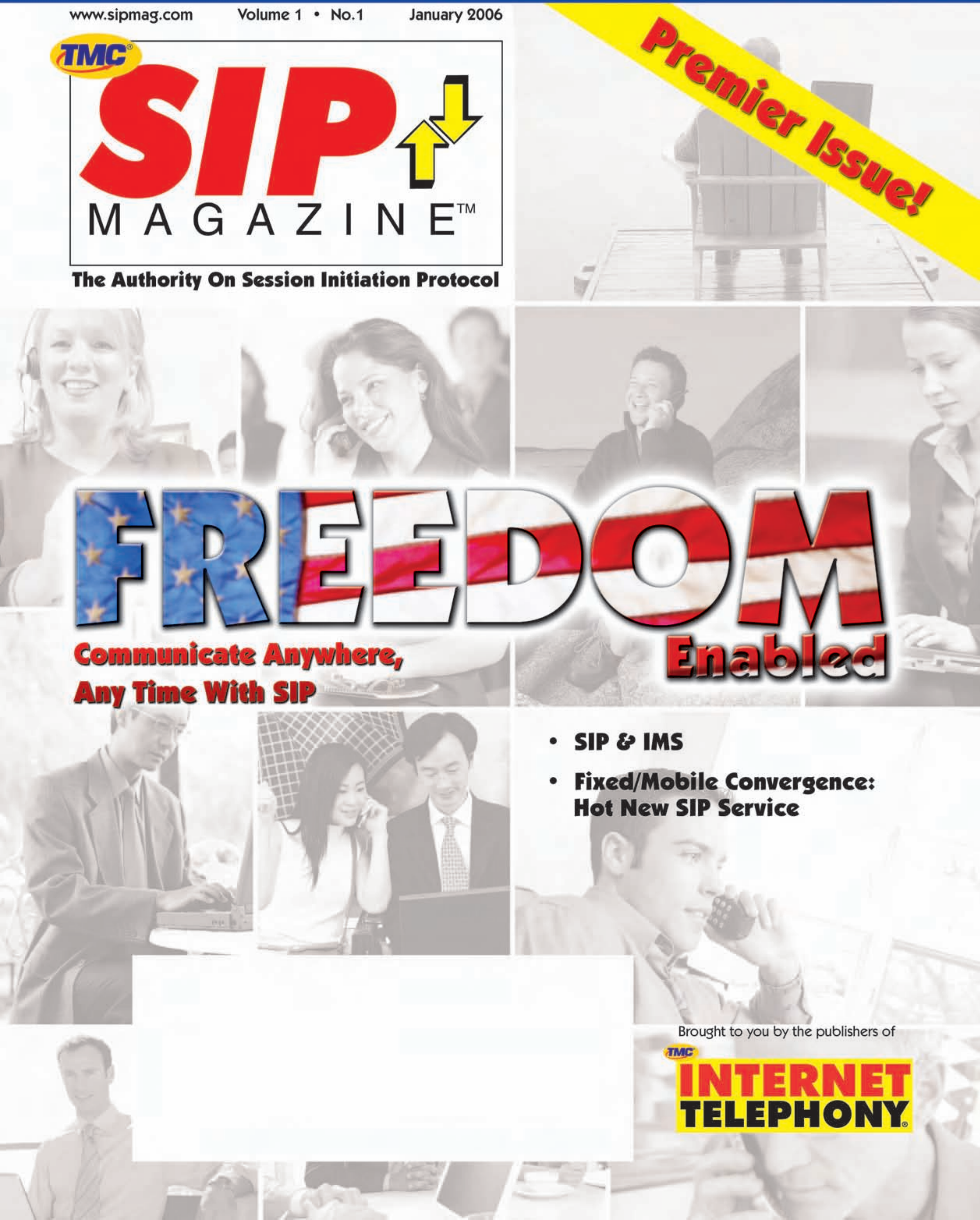
The Authority On Session Initiation Protocol



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Welcome to *SIP Magazine*!

EDITOR'S NOTE

by

Greg Galitzine

It's been a long time since I've used the term (r)evolution; in fact it was almost nine years ago when we launched *Internet Telephony*. And for all the changes that we have witnessed in the past decade, today we find ourselves in the midst of a telecommunications (r)evolution.

According to our good friends at the SIP Center, "…the Session Initiation Protocol (SIP) is a fundamental building block that service providers can use to harness the power of the Internet Protocol and transform their traditional revenue streams."

Incredibly, this single building block is at the center of all the amazing innovation and transformation taking place in telecom. SIP is at the core of the very services that will generate revenue for carriers, create opportunity for developers, and help enterprises and end users of every stripe to harness the power of IP and wring new levels of efficiency out of their communications systems for years to come.

As we embark on a new journey with the launch of SIP Magazine, we promise that we will remain true to our mission of educating and informing our constituents, alerting and reporting the key facts and trends that readers of this magazine will be able to use to make the decisions they need to make as they move forward with their communications plans.

SIP (define - news - alert) enables application developers to write to a common standard, enabling services to run on multiple platforms, across disparate networks. SIP underlies the fabric of a movement with massive potential to transform the communications space: the adoption of IMS or IP Multimedia Subsystem. SIP enables so many of the innovations that we will come to take for granted someday. We pledge to be the resource you want to turn to first for information regarding this exciting space.

I'd like to take a moment to thank the distinguished individuals who have agreed to join our editorial advisory board, as experts who will help drive the conversation forward through various levels of participation in creating the most compelling content for you, the readers of SIP Magazine:

- Erik Lagerway, Independent Consultant
- Kenneth Osowski, Pactolus Communications Software (news - alert)
- Jonathan Rosenberg, Cisco Systems (quote - news - alert)
- Henning Schulzrinne, Columbia University; SIPquest (news - alert)
- Richard M. Williams, Connect2Communications (news - alert)

And, lastly, at SIP Magazine we invite our readers to participate in the conversation. I encourage you to e-mail me with any questions or comments you might have regarding the industry, the technology, the publication, what have you… Complaints and compliments are equally welcome.

– Greg Galitzine

SIP MAGAZINE™

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

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Circulation Director, Shirley Russo, ext. 157 (srusso@tmcnet.com)
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SIP Magazine® encourages readers to contact us with their questions, comments, and suggestions. Send e-mail (addresses above), or send ordinary mail. We reserve the right to edit letters for clarity and brevity. All submissions will be considered eligible for publication unless otherwise specified by the author.

Identification Statement
SIP Magazine® (ISSN: 1098-0008) is published bimonthly by Technology Marketing Corporation, One Technology Plaza, Norwalk, CT 06854 U.S.A. This issue, Volume 1, Number 1 is dated January 2006. Annual print subscriptions: free, U.S. qualifying readers; $29.00 U.S. nonqualifying, $39.00 Canada, $60.00, foreign qualifying and nonqualifying. Periodical postage paid at Norwalk, CT and at additional mailing offices.

Postmaster: Send address changes to: SIP Magazine®, Technology Marketing Corporation, PO Box 21642, St. Paul MN 55121 U.S.A.

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A Technology Marketing Publication,
One Technology Plaza, Norwalk, CT 06854 U.S.A.
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SIP MAGAZINE™

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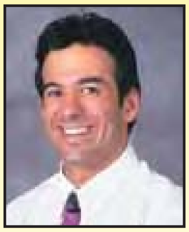

by
Rich Tehrani

How often have we found ourselves in a seemingly endless search for a scrap of paper, a business card in a Rolodex, or an old email in hopes of finding a phone number or email address for a colleague or old friend? More importantly, how much time and energy do we expend looking for them?

We are now on the verge of entering a world of united communications where we will be able to reach each other more effectively than ever before, regardless of device, operating system, or network. We will simply communicate without the need to worry about scrambling for those lists of phone numbers or other unique identifiers tied to today's soon-to-be outdated devices and networks. Because presence data will allow us to know when the person we are trying to reach is available, we won't waste time trying to reach them until we know we can. Likewise, we won't be constantly interrupted during meetings and other important events if we choose not to be.

SIP will prove to be the standard protocol for human communications of the future.

The new world of communications will be much better than that which we know today. We will be happier. We will be more productive. We will take control of communications — not the other way around. One of the engines driving this future of simplified and more effective communications is SIP.

SIP — short for session initiation protocol — is already being used extensively in instant messaging, voice, and video conversations. Furthermore, it is at the core of IMS (IP Multimedia Subsystem), an architecture designed to allow wireless and wireline service providers to provide next generation services in an integrated fashion.

