



SIP PACKS A PUNCH

Having this voice protocol in your corner brings easier integration, more choices, lower costs and the innovation muscle that only a ubiquitous standard can provide **BY PETER MORRISSEY**

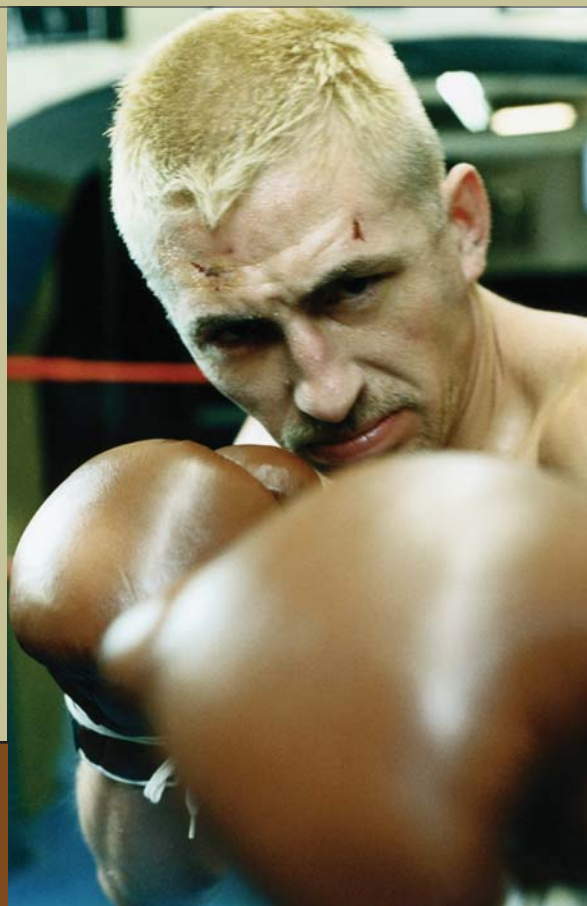


SIP has emerged as the undisputed protocol champ in its class, knocking H.323 for a loop. Session Initiation Protocol has not only proved it can deliver standards-based VoIP (voice over IP) phone systems, it also has the legs to impact many other technologies involving real-time communications, including instant messaging, presence, 3G wireless and packet over cable. In addition, SIP is making it a lot easier to add related services to your network, such as voicemail, IVR (interactive voice response) and unified messaging.

Many existing products boast SIP functionality, and new offerings are following suit. All have the potential to make business communications more productive and less expensive. If you're considering a purchase in these areas, ask prospective vendors about their road maps for incorporating SIP functionality. Get specifics on what features the vendor plans to add, and when, and demand interoperability assurances.

The Basics

SIP, an Internet protocol most recently incarnated in the IETF's RFC 3261, is designed to locate, set up and control connections among endpoints or applications on an IP network. SIP is mod-



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ular in that it can use any media transport, and it makes use of the SDP (Session Description Protocol) to determine the media capabilities of endpoints attempting to connect. For example, a VoIP phone could call a video phone, and the two devices could communicate based on the audio capabilities common to both.

Once a connection is negotiated, SIP gets out of the way, which makes it scalable. Servers can be used to locate and connect all the endpoints involved in a session, including those that support IM. SIP-enabled voicemail and IVR systems can be added to the VoIP PBX function easily, as long as there's IP access.

SIP is also ideally suited for presence applications and can integrate with Web applications, easing CTI (computer-telephony integration) and opening up more options for features like "click-to-talk."

Development and support are simplified by having one protocol and architecture. Any application that requires instantaneous communication is probably already supported by SIP or will be. The beauty of this is that one simple infrastructure, consisting of a SIP Proxy Server and a SIP Registrar Server, can direct requests and track location information. Additional applications can be added with little disruption.

