VOICE-DATA CONVERGENCE

SIP: Gaining Momentum

Edwin E. Mier and Martin Milner

How close is SIP to world domination? It's getting there, but SIP implementers around the world have varied opinions.

SIP support is swelling. Vendors—as well as enterprise users—are increasingly serious about putting SIP to work. And long-held SIP expectations—interoperability, cost savings, new advanced features—are coming to pass, albeit slowly

That's the bottom line in this year's "State of SIP" assessment, based on ongoing research, Miercom's hands-on reviews of the latest SIP-based wares and features, and a detailed survey of the vendor technical community driving SIP standards and their implementation—work conducted by Miercom in association with *BCR*.

In early March, Miercom contacted, and then emailed a detailed survey to, vendors of products based on or involving SIP, of which we were aware—72 of them. Given the rapid emergence of new companies with SIP-based products, though, we estimate the SIP vendor community has expanded to 80, perhaps even more, since then.

By the deadline for replies, we had received about 40 completed, validated surveys. The responses tallied here, then, represent roughly half of today's SIP-supporting vendor industry. In product areas including IP-PBXs, most leading vendors took part. In other areas, some vendors are notably missing. SIP gateway leaders AudioCodes and Mediatrix, for example, did not respond. (Their wares are, however, rebranded by some of the respondents.)

Also, we estimate that there are some two dozen players in the SIP-phone marketplace—offering assorted SIP hard phones, softphones and wireless/802.11 endpoints. And some notable ones—like Snom (www.snom.com), Grandstream (www.grandstream.com) and ACT (www.act-tel.com.tw)—did not respond.

The survey had two parts: In one, probing questions were asked about the vendor's view of SIP—standards, problems, interoperability and SIP's future. The other part asked for comparative details about the vendor's SIP-based product(s)—including specifics about claimed interoperability with third–party products. Respondents were offered the option to remain anonymous. And a handful—less than 10 percent of the total—did request that they not be identified as the source of their responses.

Before analyzing the results, some background explanation is needed on the thorny issue of "SIP features."

Feature Presentation

Few aspects of IP-telephony are more contentious, or the crux of marketeering boasts and one-upmanship, than "features." We queried vendors throughout this project about "SIP features"—how many features were supported, how they were implemented, which are generic vs. proprietary, and so on. We did not, however, clearly define what we meant by "feature."

Even so, many vendors provided specific SIP feature counts for their products. It's clear that some of these reflected sincere technical estimates, and that some others have been over-inflated by marketing spec-manship.

What one vendor considers a single feature another vendor may count as two, three, even five discrete features. We have seen such multiplication applied even to long-accepted telephony functions, like "call transfer." Unfortunately, there is no single industry index, with consistent names and definitions, that can be used as a SIP feature guide. Such a reference, especially one updated with IP telephony features—like "hoteling," also known as "free seating" and by various other names—would bring much-needed consistency to the industry, and especially help enterprise customers comparing products.

For now, though, we offer readers some of our observations regarding SIP features.

SIP features need to be thought of in terms of endpoint-based, PBX (or call controller)-based and trunk based. Even this breakdown is not infallible, since a feature can involve both PBX and endpoints. Consider an inbound call that gets bounced to voice mail. The re-routing after, say, three rings is managed by the SIP-supporting PBX. But the SIP endpoint needs to support a message-waiting indicator, or MWI. (Such call routing, even MWI, is now defined in reasonably solid SIP specs, by the way.)

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By “endpoint-based,” we mean that something has to be built into or supported by the SIP endpoint in order for the feature to work—whether the end user sees or gets involved with it or not. An endpoint-based SIP feature, then, is necessary for the endpoint to interact with central call control, or with another endpoint.

Clearly many SIP features—such as Transport Layer Security (TLS)-based encryption of SIP call control—involve both SIP call controller and SIP endpoints. The feature will not work unless both ends support it, and implement it in a consistent and interoperable manner. In this case, a SIP feature called “TLS-based SIP call-control security” entails both PBX-based and endpoint-based components.

As it turns out, one aspect of security—agreement on how to distribute encryption keys between and among call controllers and endpoints—is cited as a main reason that ubiquitous SIP security is slow in coming.

Cisco offers a reasonable breakdown when it comes to feature distribution. It says that SIP features are about 50 percent endpoint-based, 40 percent PBX (or other call controller)-based, and about 10 percent trunkside-based.

That said, we need to understand what is meant by a “SIP feature,” and why some vendors say the 20 or 30 SIP features they support is sufficient, while other vendors claim to support 400, 500 or more SIP features.

There are really five categories of SIP features. There are: “solid features,” like call hold and transfer, etc.; “still-solidifying features,” of more recently documented, complex functions, like multiline appearance; “feature-code access,” as might be used, say, to access a bridge server and set up a six-party, scheduled conference; “proprietary SIP extensions,” for capabilities not yet spelled out in SIP specs, usually where a SIP equivalent is needed to a proprietary-protocol feature; and “private property” functions—SIP-related but not directly SIP-based, like the software and processes employed by a particular vendor to download firmware to, or bulk-configure, that vendor’s SIP phones.