SIP (define - news - alert) is the protocol that helps us be accessible anywhere and everywhere — and on our terms. It allows us to stay connected to corporate PBXs, regardless of our location, and it lets corporations purchase SIP-based VoIP trunks to connect to their PBXs, eliminating the need for traditional voice T1s and other related equipment.

SIP will prove to be the standard protocol for human communications of the future. It is a peer to peer protocol, meaning it does not need a centralized server to work. But it can interface with such a server, if needed, in a service provider implementation, for example.

SIP is powerful and SIP is complicated. SIP can drastically reduce communications costs and increase communications speed and efficiency.

As its name suggests, this magazine is devoted to the world of SIP. Our aim is to educate you, our reader, on the latest developments regarding SIP and to help you understand how the products on the market can help you save money, make money, and become more efficient, depending on your needs.

TMC has been publishing magazines in the technology space since 1972. We launched the first magazine in the call center space in 1982, Customer Interaction Solutions; the first magazine in the VoIP space in 1998, Internet Telephony; and now the first magazine in the SIP space. Thank you for you loyal readership over the decades. We promise that, in the TMC tradition, this magazine will be the objective voice of SIP and will help you to do your job better.

So, what better way to get the ball rolling on this first issue than to find out from some of the thought leaders what their thoughts are in this space. We offered these knowledgeable members of the SIP community an opportunity to weigh in on a series of questions — about the future of SIP, the impact that widespread adoption has had and will have on their product plans, and, perhaps most importantly, how SIP benefits the end user. For your perusal, deliberation, and reaction, their comments follow.

Where is the SIP market headed?

Anjali Gupta, Flextronics: SIP has become ubiquitous in the last couple of years for developers of next generation networks. The industry has witnessed the emergence of a new breed of alternate carrier — which is growing at a tremendous pace — by betting on IP for providing innovative value added services. This will continue to fuel unprecedented growth in SIP, at least for the next couple of years.

Henning Schulzrinne, Columbia University: Now that


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PUBLISHER'S OUTLOOK

almost all telecom-related vendors offer SIP equipment, there are probably three growth areas: IMS, 802.11, and IM/presence. All of these have some question marks, but at least the technology for them is now becoming commercially usable.

Ian Colville, Aculab: (news - alert) The SIP market is, undoubtedly, headed for a bright future, having gone well beyond a 'heralding of the dawn' phase. The broad industry commentary on protocols for use in next generation networks is focused almost exclusively on SIP. There are areas of concern, such as security, which provide opportunity for innovation — and solutions.

Jeff Ford, Inter-Tel: Inter-Tel (news - alert) believes that SIP will become the dominant standard for IP stations, gateways, and applications in the next generation enterprise communications network. It will enable businesses to fully leverage emerging third-party applications and endpoints, as well as next generation multimedia business enhancement tools, such as collaboration, conferencing, and presence management.

Ken Osowksi, Pactolus: (news - alert)SIP is formulating the multimedia story for IMS-enabled networks. It will become the core signaling/event notification protocol for all real-time media — voice, video, messaging, presence events, multimedia messaging — that never before had been wrapped in a single application framework. All embodied interfaces, such as MMS and SMS, will be consumed by SIP, from the core network to the handset.

Al Brisard, Pingtel: (news - alert) SIP is taking enterprise communications far beyond its origins in voice. SIP is creating a robust enterprise communications framework that enables people to communicate in real time, whether by voice, video, or IM. SIP will be the unifying protocol that will blend communications and information to create increased value. We are at the beginning of a market transformation driven by the adoption of SIP and that will change how products are developed, packaged, and consumed at every level.

Peter Brockmann, SIPquest: (news - alert) SIP will dominate IP telephony for five reasons: First, SIP is simple and easy to debug. Second, SIP is standard, namely IETF standard, so there is a wide base of standards that it depends on, and that depend on it. Third, SIP is secure (see #2). Fourth, SIP seamlessly crosses the enterprise-service provider divide. Lastly, SIP applications abound since developers can build for either market with ease. Check out my whitepaper on this: http://tmcnet.com/234.1.

What are your expectations or estimates for the growth of the SIP market?

Ken Osowksi, Pactolus: For 2006, expect that incumbent service providers and carriers will start their migration off of TDM-based services platforms and move to SIP-based services and endpoints. This growth will continue to be fueled by increased competition from Tier 2 and 3 players that have SIP-based network approaches at the core of their business models. This market dynamic will force all players big and small to adopt SIP.

Peter Brockmann, SIPquest: We believe that the SIP market will explode, with phenomenal growth over the next three years. SIP will be as much a harmonizer in VoIP as IP was to the data network a decade ago.

Al Brisard, Pingtel: SIP has already begun to reshape the communications market as evidenced by its rapid adoption and offerings at every level. SIP has significant momentum to a point that it will not be stalled or derailed by key players that depend on their proprietary models to lock in their customers at ridiculous prices.

Jeff Ford, Inter-Tel: We feel that SIP will see modest growth over the next two years. However, the real growth should come into play sometime in 2008, when we should begin to see SIP emerge as the dominant interface. Depending on a number of factors, including the development of third-party applications and devices, we could expect this escalation to extend another four to five years.

Anjali Gupta, Flextronics: (news - alert) SIP will continue to enjoy broad adoption, at least for the next few years, and is expected to witness a solid double-digit growth. The current SIP-enabling technologies market is estimated at \$33M and expected to grow beyond \$100M in the next five years. Though enterprise segment was the first to witness broad adoption, SIP network infrastructure market as a whole, currently estimated at \$4B is expected to grow beyond \$10B by 2009 with both incumbent carriers and CLEC's embracing SIP.

Henning Schulzrinne, Columbia University: It is pretty clear that lots of companies are now transitioning from the evaluation and experimentation to the purchasing phase.

Ian Colville, Aculab: Figures abound for forecasts, all of which confirm SIP is in the ascendancy. Aculab products benefit from an integrated SIP stack offered under a cost free license, which gives customers a great boost and contributes to the growth of the market.

PUBLISHER'S OUTLOOK

How does the development of SIP affect your product plans?

Ian Colville, Aculab: SIP is a central pillar of Aculab's IP-centric product plans. Aculab offers a feature rich SIP stack and will continue to invest in enhancements in line with demand and the evolution of the protocol. Together with Prosody X, our SIP offering enables a wide variety of telco and enterprise communications solutions that bring many advantages.

Peter Brockmann, SIPquest: SIPquest was founded to exploit applications of SIP. SIP is an integral component of everything we do. We use SIP in our multimedia suite for Audio, Video and Data Conferencing, the Desktop Console application and our Mobile Console product for mobile WiFi devices, WiFi and Cellular devices and IMS devices.

Jeff Ford, Inter-Tel: Inter-Tel has supported SIP in our product lines since February, 2003. In fact, Inter-Tel is continuing its thought leadership by continuing to develop and release new SIP-based products and applications that will provide value and improve business processes.

Anjali Gupta, Flextronics: The broad adoption of SIP enables Flextronics Software Systems (FSS) to further strengthen its position as the leading SIP vendor, and introduce innovative products, both for OEM's and carriers. These ingeniously designed products enable OEM's to roll-out innovative solutions that can be rapidly deployed thereby facilitating SIP adoption.

Ken Osowksi, Pactolus: As SIP pioneers, we're past baseline incorporation issues, so two product areas stand out: First, incorporating advanced user interactivity, call handling, and rich billing options into voice services, heightening the user experience and reducing their costs, and second, helping service providers achieve broad cross-market/subscriber voice service reach. For example, enabling long-haul IP network operators to deliver wireless services and leverage their backhaul capability.

Al Brisard, Pingtel: Pingtel sees SIP as the linchpin for enterprise communications going forward. Pingtel will be at the forefront of SIP developments and will lead in key areas that are most important to customers in the coming months such as security and presence. And this development will always be done in an open, standards-based way.

How do your customers benefit from SIP?

Jeff Ford, Inter-Tel: Like all industry standards, SIP allows our customers to take advantage of best-of-breed applications and devices and build the best possible solution for their business. In addition, standards-based environments serve as catalysts for developing many creative and innovative third-party solutions.