Sure, it's possible to accomplish all this using proprietary systems. However, that route is not necessarily easy and may lock you into a single vendor. Not surpris-

ingly, some vendors with big investments in proprietary alternatives argue that SIP is still a work in progress. But after testing VoIP phones, we say SIP brings to the table solid functionality and interoperability. (For the nuts and bolts of how SIP works, see "It's Time To Take a Look at SIP," at www.nwc.com/1407/1407ws12.html).

All Aboard

Lots of companies are cooking up products with SIP as a main ingredient. These include such big names as Alcatel, IBM, Microsoft, Novell, Polycom, Siemens, Sun and 3Com, and smaller players like BroadSoft, ipDialog, Mitel Networks, Snom Technology and Zultys Technologies. SIP-enabled VoIP is in the forefront, but messaging and presence, while not quite as far along as VoIP, appear on a lot of vendors' near-term road maps.

For example, IBM has made a strategic decision to incorporate SIP functionality into all future versions of its corporate messaging product, IBM Lotus Instant Messaging and Web Conferencing (formerly known as Sametime and an Editor's Choice award winner; see "IM Grows Up," at www.nwc.com/1412/1412f3.html). The product's messaging gateway now supports the SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) protocol. This was an obvious choice for a gateway that can link messaging systems from different vendors until they all start incorporating SIP functionality into their client and server products. SIMPLE could do for messaging what SMTP does for disparate e-mail systems.

IBM further plans to incorporate SIP functionality—including presence-based features that make it easy to send availability information with documents—in all its client and server software. Presence, which IBM implements in a nonstandard environment, makes it possible to communicate the availability of an individual via office phone, mobile phone or IM and the preferred method of contact at any given time. An obvious place to implement and manage presence is with groupware, where it can be integrated with schedules. This is why IBM plans to incorporate it into Lotus Notes.

For its part, Microsoft has for a number of years incorporated SIP support in its MSN messenger clients, including its most recent version, MSN Messenger 6.0. Microsoft is also using SIP for its XP Messenger client, which comes installed on every XP desktop. Clearly, like IBM, Microsoft has made a strategic decision to base future voice and video communications on SIP, turning its back on the H.323 standard that powered the NetMeeting product and provided similar functionality.

Indeed, while H.323 has been around longer, you will be hard-pressed to find new development for H.323 products. Most vendors interested in standards-based communications have shifted their efforts toward SIP.

Why? SIP is more extensible, simpler to implement and easier to troubleshoot thanks to its simplicity and text-based messages. The SIP protocol requires fewer packets to be exchanged to set up a call, which means less network traffic, less work for the servers involved

Executive Summary

VoIP Using SIP

Imagine having a dynamically updated icon on your desktop that shows the best way to get in touch with a co-worker. You could see if she were in a meeting, for instance, but had her wireless laptop with her and was available via IM for urgent questions. Or, if she were on the phone in her office, the icon could notify you when she hung up. There are, of course, proprietary ways to provide such functionality. But for those of you who, like us, are partial to standards, say hello to SIP.

Session Initiation Protocol has been around for a few years, but lately it's really picking up steam. Many smaller players are vying for market attention, while big-name vendors are jumping on the bandwagon. Microsoft, for example, seems excited about providing presence services via SIP, and IBM plans to revamp its client and server software to utilize the protocol.

For those who are ready to get down with SIP now, we brought standards-compliant VoIP phones from ipDialog, Mitel, Polycom, Siemens, Snom and Zultys into our Syracuse University Real-World Labs®. How did they fare? Frankly, we were knocked out: Each and every one earned a perfect interoperability grade. Chalk one up for standards compliance.

and thus greater scalability. SIP is also more flexible than H.323. There's a whole industry developing based upon SIP products. Take a look at www.sipcenter.com for a sampling of some of the equipment and application vendors. A standard like this provides a much larger market for developers than proprietary products can provide, which in turn increases competition and innovation.

Many feel that H.323 is still superior for managing video services. But this hasn't stopped companies like Wave Three, VCON Visual Communications and others from providing SIP-based video products. And some IP vendors with legitimate interests in standards-based connectivity adapted H.323 for their phones before SIP was a viable standard.