■ **Solid SIP Features:** As of today there are 25 to 30 such features. These include the dozen or so defined in the core SIP spec—RFC 3261—plus another dozen or so from SIPPING RFCs (SIPPING is one of the key IETF SIP-standards working groups). For these features, the likelihood of interoperability between independent SIP products is very high.

■ **Up and Coming SIP Features:** Then there are another 30 to 50 features detailed in various SIP “Internet Drafts”—which are on their way to becoming RFCs. Such Internet Draft-based SIP features are considered by many in the industry to be needed, clear and more or less unambiguous to implement. For these features, the likelihood of interoperability between independent SIP products is hit and miss.

Features range from “solid” SIP compliance to “private property”

FIGURE 1 The State Of SIP Specs

Rate the state of current SIP specifications (fro m all sources—IETF, SIP/SIPPING, SIPForum, etc.) for implementing these feature and functions”

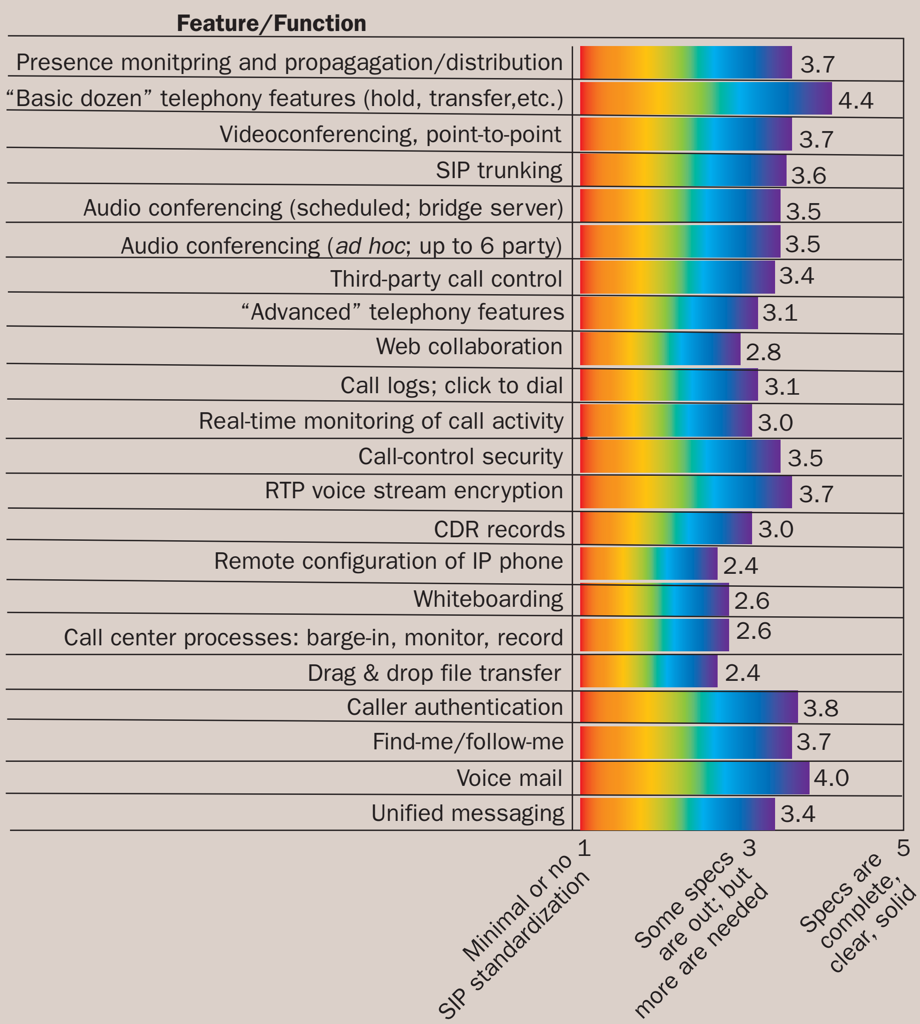
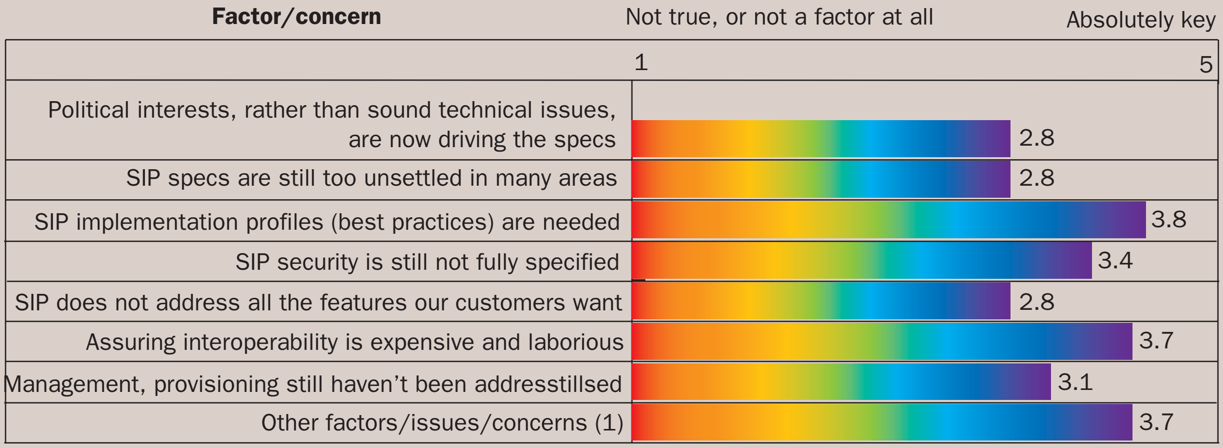


FIGURE 2 Problems, Challenges Facing SIP

"How important, relatively speaking, are these factors in impeding SIP proliferation?"