Al Brisard, Pingtel: In the end, customers care about three things: value, cost, and quality. SIP combined with open source development, enables customers to get what they want, from the vendors they want, at a price they want to pay, with the quality and flexibility they require. SIP is the catalyst that will provide application unification tied to real time communications.

Anjali Gupta, Flextronics: SIP has become the de facto standard. It provides better value for end customers by providing imperatives such as multimedia, presence, device independence and instant messaging capabilities. As a result, carriers can increase their efficiency by migrating to SIP and provide an enhanced user experience to reduce churn and increase ARPU.

Ken Osowksi, Pactolus: As devices, intermediate networks, and core networks all become SIP-based, everything becomes more functional and affordable for both service providers and consumers as proprietary devices and single function networks become obsolete. Specifically, while all cell phones today have Java applets, there'll be new events and applications that are driven by 3rd parties directly to the consumer.

Henning Schulzrinne, Columbia University: I think the largest benefit is the ability to mix-and-match systems. For example, we are locally running a commercial SIP PBX system for the department that combines at least three brands of "hard" phones and several soft phones.

Peter Brockmann, SIPquest: SIPQuest customers are assured of great software products with rapid debugging, rapid integrations, secure implementations, and scalable deployments.

Ian Colville, Aculab: Quite apart from the obvious value inherent in an Aculab SIP solution, which helps to protect margins in a competitive arena, the feature rich nature of our API enables customers to use SIP for more than basic call control – third-party call control and the efficient use of media resources being good examples.

CommuniGate Systems Announces Availability of SIP Farm to Deliver Industry's First Highly Resilient Carrier-class and Scalable VoIP Framework

CommuniGate Systems (news - alert) is launching its new SIP Farm, patent-pending technology on the CommuniGate Pro Dynamic Cluster application server platform. The technology enables large scale deployments of SIP-based VoIP to telco-level capacities. The multi-node, all-active SIP Farm technology makes it possible to host millions of subscribers with either a consolidated cluster or distributed geographic placement of cluster members while providing 99.999% uptime.

CommuniGate Pro 5.0 SIP Farm removes all barriers to adoption of SIP as a standard, making it possible to scale to tens of millions of subscribers with redundant signaling. The complex nature of real-time communications has strained the database-centric architectures of the past. Carriers have found it hard to manage systems not designed for the sheer load that IM presence status information and multiple endpoints per user on the system.

"We're introducing technology today that eliminates the pains of scaling SIP technology for sites with tens of millions of subscribers. The lack of redundancy or susceptibility to signaling failure is the result of limited architectural approaches in most SIP systems," stated Thom O'Connor, Director of Product Architecture, CommuniGate Systems.

"With CommuniGate Pro SIP Farm, redundancy and capacity is expanded in one or even multiple locations acting as a single cluster, with the ability to add or remove nodes from the cluster and providing immediate re-allocation of sessions in the event of system failure.

http://www.communigate.com

SIPfoundry Releases Version 3.0 OF SIPxPBX

SIPfoundry, (news - alert) a non-profit organization for the development of open source Session Initiation Protocol (SIP)-based communication solutions, released its latest open source PBX: sipXpbx. This release delivers a fundamentally new approach to addressing the real-time communications needs within the enterprise market. This new release of a complete SIP PBX that handles real time communications is available to users as a single file download.

Building on the sipXpbx platform, developers at SIPfoundry have expanded its functionality and capabilities to include: an advanced scalable architecture, a presence server, support for multiple and nested auto attendants, advanced call control, forwarding identification support, and interoperability support tested with dozens of SIP-compliant phones and gateways.

In addition, the system configuration and management tool was re-architected to make it more robust, efficient, and extensible to support larger enterprise installations and make it easier for third-party manufacturers to integrate features and functionality. Manufacturers of endpoints can now easily add support for their products in just a few hours, to enable complete provisioning and management of their devices. In this release, complete provisioning and manageability of AudioCodes gateways and Polycom, SNOM, and Grandstream phones were delivered.

http://www.sipfoundry.org

Interpeak and TeleSoft to Offer SIPNET

Interpeak (news - alert) and TeleSoft International (news - alert) announced a collaboration focused on the promotion of combined technologies for the consumer electronics, mobile handset, and mobile multimedia

markets. The companies will create a new product for the embedded market, SIPNET, which will combine TeleSoft's CompactSIP and Interpeak's IPNET.

This joint effort leverages each company's strengths and expertise: Interpeak's networking, security and wireless technologies and TeleSoft's flexible, full-featured and compact SIP. The horizontal platform will have potential across a range of embedded products — including set-top boxes, handsets, consumer electronics, PDAs, and other multimedia devices — and applications, including instant messaging and both wireless and wired VoIP.

As networks converge on an IP infrastructure, due to the widespread availability of broadband access and advent of rich multimedia-based communications, SIP will become the primary protocol technology for signaling, session capability negotiations, and the establishment of multimedia-based communications. SIPNET will stand ready to meet the increased SIP traffic, with the requisite secure and mobile components as well.

http://www.interpeak.com
http://www.telesoft.com

Antepo Integrates Presence and Telephony With VoIP Upgrade for OPN System

Antepo, Inc. (news - alert) unveiled the first preview of its upgrade for OPN System, code-named "Rivoli" which adds voice and video capabilities to its award-winning EIM and Presence Management server. The Rivoli release further establishes OPN System's native support for SIP (Session Initiation Protocol) to offer broad integration with SIP-enabled services, devices, and applications.

With OPN System Rivoli, enterprises and service providers can seamlessly integrate IP-based telephony capabilities with enterprise-wide Presence management across a range of multimedia communications systems and services. The module implements a complete SIP registrar and proxy server, so users can leverage a range of SIP and SIMPLE-based endpoints to make multimedia calls over the Internet — and to generate and manage Presence through OPN System.

For service providers, Rivoli works with CounterPath's award-winning eyeBeam client to offer a robust, scalable Presence engine to deliver voice-centric communication offerings.

For enterprises, Rivoli offers the first and only alternative to Microsoft Live Communications Server for the company-wide integration of IP-based telephony systems, and the addition of advanced Presence management capabilities to a full range of voice, video, and IM communications.

http://www.antepo.com

Brekeke Software Announces Coming Release of OnDO PBX v1.5 With ARS Failover

Brekeke Software, Inc. (news - alert) announced the coming release of their OnDO PBX version 1.5, an enhanced IP-PBX software telephony system, which now supports Automatic Route Selection (ARS) failover.

With ARS failover, OnDO PBX seeks an alternate route if the specified route is unavailable, and makes outgoing calls via the best route based on the situation. The benefit of ARS failover is improved efficiency and reliability of your telephony system.

Additional benefits include increased ease in creating connections between OnDO PBX and SIP products and services. Setting up routing rules enable ARS failover to select from all available routes. ARS Failover also allows for SIP products and services which are unavailable, to be considered inactivated for a specified period. The route is re-activated based on automatic recovery settings or by manual re-activation through the administration tool.

"Future opportunities for VoIP technology using SIP continue to grow. For example, if a natural disaster disabled PSTN lines, VoIP communication by satellite has been recognized as a method for re-establishing communication systems," states Shin Yamade, President and CEO for Brekeke Software.

http://www.brekeke.com

Excel's Integrated Signaling and Media Gateway Enhanced to Allow Any-to-Any Voice Network Connectivity

Excel Switching Corporation (news - alert) announced the availability of SIP and ISDN functionality in an enhanced version of its popular IMG 1010 integrated signaling and media gateway. For service providers that are planning to bridge TDM and IP networks, or connect two IP networks, the IMG 1010 provides new levels of density, integration, and transcoding on a future-proof platform that can evolve as services evolve.

A carrier-grade VoIP (define - news - alert) gateway that offers any-to-any network connectivity, Excel's IMG 1010 has been enhanced with SIP and ISDN functionality to enable service providers to quickly and cost-effectively add new VoIP capacity, while simultaneously providing a clear migration path to an all-VoIP environment.