There's further evidence of SIP's knockout power on the server side: Microsoft's new RTC (Real Time Communications) server will use SIP for all of its media-related communications. The first application that Microsoft will introduce on this platform will be a corporate messaging product. The company also plans to provide RTC as a development platform. Siemens, for example, will incorporate RTC in its future products to enable it to standardize its messaging and presence applications.

The bottom line, though, is that the industry has declared SIP the winner.

Yes, But ...

We know what you're thinking: Microsoft appears to have embraced SIP with enthusiasm, but given its track record with standards, how long will the true spirit of interoperability survive? Along with our review of SIP

phones (see "Polycom KOs Proprietary VoIP Woes," page 34), we did some testing with MSN Messenger and XP Messenger, and found that there is indeed a lot of interoperability at the client level. For example, we used BroadSoft's Proxy and Registration servers to set

SIP provides a much larger market for developers than proprietary products can provide.

up voice and video sessions between Microsoft clients (see "The BroadSoft Connection," page 32). Taking it a step further, we established voice conversations between all of our SIP phones and Windows clients.

Microsoft's client implementation of SIP is not perfect, however: Whenever we called XP Messenger with a SIP phone, the local XP Messenger client responded by displaying a message that the calling device wanted to establish a "voice and video conversation," even though the phone had no video capability. This happened with every SIP phone, even when attempting to establish a voice-only call from version 4.6 of MSN Messenger, which has no video capabilities! This was disappointing because one of the strengths of SIP is that it can use SDP to negotiate media capabilities between devices. The Microsoft client obviously wasn't designed to take advantage of this design. Microsoft had no response to our queries about this problem.

It's also worth noting that we found it impossible to

SIP Shows Some Hospitality

The Hotel Commonwealth, a spanking new luxury hotel located on Boston's historic Commonwealth Avenue and affiliated with Boston University, is sold on SIP.

"We started with a totally blank slate," says Stewart Randall, principal consultant with Communications Design Associates, "but we knew we wanted the telephone infrastructure to be leading edge." Randall kept his options open: "We went with a structured wiring solution where we had Cat 5e running to all rooms and fiber between floors."

BU acts as the hotel's ISP and provides a full Class C link. "We ran 54 strands of dark fiber from the hotel to the university's core," Randall says.

For its in-house VoIP phone

service, the hotel issued an RFP and weighed Avaya, Cisco, NEC and Nortel solutions before choosing Alcatel's Omni PCX, which gained native SIP functionality earlier this year. It was Alcatel's commitment to SIP and open standards that gave it the edge over Cisco.

"Our general recommendation to anyone is always to go with open standards," Randall says.

All 150 guest rooms and the lobby areas will contain decorator-approved cordless Pingtel Expressa SIP phones that offer guests many advanced features, including speed dialing and calendars, as well as the all-important wake-up call, which can be scheduled using the phone's LCD-screen menu. The Pingtel phones are connected to a Linux

server and run Personal Java, allowing for easy third-party development of applications.

For a luxury hotel, being on the cutting edge carries some risk, but Timothy Kirwan, the hotel's managing director, says he believes the rich functionality provided by the IP phone system will pay off in guest satisfaction. "We're one of only two or three hotels in the country that have taken the plunge," Kirwan says. "It's a groundbreaking thing for our business."

Check out the Hotel Commonwealth at www.hotelcommonwealth.com, and listen to our interview with Stewart Randall and Timothy Kirwan at www.nwc.com/1416/1416p1.html.

transfer a call or do a teleconference with a Microsoft Messenger client. Still, it worked fine for simple phone conversations, though it suffered from more latency and echo than the hardware phones we tested.

Call Me

One area where SIP is making strides is IP telephony. Carriers providing IP Centrex services have long been advocates, and the eponymous BroadSoft SIP platform we used for our testing helps carriers and ISPs offer the service. This cost-effective, reliable turnkey product, assembled from best-of-breed SIP vendors, integrates with its own customizable software to make it easy for a carrier or ISP to roll out IP Centrex and advanced services.