(1) "Other," most frequently cited: Complex and unsettled NAT solutions; lack of a technical standard and single method for firewall traversal; and IT staff inexperience in implementing, provisioning and maintaining SIP.

■ **Feature Code Access:** Some IP-PBX vendors have made a number of their PBX-based features SIP-accessible via "feature codes." These are strings of typically 3–5 characters, which a SIP endpoint conveys to the PBX or call controller—within a legitimate SIP message—to invoke a particular feature. One vendor popularizing this approach is Avaya, which offers access to 50-plus selected Communication Manager functions via such SIP-conveyed feature codes. Any SIP endpoint can invoke any of these features by sending the right feature code in the proper message. For these features, the likelihood of interoperability between independent SIP products is very high.

Of course, what a code "456*" invokes in an Avaya SIP environment is up to Avaya, and no other vendor's PBX is likely to understand it or react in the same way, if at all. Still, any third-party SIP endpoint that issues this sequence should be able to access and invoke this "SIP feature." A feature code would typically be programmed into a soft key on the SIP phone—assuming the SIP phone supports soft keys, and cheaper ones do not. Feature-code listings are often openly published by their respective switch makers. And Avaya, as well as others, offers interoperability testing to third parties to assure the vendor's SIP feature-codes are properly implemented.

■ **Proprietary SIP Extensions:** These, too, are "SIP features," which may be based loosely on draft SIP specifications, or not. In either case, the vendor has implemented the feature using its own, proprietary message headers. These are allowed within the SIP environment, but it is accepted that these will only work with that vendor's equipment. Vendors hesitate to fully disclose the exact details of these implementations, although they may be shared selectively with partners. Some vendors have implemented 100 or more SIP features in this manner. Naturally, for these features, the likelihood of interoperability between independent SIP products is very low.

■ **Private Property:** Some vendors include in their count of SIP features some things that may not even involve the SIP protocol, but which add marketplace distinction and value to the vendor's SIP-based offerings. For example, management, provisioning and configuration capabilities and applications are often lumped under a vendor's aggregate headcount of "SIP features."

An example of this is Cisco Discovery Protocol, or CDP, support, which Cisco proudly touts as a feature it has built into its new SIP phones. CDP remains Cisco-proprietary, and is unquestionably valuable in Cisco-based infrastructures for automatic configuration and monitoring. But it has no direct relationship to any SIP-standard feature.

Survey Says...

To assess the state of SIP we asked SIP vendors to respond to questions using a 1-to-5 scale. The tendency is for ambivalent respondents—who don't feel strongly one way or the other—to answer "3," the midpoint. It is the degree that the average exceeds or drops below this midpoint that gives meaning to the responses.

Figure 1 shows implementers' ratings of how solid and well developed SIP specifications currently are in about 20 functional areas related to real-time communications.

One striking result: There is no functional area that respondents universally agree is fully and clearly defined in current SIP specifications, although the "Basic Dozen" telephony features (call hold, transfer, etc.) comes closest, ranking 4.4 out of 5. Implementing voice mail in a SIP environment is also considered stable and well-defined by SIP specs, rating 4.0 out of 5.

In six other areas, the respondents indicated the SIP specifications are reasonably well defined and unambiguous to implement. These are: presence; point-to-point videoconferencing; SIP trunking; RTP voice-stream encryption; caller authentication; and personal call routing—also known as "find me/follow me." All of these rated above 3.5 on the scale, in terms of the specifications being "complete, clear and solid."

The fact that respondents indicated there had been some SIP standardization activity in all these areas was also unexpected. In a few areas, though, the responses were widely disparate: Some said sufficient standards existed, while others said there was no significant standards progress.

This could just be differences in perception. But it could also be the result of too many specifications being issued too quickly by too many organizations. Indeed, the output from a half-dozen working groups in the IETF impacts the SIP environment. (Besides SIP and SIPPING, there's also SIMPLE, for example, which addresses instant messaging and presence.) And a handful of other organizations are also involved. The SIP Forum, for example, just finalized a spec for SIP trunking between SIP PBXs and service providers. Others include the European Computer Manufacturers Association (ECMA), which developed a spec for third-party call control over SIP.

At the "not-so-well-developed" end of the SIP-standardization spectrum is "remote configuration of SIP phones." Indeed, our testing has found that, while vendors of SIP-supporting IP-PBXs—like 3Com and now Cisco—have their own management applications for configuring and monitoring their own SIP phones, generic standards-based management of third-party SIP phones is still sorely lacking.