The IMG 1010 supports SIP, H.323, SS7 and ISDN simultaneously, all-in-one compact 1U design that can accommodate up to 768 channels, for the ultimate in flexibility. As a result, the IMG 1010 sets a new standard for simplicity, versatility and cost-effectiveness in the media gateway market, offering a solution that empowers service providers to seamlessly transition from first generation, network-based transport-oriented services to the subscriber-driven multimedia SIP applications found in next-generation IP-based services.

http://www.excelswitching.com

TotallyFreeConferenceCalls Selects Vapps SIP-Based Platform to Deliver Next Generation Conferencing Services

Vapps, (news - alert) a leading global provider of audio conferencing systems, announced that TotallyFreeConferenceCalls has chosen the Conference Bridge 1000 (CB1000) platform as the foundation for delivering next-generation conferencing services. Together with MetaSwitch's fault-tolerant Class 4/5 softswitch architecture for bridging legacy circuit switched and IP Multimedia Subsystem (IMS) networks, TotallyFreeConferenceCalls can now integrate service across multiple network types to reach new markets and

expand its customer base.

A VoIP-native IMS application server network element, the CB1000 is a SIP-enabled, carrier-grade conferencing platform that delivers robust, seamless reservation-less conference calls on both legacy and IP-based telecom systems. CB1000 supports up to 18,000 total conference participants in multiple simultaneous conferences and offers both standard and advanced calling features including: participant lists, volume control, mute, multiple languages, call detail records and web-based on-demand feature control.

"CB1000 was selected based on the platform's rich features, SIP interoperability and carrier grade reliability," said Todd Zweig, General Manager of TotallyFreeConferenceCalls. "By allowing us to bridge traditional PSTN calls with the ubiquity and flexibility of IP communications, we are able to offer customers exciting new services with lower associated operation costs."

http://www.vapps.com

TelTel Launches High-Quality PC to Any Phone, Anywhere Calling Via TelTel's Managed Peer-to-Peer SIP Backbone

TelTel, (news - alert) a provider of SIP-based global Internet telephony, today announced the launch of TelTel-Out, which delivers high-quality PC to any phone, anywhere calling, and TelTel Call-Forwarding, a new call forwarding service that allows forwarding of incoming calls to any PSTN number or SIP URL. TelTel-Out and TelTel Call-Forwarding are both now available to TelTel's over 1.3 million users and through service providers via its popular SIP Virtual Network Operator (SVNO) program.

TelTel-Out provides better quality calls than traditional PSTN line calls. The new services use TelTel's managed SIP-based peer-to-peer backbone to optimize call routing and deliver crystal clear calls to virtually anywhere in the world.

TelTel's popular SIP Virtual Network Operator (SVNO) program allows service providers and enterprises to immediately offer VoIP services like TelTel-Out and TelTel Call-Forwarding that are second to none in quality of service. This program has been successfully deployed worldwide and has proven to be a strong revenue generator for both enterprises and service providers.

http://www.teltel.com

Linksys Announces Sip-Based IP PBX, Desktop Phones, and Gateway for Internet Telephony Service Providers

Linksys, (news - alert) a Division of Cisco Systems, Inc., the recognized leading provider of voice, wireless and networking hardware for the consumer, Small Office/Home Office (SOHO), and small business markets, today announced a new line of SIP-based telephony products for Internet Telephony Service Providers (ITSPs) targeting large residential, SOHO, and very small business customers.

The new line of IP communication solutions includes an IP PBX/Key system, a wide range of IP desktop phones and an Analog Gateway for connection to the Public Switched Telephone Network. Used together with an ITSP voice service, they provide a complete IP telephony system for up to 16 users.

"With the new Linksys SIP-based IP communication offerings, ITSPs can offer residential and small businesses a voice service with many of the features found in large business voice IP networks, such as multi-line service, music on hold, auto attendant, and more at a more


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affordable price," said Jan Fandrianto, vice president of voice engineering at Linksys. "The new IP PBX and IP phones bundled with a service provider offering will make the deployment of voice networks easy to install and simple to use at a price small businesses can afford."

This new solution will complement the recently announced Linksys One solution. Linksys One is an ideal solution for small business with 5-100 users needing a complete communications solution that addresses voice, video, and applications. The LVS series was developed to address the residential and very small business of 1-4 users that may grow to 16.

http://www.linksys.com

Cedar Point Expands Safari C3 SIP Capabilities to Enable Deployment of Advanced Voice Features

Cedar Point Communications, (news - alert) the leader in integrated packet-based voice switching technologies for the cable industry, today announced new capabilities of its SAFARI C3 Multimedia Switching System that will enable cable system operators to deploy Session Initiation Protocol (SIP)-based multi-vendor advanced voice communication features.

The expanded functionality will enable interoperability between SAFARI C3 and a wide range of feature servers, allowing cable system operators to leverage such third-party applications as messaging, IVR and call attendant.

"Competition among traditional wireline carriers, third-party VoIP services and cellular providers has created an increasingly cluttered telephony landscape," said Dave Spear, executive vice president, strategy and market development for Cedar Point Communications. "By implementing SIP triggers to support third-party call control, we will allow our customers to provide real differentiators in their service offerings, ranging from day-to- day voice features to fixed-mobile convergence."

"One of cable's voice strengths today is its unique ability to combine both carrier-class PacketCable services that equal or exceed the quality and reliability of incumbent providers and SIP-based features that provide operators and providers with unparalleled flexibility," said Andy Paff, president and CEO of Cedar Point. "As we and the industry move toward an IP Multimedia Subsystem (IMS) architecture in the future, operators will be able not only to implement third party services, but to combine those services to create entirely new offerings. The expansion of SIP capabilities allows us to ease the migration to IMS networks."

http://www.cedarpointcom.com

SIP-Based Software Phone Brings Theora Open Source Video Codec to VoIP Community, Developers

Ecotronics Ventures LLC (news - alert) announces a new release of its Kapanga Softphone, now supporting Xiph.org Foundation's Theora, the open source video codec based upon On2 Technologies' VP3 codec. Kapanga Softphone extends the set of tools available to the VoIP community by integrating Theora into its SIP-based video capabilities

Kapanga Softphone is the premier VoIP softphone that integrates voice, video, and fax over IP into one software phone. Using the Session Initiation Protocol (SIP), Kapanga Softphone interacts with any VoIP infrastructure, such as VoIP phone service providers, PBX systems for enterprise and small businesses, and other VoIP-based systems.

"Since our launch this year, many Kapanga users have used our SIP client to extend their open source PBX deployments", said Martin Cadirola, President and VP of Business Development for Ecotronics. "We believe we can help speed up the development of open source PBX video functionality by providing a Theora-capable VoIP-based software phone", he added Kapanga also features other open source audio codecs, like Global IP Sound's iLBC and Xiph Foundation's Speex.

http://www.kapanga.net

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8x8 Expands VoIP and Videophone Service Offerings with CounterPath-Powered Packet8 Softalk

8x8, Inc., (news - alert) the Packet8 broadband VoIP and videophone communications service provider and CounterPath Solutions, Inc. unveiled 8x8's latest VoIP service, Packet8 (news - alert) Softalk, a co-branded, video-enabled SIP softphone for use with the Packet8 residential, video and Virtual Office Internet phone services.

Developed using CounterPath's software development kit (SDK), Softalk enables subscribers to make and receive voice and video phone calls directly on their personal computers without the need for a regular analog phone. All that is required is a desktop or laptop personal computer running Microsoft Windows with built-in or external speakers and microphones or a computer headset.

Packet8 Softalk(TM) will be offered in two plans, Basic and Premium. Packet8 Softalk (TM) Basic allows unlimited IP to IP ("in- network") voice and video calling between Softalk users and existing Packet8 voice and videophone subscribers free of charge. There is no activation or disconnect fee for the Basic service.

"With the launch of Packet8 Softalk, we are not only providing potential new customers with an opportunity to test Packet8 service, we are also giving our existing Packet8 subscribers an easy way to access their service even while they are away from the home or office," said 8x8 Vice President of Sales & Marketing Huw Rees. "Packet8 Softalk can be downloaded to a laptop so that when a subscriber travels all they need to do is go online to make phone calls and get all the great benefits of the Packet8 service."

"8x8 has built a very compelling value proposition for its customers," said Donovan Jones CounterPath's Vice President Sales & Marketing. "As business and residential customers maximize the benefits from IP communications, softphones which allow for high quality Voice, Video and IM, will become a critical component of a service provider's offering.

Through our licensing agreement, we are looking forward to further augmenting 8x8's service offerings with high quality, rich, robust applications."

http://www.8x8.com
http://www.packet8.net

MiRS Deploys Fixed Mobile Convergence Solution From Tekelec

Tekelec (news - alert) announced that its core convergence switching solution has been selected by MiRS, Israel's integrated digital enhanced network (iDEN) wireless operator.