For example, we set up unified-messaging services that automatically sent voicemails to an e-mail address. Web-based provisioning, as well as a Java application called CallPilot, made it a cinch to forward calls to another phone or redirect them to a remote location. We could even conference in other phones when using Microsoft's Messenger client, and we easily set multiple numbers that would ring simultaneously. CallPilot even provided a detailed log of all dialed, missed and received calls.

Need another example of the SIP protocol's flexibility? BroadSoft plans to add IVR and speech-to-text services to its offering by partnering with Holly Australia.

Clearly, SIP is making inroads with carriers and even consumers (see "Start Small" at www.nwc.com/1416/1416rd2.html), and though the enterprise voice side is a little slower on the uptake, SIP is gaining momentum there as well. Unlike carriers, many enterprise users have been willing to tolerate VoIP products that have been just as proprietary as the legacy systems they are meant to replace. In fact, some companies are locking into the same vendors that provide their voice and data networks, winding up with deeply proprietary setups.

That's starting to change, though. In our tests of SIP-enabled VoIP phones, we found them not only interoperable, but very functional. In fact, we made a point of using SIP phones exclusively for the duration of testing, and all calls went off without a hitch. Qual-

ity was also a pleasant surprise: All these calls took place across the Internet, from our Syracuse University Real-World Labs®, using BroadSoft's proxy and PSTN gateway services at its locations.

Still, despite the fact that SIP connects so well, you will be hard-pressed to find end-to-end SIP connectivity between organizations. The PSTN is still used to talk to the outside world. That means you'll need a gateway translating from SIP to the legacy PSTN technology. The advantage SIP gives you here is that you can choose from lots of gateways based on price and features, instead of being locked into the one provided by your vendor's IP PBX. The gateway just has to be reachable from the inside via IP.

We tested another example of SIP in the enterprise, a SIP-based PBX from Zultys (which also participated in our phones review). The MX 1200 supports as many as 1,200 SIP phones and has all the functionality of a conventional PBX and more, including voicemail, automated attendants and ACD. It also includes a switch, a router and support for standardized Power over Ethernet (IEEE 802.3af). In addition, the MX 1200's built-in DHCP and TFTP servers have GUIs that made deploying SIP phones a breeze.

Market-leading PBX vendor Alcatel also added native SIP functionality to its OmniPCX Enterprise product earlier this year, making it possible for enterprises to choose phones from multiple vendors for use with their PBXs. Alcatel has demonstrated that this vision is for real via a multivendor installation at The Hotel Commonwealth, a Boston luxury hotel: The office phones are from Alcatel, while guest phones are supplied by Pingtel (see "SIP Shows Some Hospitality," page 31, for more on the Hotel Commonwealth).

Security Watch



Any application accessible on an IP network is a potential security risk. SIP phones and servers are no exception: A number of vulnerabilities, affecting a long list of vendors, have been discovered; see www.cert.org/advisories/CA-2003-06.html.

At minimum, limit access to any SIP device. Also, purchase products from vendors that specialize in securing SIP. For example, SecureLogix Corp., which won our Well-Connected Product of the Year Award, is working on a device designed to secure SIP environments, and BroadSoft uses a product from Kagoor Networks to do the same.

The BroadSoft Connection

For our cover package we partnered with BroadSoft, a leading provider of hosted communications platforms based in Gaithersburg, Md.

BroadSoft has adopted the SIP standard in the architecture of its communications products (see www.broadsoft.com). We took advantage of the proxy, registration and PSTN gateway services provided by the company's BroadWorks platform to implement our testing of SIP phones, and we made use of its Web-based provisioning software as well.

Founded in 1998, BroadSoft delivers a carrier-class platform that has enabled companies like Telstra, the main carrier in Australia, to provide advanced IP Centrex applications. BroadSoft also provides the infrastructure for hosted PBX services at Computer Sciences Corp. and Science Applications International Corp.