And the standards still have a ways to go for implementing other specialized applications in SIP environments, the responses indicate. Among the least-standardized applications are: white-boarding; drag-and-drop file transfer, and call/contact center processes, like barge-in and recording.

Problems And Challenges

We asked the vendor community to rate factors and issues that might be impeding the proliferation of SIP (Figure 2). Respondents rated the significance of each, from 5, meaning "absolutely key" to impeding SIP's progress, to 1, meaning "not a concern at all." Vendors could also list and rate their own additional concerns affecting SIP proliferation, and some did.

There was general agreement on two issues:

1.) More detailed guidelines are needed on how to consistently implement SIP specs.

2.) Assuring interoperability between SIP-product vendors remains expensive and laborious.

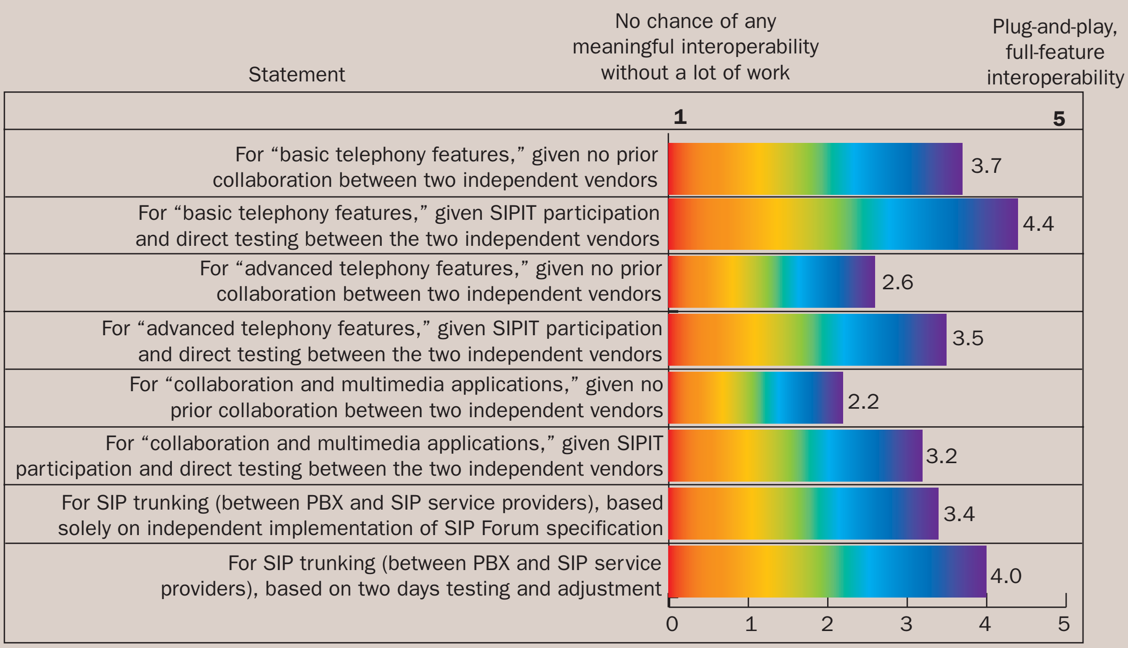
These rated 3.8 and 3.7, respectively, on the 1-to-5 scale of significance. Neither of these should be considered a surprise.

Then there were "other" concerns, which also rated high. The most frequently cited one was firewall traversal and NAT handling in a SIP environment. It is true that SIP—and most other VOIP protocols for that matter—are inherently incompatible with network address translation and tight

Respondents concede that assuring multivendor interoperability remains expensive and laborious

FIGURE 3 SIP Product Interoperability

"Assess the state of inter-vendor SIP-product interoperability given these conditions, using a 1 to 5 scale (1 = No chance of any meaningful interoperability, 5 = Plug-and-play, full-feature interopability)"