The solution enables MiRS' transition to the Internet protocol multimedia subsystem (IMS) architecture and enables next-generation capabilities, including fixed mobile convergence (FMC) and hosted session initiation protocol (SIP) services. Tekelec's IMS architecture drives switching for both MiRS' entire mobile device line and SIP-based clients, and it supports the execution of common network applications.

"We selected Tekelec's convergence solution because of the company's leadership in deploying proven, carrier-grade next-generation switching solutions around the globe," said Shaul Shmaya, MiRS' director of engineering.

"Tekelec's next-generation switching products provide real-world solutions for MiRS and other global operators to migrate to IMS-based architectures and offer the most advanced services while reducing costs," said Tricia Hosek, president and general manager of the Tekelec Switching Solutions Group. "Our fixed mobile convergence solution lets operators secure a leading position in lucrative new markets."

Dr. Ayal Itkovitz, CEO of Convergin, added, "These market-leading convergence solutions help mobile carriers add VoIP and IMS services. By enabling the convergence of SIP and cellular, MiRS is well positioned to capture additional lucrative markets as well as to strengthen its position in its existing business and enterprise markets."

http://www.tekelec.com
http://www.motorola.com

TeleData Technology Introduces Cutting-Edge Unified Messaging and PBX VoIP Integration Capabilities

TeleData Technology, Inc., (news - alert) a provider of integrated telephony solutions, announces its release of version 10.4 of the T3 Engine, including expanded PBX integration for VoIP, enhanced unified messaging capabilities, and a newly designed Web-based fax interface.

Version 10.4 continues to expand upon TeleData's vision to provide a comprehensive telephony solution developed on the 'building blocks' approach. This unique platform-independent approach allows TeleData to offer highly customized telephony applications to its customers by activating or deactivating specific services, and providing tools for further customization that don't require in-depth telephony programming knowledge.

"For years we've been developing a clientless unified messaging environment to support our customer's security policies and service packs updates, and we now see a shift in the industry that validates this approach," states Yaniv Livneh, CEO of TeleData. "Our latest release continues to provide this flexibility to our clients, while integrating voice into the email environment in innovative ways. Version 10.4 of the T3 Platform provides

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our clients with the ability to develop highly customized telephony services that can be rolled out across a wide variety of technology environments, including VoIP."

Release 10.4 fits effortlessly into a VoIP or traditional TDM PBX strategy with the addition of a SIP-enabled platform. The T3 Platform, combining feature-rich messaging functionality with automated attendant, speech recognition, multi-modal (unified) messaging, interactive voice response, fax and networking capabilities, is now offered to customers as a traditional TDM, SIP-enabled, or a hybrid TDM/SIP solution. By enabling TDM and SIP in the same platform, TeleData provides flexibility to customers who need to share messaging and application services across a mixed TDM and VoIP PBX environment.

http://www.myt3.com

Marconi Unveils New ViPr Media Center

Marconi Corporation plc (news - alert) announced the new ViPr Media Center. The ViPr Media Center emphasizes group conferencing with a Session Initiation Protocol (SIP)-based video conferencing suite that delivers high quality video telephony over a wide range of bandwidths and provides full interoperability with standard telephone systems and legacy video conferencing systems.

ViPr Media Center is a video telephony and multimedia system that seamlessly integrates high-quality, real-time video telephony with rich conferencing and presentation tools. This newest product family offering is available in configurations that work from the executive suite or co mmand center to large conference halls. ViPr allows for natural conversations and highly productive meetings, saving costs in time and travel, and inspiring higher levels of communication responsiveness and more fully enhanced situational awareness than conventional video conferencing, while supporting up to 15 delegates on one conference.

ViPr Media Center is a multiparty video telephony platform that eliminates the Multi-point Control Unit (MCU), a costly device that many traditional video conferencing products require for multiparty calls. The MCU introduces delay that makes real-time video telephony impossible. Eliminating the MCU allows ViPr to deliver critical real-time communications such as remote hearings, where participants must see every gesture and hear every vocal nuance conveyed with digital accuracy. ViPr Media Center enables each individual user to customize their own preferences for their desktop interface, whereas systems designed with an MCU deliver a "one-size fits all" audio and video interface format that cannot be personalized at the users discretion.

http://www.marconi.com

Covergence Delivers Industry's First Complete Interoperability Solution

Covergence, (news - alert) the first company to deliver a unified security and management solution for applications and services based on the Session Initiation Protocol (SIP), announced today that it is shipping the first complete Microsoft LCS - IBM Sametime interoperability solution. The Covergence Eclipse(TM) series enables seamless communication between collaboration applications from different vendors.

Enterprises have been quick to invest in collaboration solutions to reap the benefits of increased productivity, accelerated business processes and reduced costs. The two predominant enterprise collaboration applications in the market are IBM Lotus Sametime(R) and Microsoft(R) Live Communications Server. Although both of these applications support connectivity to external collaboration communities using protocols based on SIP and its extensions for instant messages and presence (SIMPLE), they cannot interoperate directly. As a result, enterprises that have both Sametime and LCS environments are left with unacceptable "islands of collaboration" where users in the LCS community cannot collaborate with users in the Sametime community.

"Enterprises have a critical, immediate need for a solution that enables seamless communication between their Sametime and LCS environments," said Bob O'Neil, president and CEO, Covergence. "Enterprises that adopt Covergence's Eclipse solution are quickly seeing the benefits of true enterprise collaboration."

The Eclipse LCS-Sametime solution features a unique, patent-pending architecture that enables it to translate between different vendors' SIP/SIMPLE dialects and enables seamless cross-domain presence visibility and messaging connectivity in a way that is transparent to the end user. In addition, Eclipse gives organizations total management visibility and control over all cross-domain collaboration activity. Using Eclipse's management interfaces, administrators can define security, control and monitoring policies that Eclipse will then enforce on all cross-domain collaboration traffic.

http://www.covergence.com

ON THE EDGE

SIP: Lighting the Way for Communications

You know a VoIP protocol has made it to prime time when it gets its own magazine! SIP (Session Initiation Protocol) has certainly gained momentum, and it doesn't look to be slowing down anytime soon.

by
Erik Lagerway

In the past year we have seen the major IM providers like Yahoo! and MSN adopting SIP as their chosen signaling protocol for VoIP. Google has openly professed that they intend to support SIP in their new IM service. Even Apple's iChat supports SIP for VoIP and video. SIP is the torch lighting the way for VoIP; there is no doubt about it.

The thrust seems almost unstoppable with everyone racing to support open standards. For consumers, this is a very good thing as it presents choices. Networks that interoperate are very important and, if providers want any hope of peering with other networks, they will need to use open standards. SIP does all of this and more.

So, what, if, anything could slow the adoption of SIP? Firewall traversal? No, we have pretty much solved that one with use of STUN, TURN, ICE, and some intelligence built into the SIP end point.

What about Security? Yes, this could be a problem in certain networks that choose to ignore the issues.

SPIT — SPAM for Internet Telephony — is not yet a huge problem, but it has had an impact on some providers who found themselves shutting down certain services because of it. You don't hear much about it because the providers would rather not admit their networks are vulnerable.

So what are the main threats? From a consumer's point of view, there are two main concerns: protecting the identity of the user and protecting the content of the calls.

Identity Theft

In one case, a large VoIP provider in the U.S. had enormous problems with users spoofing other SIP IDs and making calls to sex lines in another country. The local carrier was appalled when the peering provider sent them their bill. This resulted in blocking service to that particular country.

Today, most SIP-enabled devices and registrars support Digest Authentication and TLS. Together, these methodologies can be used to thwart ID theft in a SIP network. It's too bad that TLS is not widely supported when connecting from SIP to the PSTN (Public Service Telephone Network), making it difficult to implement across the board.

S/MIME (Secure MIME — Multipurpose Internet Mail Extensions) is another mechanism used to encrypt individual messages, much like PGP (Pretty Good Privacy). It was primarily developed for e-mail, but it certainly can be applied to SIP for use with VoIP and IM.

What about DoS attacks? A few months back, another U.S. provider was hit with a DoS (Denial of Service) attack that consisted of many SIP requests overwhelming the network and castrating the service. The provider had to interrupt service in order to fix the problem, cutting off thousands of users in the process.