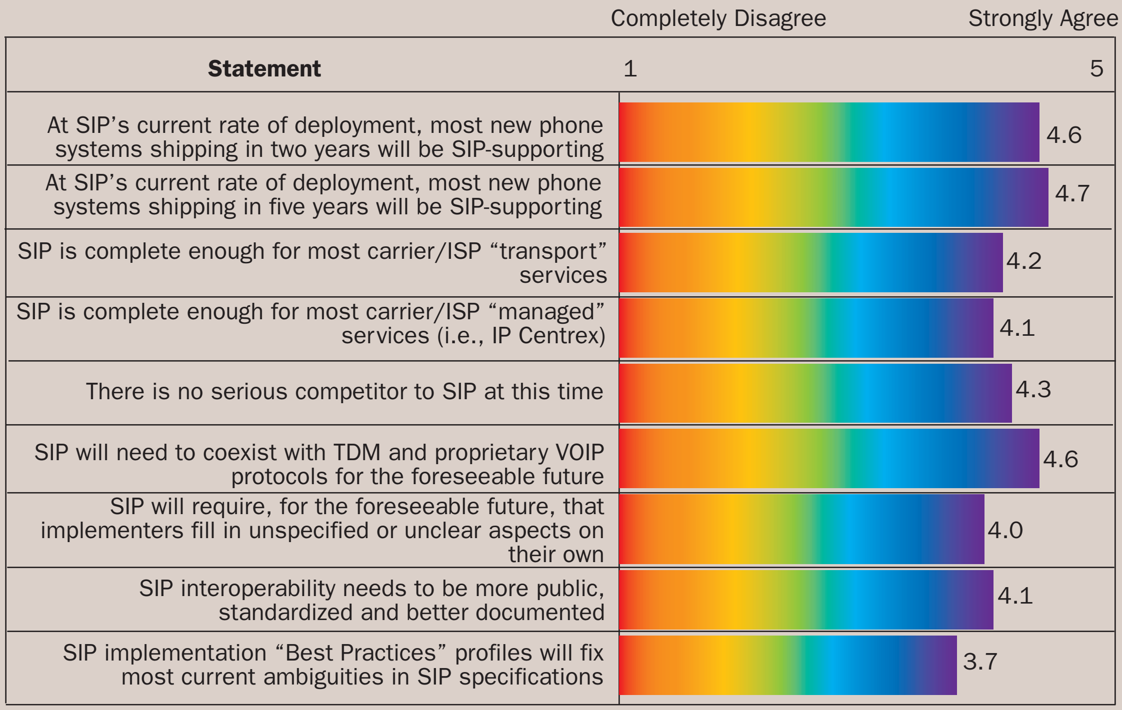


FIGURE 4 SIP's Future

"Do you agree or disagree with these statements about the future of SIP?"

firewall security. There are work-arounds, and several draft-standard solutions have been proposed. But it's clear that the SIP vendor community wants a single technical standard for handling NAT and firewall traversal.

Several by-now-familiar issues rated on the low side, meaning respondents did not consider them very significant to impeding SIP's progress. The SIP implementers indicated generally that: political interests still had not taken over the SIP standardization process, that SIP specs were no longer too unsettled, and that SIP was not sorely lacking in features customers want.

On the issue of interoperability, respondents make clear that vendors need to collaborate with each other in order to assure interoperability (Figure 3). This means that enterprise users planning to run vendor X's SIP phones on vendor Y's SIP-based PBX should make sure that vendors X and Y have collaborated to ensure interoperability.

The responses indicate that the prospects for interoperability increase 20 percent or more when the two vendors involved have collaborated. For basic telephony features, this makes the difference between a fairly positive 3.7 rating, and a definitive 4.4 rating. SIP-trunking interoperability similarly rises from 3.4, to 4.0 when the parties at either end (typically SIP PBX or gateway vendors on the enterprise and SIP service providers) have collaborated to assure interworking.

Not so promising at this time, however, are interoperability prospects for SIP-based "collaboration and multimedia applications," rated just 2.2, and "advanced telephony features," rated 2.6. As before, though, interoperability chances improve greatly in these areas when the two vendors have collaborated.

Only Game In Town?

The highest level of agreement in the survey—nearly universal at 4.7 out of 5 (5 meaning "strongly agree")—was that new phone systems shipping in five years would be SIP-supporting (Figure 4). That was just slightly above the 4.6 agreement that new systems shipping in two years would all be SIP-supporting.

This general optimism might be expected from vendors and individuals whose collective futures are keyed to SIP's success. But there was also strong agreement—4.3 out of 5—that "there is no serious competitor to SIP at this time."

Looking ahead, there was also broad agreement that SIP will need to co-exist with TDM and proprietary VOIP protocols for the foreseeable future. Then, on two other, related issues there was also general consensus. First, SIP will require, for quite a while, that implementers fill in unclear or unspecified aspects of the standards on their own. Subsequently, SIP product interoperability needs to be made more public, standardized and better documented, the respondents said.

From among the responding vendors, 17 said they currently offer a SIP-supporting PBX or call controller. These are shown in Table 1, p. 20, along with some salient points of comparison.

Seven of these PBXs run on SIP alone (including 3Com, Interactive Intelligence, and Pingtel). Eight others (including Cisco, Mitel and Siemens) now run SIP natively, alongside one or more other call-control protocol, usually the vendor's

3Com Riding The SIP Wave

Capitalizing on its strengths, 3Com is using SIP as a connectivity tool for expanding the capabilities of its mid-to-high-end VCX IP-telephony platform, and this month is extending SIP support to the SMB-class NBX, which 3Com has shipped to 28,000 sites over the last eight years. In addition, IBM in late March cited 3Com's broad native-SIP support as a key reason for tapping the VCX as the pre-integrated IP-telephony component on IBM's System i-based "all-in-one" business computing package.

The Miercom team exercised the SIP capabilities of the NBX. With the pre-release version of NBX 6.0 we evaluated, 3Com adds SIP as a new call-control protocol stack to the system. We verified smooth interworking between NBX IP phones, which now support conventional VOIP RTP streams and third party SIP-based phones, including Cisco's 7960 and the popular "eyebeam" SIP-based softphone, from CounterPath Solutions.

Another aspect of SIP we exercised at the vendor's Marlborough, MA-based labs was internetworking between NBX and big-brother VCX systems via SIP trunks. Calls between SIP phones on the VCX and NBX phones were clean and seamless, with top-quality connections, using a common dial plan. A forthcoming release, not yet reviewed, will enable 3Com 2100-series and 3100-series phones on NBX systems to download and run SIP, too.

3Com continues to expand the connectivities, capabilities and add-on options of both the NBX and VCX systems—all enabled by SIP trunking. Among the ones we reviewed were: new high-redundancy SIP-PSTN gateways; gateways that let the 3Com SIP environment interwork with legacy PBXs and TDM/digital phones; the latest version of 3Com's IP Messaging and IP Conferencing systems—which deliver video, IM, softphone and all manner of productivity features via a slick new desktop client; and the multimedia IP Call Center, supporting hundreds of agent seats. Another useful component we viewed is the 3Com Telecommuter Module, which acts as a SIP firewall and handles all VOIP processing for remote teleworkers.

3Com, a charter member of the SIP Forum, has been carrying the SIP torch longer than most competitors. The vendor makes clear that it has implemented as much functionality in its IP-communications portfolio as current SIP specifications now allow, and expects to add a lot more as the standard grows□

Cisco: New Frontiers Via SIP

One of the biggest announcements recently was Cisco's unveiling a few months ago of its new, broad, SIP-based Unified Communications System—a new generation of IP-telephony offerings, coupled with a wave of new SIP-enabled capabilities, connectivities and applications.

SIP is hardly new with Cisco. The vendor has long offered an alternative SIP software stack for its 7900 series IP-phones—widely used by other vendors to demonstrate their products' SIP interoperability. And Cisco has been shipping various IOS-based products for SIP traffic handling, ranging from proxy servers to SIP gateways.

At the core of Cisco's new SIP environment, two years in the making, is the Unified CallManager 5.0. The new call controller ships on Linux-based appliances and includes native SIP support, right alongside the legacy Skinny (SCCP) protocol. More importantly, virtually all Cisco's hundreds of phone features are now also supported in the SIP environment. Otherwise, the new SIP environment is as robust as before, with high availability via clustered servers, Secure Remote Site Telephony (SRST) fail-over, load balancing and scalability up to 30,000 IP phones per cluster.

The Miercom test team confirmed that SIP/Skinny interoperability—between legacy Cisco phones and its new line of SIP-oriented IP phone sets—is seamless, full-featured and user transparent. What's more, all previous security capabilities, including certificates, signed firmware loads, and media (RTP) and signaling encryption, are fully supported in the Cisco SIP world, as well as between SIP and SCCP phones.

Cisco also exploited new capabilities that its new SIP environment enabled. One of these is presence, where users can readily see the availability of other users on the system. In fact, Cisco extended this presence support to also work with SCCP phones, so users can implement presence while migrating to SIP at their own pace. The new Cisco Unified Presence Server handles enhanced presence processing.

The SIP environment has spawned new applications including the Cisco Unified Personal Communicator, a slick, pure-SIP client application, with versions for Windows and Mac users. Via a compact, expandable, and very effective interface, the user gets desktop video, softphone, call logs, presence, Web collaboration and client access to Unity messaging and MeetingPlace□

TABLE 1 SIP-Supporting IP-PBXs (1)

Vendor	IP-PBX/Call Controller	SIP role	SIP features	SIP security (2)
3Com	VCX	Native SIP solely	175	DA; adding TLS, sRTP in summer
Adtran	NetVanta 7100	Native SIP solely	30+	none
Avaya	SIP Enablement Services (SES)	SIP-based subsystem (w/ Avaya Comms Mgr)	60+	TLS, DA
Cisco	Unified CallManager, CallManager Express	Native SIP, dual-stack (w/ Cisco SCCP, others)	170+	TLS, sRTP, DA
Dialexia	Dial-Office	Native SIP solely	20	none
Digium	Asterisk Business Edition	Native SIP, dual-stack (w/ Asterisk IAX)	—	TLS, sRTP
Interactive Intelligence	Customer Interaction Center (CIC)	Native SIP solely	—	adding TLS and sRTP in late fall
Mera Systems	Mera IP PBX	Native SIP, dual-stack (w/ H.323)	''All critical PBX features"	DA
Mitel	3300 ICP	Native SIP, dual-stack (w/ Mitel MiNet)	5; over 300 via feature codes	TLS, sRTP
NEC Unified	Univerge	Native SIP, dual-stack (w/ IP Protims)	All RFC 3261 plus SIP extensions	sRTP
Nortel	Multimedia Comms System 5100	SIP-based subsystem (w/ Nortel CS 1000)	450+	adding TLS and sRTP later in '06
Personeta	TappS NSC	Native SIP	—	TLS, sRTP
Pingtel	SIPxchange ECS	Native SIP solely	20+	TLS, sRTP
Siemens	HiPath 8000	Native SIP, dual-stack (w/ H.323 and MGCP)	200+	TLS
Swyx	SwyxWare	Native SIP, dual-stack (w/ H.323 and others)	—	—
Veraz	ControlSwitch	Native SIP, multi-stack (w/ H.323, others)	end-device dependent	TLS
Zultys	Enterprise Media Exchange (MX)	Native SIP only	100+	sRTP

(1) Based on responses received by April 20, 2006 to a survey emailed to all SIP-supporting vendors known to Miercom.