Often in a VoIP network, a user does not know the person who is calling them and may want to decide whether or not they want to take the call before answering. The IETF is close to finalizing a new security method for VoIP, called "SIP Cert," which is meant to provide a secure way to make sure that callers are, in fact, who they say they are.

More information on the following technologies can be found online:

TLS: http://tmcnet.com/230.1 S/MIME: http://tmcnet.com/231.1
SIP Cert: http://tmcnet.com/232.1 SRTP: http://tmcnet.com/233.1

Securing the Media — Encryption

Eavesdropping is an issue not only in VoIP, but also in traditional telephony. To think that someone could overhear your conversations is disconcerting to say the least.

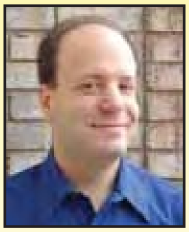
Today, VoIP networks using SIP generally use RTP (Real-time Transport Protocol) for the actual voice and video that we see and hear during a call. Encrypting this data can be done by implementing Secure RTP, or SRTP. Since SRTP was built to be very efficient, it uses up little additional bandwidth and CPU.

The bottom line is that SIP is the best standards-based protocol for building VoIP and Video services on the Internet today. The SIP community is rich with IP Communications leaders and the support from this network is tremendous. This protocol will could very well mature into a complete IP Communications framework. The battle for VoIP has been won; now what about IM? Stay tuned as SIP wages war on closed IM protocols and takes on the enterprise with SIMPLE (SIP for Instant Messaging using Presence Leveraging Extensions).

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

SPEAKING SIP

SIP and Fixed Mobile Convergence: Realizing the Component Architecture



by
JD Rosenberg

As the co-author of the Session Initiation Protocol specification, I am frequently asked about the "hot new services" SIP enables. My answer invariably begins with an explanation of how SIP can enable many new services. Right now, Fixed Mobile Convergence (FMC) is one of the hottest SIP services. FMC is interesting in that it is a highly compelling application, and it also shows off some of the key tenets of SIP's architectural philosophy.

What Is FMC?

Its definition varies somewhat depending on the context. Sometimes it refers to a unified service offering where a subscriber has a fixed-line phone and a mobile device from the same provider, both devices sharing a single phone number. More commonly it refers to a service that uses a dual-mode client that supports some kind of traditional 2G cellular voice access, such as CDMA or GSM, in addition to broadband IP, typically through Wi-Fi. The phone can seamlessly roam from the 2G cellular network onto Wi-Fi and back, handing off calls as the user travels. The benefits to subscribers include increased network coverage (especially in the home or office, where Wi-Fi is nearly ubiquitous but cellular coverage sometimes isn't), the ability to roam internationally (anywhere there is Wi-Fi) and possibly lower service costs.

There are many ways to realize FMC. One of them is with Unlicensed Mobile Access (UMA), which tries to make the Wi-Fi network look like the cellular network from a protocol perspective. The more interesting approach is to use SIP. Technically speaking, the SIP approach is fascinating because it's a great example of SIP's component architecture. The SIP component architecture says that, instead of standardizing specific services and features, you define a set of coarse-grained call control primitives, and then build up complex services by combining those primitives together. Despite the apparent complexity of the service, FMC doesn't require any extensions to the SIP at all. Though it was not envisioned or considered at all during SIP's design, the existing set of SIP call control primitives can be combined to realize FMC.

How It Works

The FMC service is typically realized through an application server, which I'll call an FMC server for simplicity's sake. When a user on the PSTN or in the SIP network calls the FMC subscriber, the call is routed to the FMC server where it is anchored. The call is routed to the FMC server using SIP's loose routing capabilities. Once there, anchoring is achieved by having the FMC server act as a Back-to-Back User Agent (B2BUA), which re-initiates the call toward the FMC subscriber. The FMC server can route the call into the cellular network or to the SIP User Agent (UA) on the dual-mode device. To decide which, it can learn about availability of the SIP UA by checking registration status, either through receipt of third-party SIP registrations, or through the Registration Event Package (RFC 3680). To achieve handoff, when the FMC subscriber moves from cellular to Wi-Fi, the UA initiates an INVITE toward a service URI. This URI routes the call to the FMC server. The FMC server, acting as a B2BUA, uses third-party call control primitives to move the media stream to the Wi-Fi endpoint and terminate the call leg pointing into the cellular network. The operation works similarly for handoff from Wi-Fi to cellular.

Many of SIP's component functionalities are used to realize FMC. These include:

Loose Routing — Used to route calls to the FMC server and back out.

Event Packages — The Registration Event Package can be used by the FMC server to learn registration status on the IP network.

Service URI — Introduced in RFC 3087, it defines the notion of a URI being used to represent a service, and constructing it in such a way that service parameters are part of the URI. The service URI here represents a handoff service, and the parameters are the context needed by the FMC server to identify the call(s) to be handed off.

Third-Party Call Control (3pcc) — Defined in RFC 3725, it introduces several primitive call flows for manipulating dialogs from a central server. These are used extensively by the FMC server.

It's important to note that, although FMC doesn't require extensions to SIP per se, it does require standards work in order to provide interoperability. Here, the standards work that is needed is the definition of the behavior of the FMC server when it receives requests with specific SIP URI, triggering either handoff or anchoring. Once defined, the FMC server becomes another component that can be used, in turn, to realize even more complex services where it is but a single component. In this way, the FMC becomes part of the "library of primitives" that makemakes up the SIP toolbox. Here, it's not a SIP extension, but rather a server with well-defined behaviors. Indeed, many standardized servers have been defined over the years. Another example of such a standardized server component is the focus, defined as the signaling center point in SIP conferences (see draft-ietf-sipping-conference-framework), and transcoding servers, defined in RFC 4117.

As FMC services roll out this year and next, I'll be one of the first to sign up. I'll get it because I think the service itself is really cool, but also because I know it'll be a great example of the philosophy that I and others have worked hard to promote as the SIP way of innovating telecommunications

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. (news-alert)

SIP: Enabling the Hidden Potential of VoIP

By Todd Simpson and Alan Hawrylyshen

GEMAYA is a term coined by David Kirkpatric in a recent *Fortune* article as an acronym for Google, eBay, MSN, AOL, Yahoo, and Amazon — the Internet heavyweights. Like IBM and the BUNCH from the heyday of large computers, the distinguished GEMAYA group is likely to pioneer a new generation of interactive services layered atop their current core offerings. The integration of voice, video, banking, gaming, and other real-time applications with instant messaging, chat rooms, and e-mail is already well underway. One example is eBay's recent acquisition of VoIP provider Skype. Ponder for a moment the potential of combining a global marketplace with robust financial services (PayPal) — both backed by voice, video, multimedia messaging, and more.

Will one-stop-shopping with such sophisticated capabilities enable GEMAYA to leapfrog traditional voice carriers with fully-bundled and low-cost (or free) services? There is no way of knowing today, but two things are fairly certain at this stage. First, driven by IP cost models and service potential, these are (finally!) exceptionally interesting times to be in the Internet telephony business. And second, regardless of the direction the market takes, the Session Initiation Protocol (SIP) is destined to play a critical role. This article analyzes both the solid core and ragged edges of SIP (and some related protocols), and investigates those areas where evolution is most intense and current implementations are most challenged. These areas include interoperability, security, quality (as a superset of QoS), middleware and 'middleboxes,' and support for rich application deployment.

The key to creating winning solutions in this rapidly-evolving space will be the ubiquitous user experience — all applications, anytime, anywhere, on any device. Such ubiquity is most likely to be derived from formal standards, as opposed to a single de-facto standard largely because no single player has enough critical mass to force-feed a proprietary solution that satisfies the absolute need for interoperability. As in many leading-edge areas, the standards are evolving and competing. Still, the Session Initiation Protocol (SIP) is the new standard for real-time services. The reasons are straightforward: SIP is a flexible, extensible, rich, and highly-leveraged specification. SIP puts few bounds on potential applications, allows for extensions and enhancements, and reuses some of the best Internet technologies to date (for example, encryption and authentication mechanisms, and MIME data types). Of course, with flexibility and richness comes the potential for complexity and confusion. Today SIP is in its adolescence; much has been learned since its infancy, and much more will be learned as it matures.