(2) TLS = Transport-Layer Security (encrypted SIP call control), sRTP = Secure Real-time Transport Protocol (encrypted voice streams), DA = Digest Authentication, in most cases using MD5 algorithm (verifies call-control message and sender identity).

TABLE 2 SIP Gateways (1)

Vendor	Gateway(s)	Gateway(s) link SIP with:	Number of SIP features supported	Interop with third-party SIP equipment, software vendors, service providers
3Com	Media Gateways	PSTN	12	4 cited
Adtran	Total Access 900	PSTN	10	3 cited
Cisco	SIP/PSTN Gateway on Cisco IOS routers	PSTN	170+	Cisco Unified IP-telephony; no third party specified
Dialexia	Dial-Office SIP to PSTN Gateway	PSTN	not specified	14 cited
FirstHand Technologies (formerly SIPquest)	Enterprise Mobility Gateway/Server	Mobile platforms: WiFi, BlackBerry, Symbian, others	15+	Testing in progress
Mitel	SIP-to-PSTN Gateway	PSTN	About 20	14 cited
MultiTech	MultiVOIP	PSTN	About 6	13 cited
Siemens	RG 8700	PSTN; legacy PBXs via Q.Sig (SIP-Q)	200+ (driven by PBX)	9 cited
Swyx	SwyxGate	PSTN	30	1 cited
Zultys	MX25	PSTN	100's	10 cited

(1) Based on responses received by April 20, 2006 to a survey emailed to all SIP-supporting vendors known to Miercom.

predecessor, proprietary protocol, such as Cisco's "Skinny" or Siemens' flavor of H.323.

For two telephony stalwarts—Avaya and Nortel—SIP is still kept at arm's length from their mainstay telephony engines—Avaya's Communication Manager and Nortel's Succession series.

Avaya makes clear that Communication Manager is moving closer and closer to SIP with each new software release. For now, though, SIP-endpoint control in the Avaya environment is via the vendor's SIP Enablement Services (SES) subsystem. And SIP handling in Nortel's world is via the vendor's Multimedia Communications Server (MCS) subsystem.

We note that the surveyed vendors are diverse in how many SIP features they support, and also in the extent they have implemented components of SIP security. These components generally include: Digest Authentication, Transport-Layer Security (TLS) and Secure RTP. Security in the SIP environment is a new and very recent addition. We did not ask whether the vendor claims its SIP security works with, or is expected to interoperate with, any other third-party SIP vendors.

Another key component of SIP networks is gateways, which manage SIP calls to and from other environments, notably the PSTN (Table 2). A noteworthy exception is FirstHand Technologies (formerly SIPquest), which offers a specialty gateway, and software that runs on various mobile devices, including BlackBerry. The package extends SIP-based applications and environments out over GPRS-type data networks, which, due to their nature and bandwidth, do not generally support full SIP networking.

SBCs And More

Our survey turned up a number of vendors offering session border controllers, or SBCs (Table 3). The SBC is a new device in SIP networking, and a testament to the fact that SIP is spreading. Where two SIP "domains" meet—which may be

TABLE 3 SIP Session Border Controllers (1)

Vendor	SBC	Key SBC features, capabilities (2)	Interop with third-party SIP equip, software vendors, service providers
Acme Packet	Net-Net SBC	• DoS overload protection • SIP-H.323 interworking • NAT/firewall traversal • Call admission control • SLA compliance	30+ equipment, software vendors, and/or service providers cited
Cisco	SBC on Cisco IOS routers	• SIP-H.323 conversion • NAT traversal • Transcoding • Authentication • TLS security	Not specified
Ingate	SIParator	• Near-end and far-end NAT/firewall traversal • Encryption and transcoding of media and/or signaling • Advanced routing • Authentication of both servers and clients	30+ equipment, software vendors, and/or service providers cited
Intertex	IX67 SIP-Capable Firewall	• SIP-capable NAT/firewall • Built-in SIP proxy and registrar • Secure RTP • Designed for SOHO environment	Not specified
Mera Systems	VOIP Transit Softswitch	• Conceals network topology • H.323-SIP interworking • Load balancing, fault tolerance • Advanced call routing • Gatekeeper and proxy	5 cited
Ranch Networks	RNXX appliance	• NAT traversal • Firewall, DoS attack protection • Rate limiting, bandwidth guarantee • QOS • Media bridging	Integrates with Asterisk
Sipera	IPCS 310	• TLS and sRTP support • DoS protection • Stealth, spam and spoofing protection • Also secures IM and video streams	3 cited

(1) Based on responses received by April 20, 2006 to a survey emailed to all SIP-supporting vendors known to Miercom.