Interoperabilty

Interoperability, or even operability, continues to be an issue in today's growing SIP-based infrastructures. While there are still significant discussions around interpretations of the SIP specification, many of these issues occur in areas the specification simply does not address. Still fundamental in this area is the issue of NAT and firewall traversal: SIP uses IP addresses in order to set up sessions (How else would it be done?), and these addresses are invalidated by Network/Port Address Translation, and firewall behavior. Many non-standard approaches are in use to solve this problem, but none can claim to be 100 percent effective owing to the overwhelming variety of NAT and firewall devices (for example, port-rotating firewalls can still be problematic). Emerging standards and processes such as STUN, TURN, and ICE should soon allow this particular area to become more cohesive.

Other interoperability issues can occur based on the implementation. For example, SIP does not specify an upper bound on header sizes, but many implementations have hard-coded bounds; messages beyond these bounds are rejected. Improper implementations of registrars and proxies, either through bugs, incomplete designs, or misinterpretation of the specification can also lead to interoperability problems. Lack of support for full DNS resolution (as documented in RFC 3263 including NAPTR and SRV support), basic security mechanisms, and flawed CODEC negotiations, might also lead to interoperability issues. Fortunately, the industry recognizes the need for solid interoperability, and efforts such as the SIP Forum's SIPit events have enabled much progress. And yet, the industry is likely to end up with pseudo-interoperability, much like with HTTP, where content works better in some browsers than in others. Of course, unlike the World Wide Web where only a handful of browsers are used, the world of VoIP currently involves myriad different systems and devices from a plethora of vendors.

Security

Now that several large VoIP access networks are operational on the Internet, the issue of security is becoming increasingly important. And like all security concerns, there is a mix of real issues and fear-mongering. The SIP specification leverages the best of mature Internet security models, which when fully implemented, distill the true areas of concern to border cases. Unfortunately, the state of deployments today tends not to include even basic security provisions, leaving networks open to many sorts of breaches.

SIP contains excellent support for ensuring point-to-point confidentiality and encryption, including the adoption of digest authentication, S/MIME, and TLS encryption, and the ability to share keys for media encryption. Many of these mechanisms, however, still have subtle technical or management drawbacks; for example, sharing keys for digest, or managing the tradeoffs between end-to-end and point-to-point architectures. For these reasons, implementing full cryptographically secure end-to-end authentication remains a challenge, especially given the realities of disparate domains of trust and the existence of the middle-ware boxes needed to overcome other interoperability issues. Finding and deploying a full solution that addresses this problem will also be essential to combating SPAM over Internet Telephone, or SPIT, as well as other nefarious attacks.

Another example where security could be compromised is with forked requests — where a single invitation is sent to multiple contact points (a home and cell phone, for example). Because the means to authenticate and authorize the response from each fork is not well specified (a proxy many only return the response from one endpoint, somewhat arbitrarily chosen) many different behaviors are possible. This problem area is known as the Heterogeneous Error Response Forking Problem, or HERFP, and remains under discussion at the IETF.

An additional example is the interplay of SIP with other Internet protocols. Routing of SIP requests is often handled via DNS using the NAPTR and SRV records. SIP itself does not specify how to validate or authorize DNS results, so tampering with SRV records can be used to misroute messages. The interplay between SIP and other protocols is a fruitful area of research and implementation innovation. Of course, security breaches that compromise underlying (and unprotected) protocols and resources are not unique to VoIP in general and SIP in particular. This widespread problem is why work is ongoing by the DNSSEC group and others at the IETF to enhance DNS integrity.

Quality

Overall user experience, or "quality," is also still hit and miss in today's networks. Again, SIP itself is not the culprit, or perhaps even the solution, but the interplay between SIP and other functions needs to evolve to address this issue. Quality includes always having connectivity, the speed of connection (ringing, for example), and the quality and consistency of media delivery. Within one homogeneous environment, adequate quality may be delivered by making logical network and bandwidth decisions, and enforcing these across the network. Across heterogeneous networks, however, the problem is significantly more challenging. Where one network may employ MPLS or VLANs to guarantee QoS, another may simply use DiffServ and where an IP network meets the PSTN, numerous interface issues (such as echo and security) can also occur. Even between peering partners carrying purely IP backbone traffic, there may be different design choices, especially involving encoding. For this reason, a network that is optimally designed for larger packets at slower intervals may not work well with a network optimized for smaller, more frequent packets. Jitter and latency correction at the endpoints may not be sufficient to compensate for allowing arbitrary media routing.

Deploying services beyond VoIP across SIP-based networks has its own set of challenges. Environments like IMS and TISPAN (the 3GPP's IP Multimedia Subsystem and ETSI's Telecoms & Internet converged Services & Protocols for Advanced Networks) anticipate video, TV, gaming, access to back-end databases, and many other session-related applications. Administering, controlling, authorizing, and guaranteeing quality of service across these multimedia networks is non-trivial. For example, while a few seconds of delay in ringing another user agent may be acceptable for telephony applications, it may be completely inadequate for a massive online gaming environment. Likewise, while non-repudiation may be manageable within voice-only applications, it must be rock solid for banking applications. Thus, the scope and complexity of the issues mentioned above become more acute in a rich application environment. And overcoming interoperability issues between SIP and existing security mechanisms, like X509, S/SMIME and the related PKI technologies, will be essential to widespread adoption of these services.

FEATURE ARTICLES

At the other end of the spectrum from IMS, which has a heavy back-end infrastructure, is the work on P2P SIP, which attempts to remove any dependence on a back end. The ability to quickly set up ad-hoc, real-time sessions using a P2P system (along the lines of Skype) has obvious advantages to the end user — and disadvantages to service and equipment providers. There is nothing fundamentally difficult in having P2P endpoints communicate with an IMS infrastructure, however, other than the previously highlighted management and control of such services. Pure P2P SIP applications will occur; planning for their integration and control is only just beginning to be discussed.

Conclusion

As the clash of the titans heats up — GEMAYA versus The Incumbents — SIP is positioned to play a central role. While SIP has certainly proved its flexibility and worth to date, it still has much room for improvement and growth. This makes the SIP battlefield an exceptionally exciting and (potentially) prosperous place to be. SIP is solidly positioned to be the underlying standard for real-time services on the Internet. And given the relentless growth in connectivity and available bandwidth, what application can afford to not be real-time!

Todd Simpson is vice president and general manager and Alan Hawrylyshen is director of VoIP protocols at Ditech Communications. (news - alert) For more information, please visit the company online at http://www.ditechcom.com.

SIP... So, What's All the Hype About?

By James Gledhill

SIP is arguably the most important Voice over IP (VoIP) protocol on the market today. Unlike its supposed competitors — namely, H.323 and MGCP (Media Gateway Control Protocol) and its derivatives, which were specifically built to reproduce PSTN like voice conversations on an IP media — SIP was not built for voice. Rather, it was built for any type of "Session." SIP was applied to voice after the fact.

This openness of SIP, by not being built specifically for voice, gives it an ability to be adapted for many things, including, but not limited to, voice conversations. SIP is used for IVR (Interactive Voice Response) and ASR (Automatic Speech Recognition) products, voice and video conferencing, push-to-talk, instant messaging (see the SIMPLE standard), and a host of other product types. Additionally, SIP's addressing scheme, which looks much like an e-mail address can provide a more user-based identifier, which is far more rich than the impersonal numeric addressing scheme of telephone numbers.

While H.323 is still widely used in some parts of the world, it is a forgone conclusion that it is a dying legacy VoIP protocol. There are a few non-standards based VoIP protocols being used and promoted — specifically by Skype (news - alert) and Cisco (quote- news - alert). But as for the standards-based VoIP protocols, SIP and MGCP are currently the predominant competitors. However, SIP and MGCP are as different from each other as POTS (Plain Old Telephone System) phones are from cell phones. While both are capable of making phone calls, they are very different in how they work "under-the-covers" — in architecture, capabilities of devices, interactions with servers, where intelligence resides, and so forth. Indeed, SIP and MGCP are very different paradigms.

MGCP simulates and tries to match the legacy PSTN (define - news - alert) network. PSTN phones connect to the phone network by constantly being connected to a single Class 5 Switch in a master-slave architecture. The Class 5 Switch controls every aspect of the phone and there is no interaction with the phone except via that switch. An MGCP device could be compared to the interaction of a dumb terminal connected with a mainframe computer. The device has very little intelligence of its own and relies on all communication going in and out of the mainframe, which controls all activities.

MGCP likes to make the case that because the phone or ATA is "dumb," it requires less internal electronics and, therefore, should be cheaper than SIP phones. This can be somewhat true, but the capabilities of the devices are also diminished and all the processing is forced on the switch, making the switch far less scalable than in the distributed processing architecture of SIP. SIP could be compared to a PC connected to the Internet. The PC does not rely on an external mainframe computer to control its every action. With the PC, the processing is distributed among many computers, including the PC. The PC may act as a client connected to other servers, but can also act as a server for other PC interacting with it. The PC can connect with many different servers — not just a on a one-to-one relationship.

True SIP devices work the same way. They can connect with many servers simultaneously and even act as servers themselves, allowing other devices to connect to them. Although the capabilities of a SIP device (like a phone) may only allow a single "session" to exist at a given time, each SIP device is both a client and a server. Every SIP device contains a SIP "user agent," which is made up of a user agent client and a user agent server. Two SIP devices can connect to each other without requiring any other component to be involved. (Note: Many service providers lock down the SIP device so it can only communicate with their server. While this may be a

requirement imposed by the service provider, it is not a requirement imposed by the SIP standard.)

In contrast to PSTN and MGCP phones, cellular phones have much more intelligence built into them and their interaction with the "network" is not a master-slave relationship. Instead, it is a one-to-many distributed relationship, where a phone may interact with one switch now, but another switch a few seconds later. Cell phones and SIP phones have the concept of registering and re-registering a device location, which can change frequently or remain the same for extended periods. Cell phones and SIP devices work on a distributed architecture where the intelligence is distributed among a number of pieces in the network, including different servers/switches and the phones. The phone device plays a much larger role in the cellular and SIP networks than it does in PSTN and MGCP networks.

The cellular industry liked SIP so much that they have adopted it as the protocol of the future cellular networks as well as for the new cellular applications framework. So, while MGCP strives to re-model the legacy PSTN network, SIP is setting the standard for new emerging products to come. SIP is a protocol that will allow us to do things we have, until now, not been capable of doing.

Those companies that truly understand the SIP paradigm do not just think about making phone calls. They think about the possibilities above and beyond how things where done in the past. They make vastly different devices than those built for MGCP. There are a number of devices available to buy that were originally developed as an MGCP device and then later converted to use the SIP protocol. Many of these device manufacturers never made the paradigm shift from MGCP to SIP, so their devices — while using SIP as the protocol — still require all the intelligence to be on a single server. They do not move any of the intelligence to the SIP device.

At the end of the day, SIP provides the best path for future IP-enabled applications. By embracing and supporting SIP now, service providers will have a technical and operational advantage over providers that are still dependant on legacy protocols.

This article was prepared by James Gledhill, Strategic Partnering, SipStorm (news - alert). For more information please visit the company online at http://www.sipstorm.com.

What Do You See When You Look into the IMS Mirror?

By Kenneth Chen

Service providers and equipment vendors, both wireless and wireline, have recently expressed great interest in IMS (IP Multimedia Subsystem). IMS is a system consisting of standard protocols plus traditional Telecom's needs for business model and operation model. Although it has its roots in wireless, IMS is a system based on Session Initiation Protocol (SIP). Therefore, it has appeal to almost everyone in the Telecom ecosystem.

However, like the Akira Kurosawa classic film Rashomon, what telecom service providers and vendors see in IMS (define - news - alert) is often very different. Each looks at IMS through the lens of what they need to compete within an existing business model. However, this view often omits what the telephony users are demanding. They have jumped on open standards, such as SIP, to provide them with a larger degree of choices at competitive prices.

Wireless Service Providers

The average revenue per user (ARPU) value of wireless communication is consistently decreasing. Subscribers are making more calls, and using more minutes, but their phone bills are getting smaller, instead of larger. Mobile data is an important method to increase ARPU. Mobile communication service providers are actively seeking ways to create new revenue sources via expanding the range of their mobile data offerings. However, the service cost calculation method of traditional IP (or Internet) services cannot satisfy the profit needs of mobile communication service providers (as IP/Internet services are free to Internet service providers).

Mobile service provides a mobile data platform that is beneficial to mobile communication service providers through user identification, signaling control, and usage-based billing technologies. This enables a wide variety of data services that conform to this standard to be launched on the mobile network, allowing these service providers to profit from an IMS platform.

Traditional Wireline Service Providers

Due to the same characteristics, IMS is appearing even more attractive to wireline service providers than to mobile operators. Wireline service providers have also taken tremendous interest in IMS because it can enable interconnection between wireless and wireline networks (for example fixed mobile convergence or FMC). Traditional telecom service providers have also found a need for IMS as the solution for their need for Triple Play.

As traditional telecoms generally adopt one system for their entire range of services, IMS with its all-in-one capabilities make it an easy and obvious choice for them. Another advantage IMS presents to traditional telecoms is the capability to support end-user control and value-added service access control, which is essential to ensuring the continual success of existing business and operation models. However, the prospect of IMS is not without its perils. As a newly emerged standard, the range of offerings based on IMS is still fairly limited.

IP-based Service Providers

Service providers that use IP-based networks can rely on pure SIP to provide the services that are being promoted for IMS. SIP — as an open standard protocol — inherently supports network interconnectivity, which is the foundation upon which new telecoms based on IP are founded. The problem with SIP is that it does not offer an all-on-one solution package like IMS. Fortunately, new IP telecoms are used to accepting and adopting new standards (for example SIP, http, radius, etc.), and rarely employ a ready-made operation system. Instead, they create customized business models and operation models based on company and customer needs.

However, this benefit also presents risks to the pure SIP operators. Pure SIP operators still need to create a set of business models that can reach beyond the traditional ideas that IP/SIP-based telephony offerings are only valuable as free or low-cost alternatives to the PSTN. Pure SIP operators need to create revenue models and profit models for themselves and their partners that provide innovative values and are recurrently profitable.

> From the end users' perspective, IMS and pure SIP also offer very different values.

SIP Versus IMS

From the traditional wireless and wireline telecom service providers' point of view, IMS might present a more attractive view because a simple and open standard like SIP does not provide enough control over their end users and value-added services to suit their business model. However, the story is quite different from perspective of IP-based service providers and content providers.

Every telecom operator is planning to use its own version of IMS. In order to distribute their services and content over a telecom service provider's network, independent IP-based service providers and content providers must work with operator-specific business and technical requirements. This limits the ability of independent IP-based service providers or content providers to easily partner with multiple telecom operators.

One of SIP's main benefits is that it offers easy convergence to value-added service and application providers that develop their products based on SIP. The partnership models between pure SIP telecom operators and independent service providers and content providers can be much more flexible, as the area is not yet solidified, thus allowing more room for creative solutions. New independent value-added service providers and content providers are also arising out of the abundant opportunities, just as with the development of IP in the past ten years.

From the end users' perspective, IMS and pure SIP also offer very different values. End users in the IMS architecture can only access services provided by the telecom operators or their value-added service provider and content provider partners. End users in a pure SIP environment, on the other hand, might have more flexibility in selecting from product offerings.

When looking at the issue of IMS versus pure SIP, one also needs to look into the fundamental differences between traditional telecommunications network and the Internet. Traditional telecom networks are limited by geography coverage and network concentration locale. Crossing over boundaries require higher cost (for example, roaming charges). As IMS is being developed to respect these boundaries, traditional wireless and wireline telecom operators are welcoming it.

A SIP-based network maintains the open and public nature of the Internet. IP telecom operators have adopted the open SIP standard and their operation model and business model derived from this open standard. If a public, open SIP network indeed becomes successful, it could provide a new open telecom environment and many business opportunities.

An open and simple standard, such as SIP, may not be able to satisfy all aspects of the operational model and business model needs of traditional telecoms. But, as an open standard unshaped by operation and business models, SIP could be the foundation for an IP-based business model that responds to the needs of users, who are embracing open standards.

Conclusion

It is impossible to predict whether one will overcome another or both will merge into a newer and better solution. No operator is presently using IMS to launch products into the market. High-growth SIP operators have not yet amassed enough customers. Notwithstanding all of the recent hype about IMS, don't underestimate SIP. It may not get as many press clippings, but an open standard often brings the power of grass roots support — which is one of the hallmarks of success with the Internet. The shape of the ultimate Triple-Play solution just might be staring at us if we look in the SIP mirror.

Kenneth Chen is chief strategy officer at TelTel, (news - alert) a provider of SIP-based global Internet telephony services. For more information, please visit the company online at http://www.teltel.com.

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