(2) TLS = Transport-Layer Security (encrypted SIP call control); sRTP = Secure Real-time Transport Protocol (encrypted voice streams); SLA = Service Level Agreement; DoS = Denial of Service; NAT = Network Address Translation (typically by firewall)

Avaya: Putting SIP To Work

SIP is changing the way things are done at Avaya. Take Avaya Communication Manager, or CM, the software brain of Avaya's phone systems. Today's CM contains a full SIP back-to-back user agent, and can connect via direct SIP trunks with other CM servers, as well as SIP-based carrier services. Increasingly, CM is being viewed internally at Avaya as a "telephony feature server," which controls and delivers Avaya's hundreds of telephony features via classic H.323, or over SIP, as well as to legacy TDM phones—or any combination of these concurrently.

For now, "back-end" handling of SIP endpoints involves Avaya SIP Enablement Services, or SES, a standalone Linux appliance, which embodies a SIP proxy and registrar. SES handles SIP call routing, endpoint registration, and SIP presence and instant messaging. Today, the SES server connects via a SIP trunk directly to the Communication Manager. But Avaya's plan is to incorporate the SES as a subsystem right alongside CM, all on the same platform.

"In the next year or so, Avaya customers will be able to deploy an all-SIP solution, which gives you all the features of Communication Manager," said Venky Krishnaswamy, director of IP Communications Research at Avaya. "An all-SIP environment won't be 100 percent transparent with the way things are done today," he said, adding that SIP brings new capabilities and features of its own, "like the ability to transcend a single PBX to deliver features across a multisystem network. SIP gives you natural mobility."

Avaya is well along its SIP-implementation roadmap. It has already deployed all-SIP production networks, in one case with 10,000 users, the vendor said. Full SIP-based support for *all* CM's features is in the works, as is full SIP-based security—via Secure RTP audio streams and TLS-based call control. In fact, SIP call control between Avaya servers is already secured via TLS.

What's more, dozens of SIP-based third-party products have already successfully completed Avaya's "DevConnect" certification. This program, entailing several days of testing at Avaya and a modest fee, ensures that the third-party SIP products can successfully tap some 60-plus features. Many of these are accessed by simply sending feature codes in SIP messages to the Communication Manager.

At the low end, Avaya already offers customers an all-SIP package—in the form of Avaya one-X Quick Edition. This is a SIP peer-to-peer system, supporting up to 20 or so stations, which resulted from Avaya's acquisition last year of Nimcat. Avaya has integrated this package within its broader SIP road map in short order. Miercom reviewed the interworking of Avaya's Quick Edition—deployed as a branch office telephony system—with other Quick Edition branches and with other SIP phones via an Avaya SIP Enablement Server and SIP trunks, and with Avaya H.323 and TDM phones on a Communication Manager.

All the features we tested across these SIP-connectivity environments worked transparently. Based on the progress we've seen within the last year, SIP at Avaya seems to be on track and on target□

Interactive Intelligence: SIP-based Contact Center

Indianapolis-based Interactive Intelligence has a compelling SIP story to tell. It started about five years ago when the vendor began migrating its advanced contact center software package—Customer Interaction Center (CIC)—from a traditional TDM-based platform to run over IP. Interactive decided that the newly emerging SIP standard was the way to go, and in 2002 it created its own SIP stack and deployed its first VOIP-based CIC version.

"SIP enabled us to replace the TDM architecture with a truly open IP network," said Duke Snyder, telephony architect at Interactive. Today, SIP underlies all of Interactive's wares, including Communité, a rich voice mail, IVR and unified messaging package, and Enterprise Interaction Center (EIC), a Microsoft-based IP communications system for enterprises, offered by the company's subsidiary, Vonexus Inc.

Since deploying SIP, CIC's capacity has grown significantly. The package now handles up to 5,000 users, and that will expand over the next few months up to 15,000 users. SIP's flexibility for routing calls in progress, along with a new, high-capacity media server, are what made this increase possible. Rather than a single server for call processing and recording, SIP let the vendor move these functions onto separate servers, dynamically routing and redirecting calls among them as needed. This also significantly bolstered system reliability by automatically re-allocating call load if a server should fail.

Currently, Interactive supports two dozen SIP phones from a dozen vendors, multiple SIP gateways, and direct SIP trunk connections with a dozen service providers. The company says it will remain a forerunner in the adoption of SIP with plans to support secure RTP encryption and TLS-based call security, and expanded support for SIP-based presence and multimedia capabilities.

"Unlike many PBX vendors that use proprietary SIP extensions, which limits certain functionality to their own systems and devices, we decided from the start to incorporate SIP—standard SIP—everywhere we could," said Snyder.

Miercom exercised the vendor's all-SIP software package, complete with pre-integrated converged voice and data applications, and confirmed its impressive scalability, and support for a number of third-party phones, gateways, proxy servers and other SIP-based devices. In Interactive's case, SIP is not only the *de facto* open standard for VOIP, but the latest enabler of new, productivity-enhancing communications applications□

between two enterprise organizations or between an enterprise and a SIP-based carrier—the SBC has emerged as the way to mediate this "peering." The table shows the key features these packages address. They include: caller authentication, NAT traversal, DoS protection, QOS processing and conversion between SIP and legacy H.323 environments.

More than a dozen providers of SIP endpoints are listed in Table 4. As noted earlier, this has become a marketplace free-for-all: Vendors are offering a broad assortment of SIP endpoints—including SIP hard phones, SIP softphones, video phones, and SIP WiFi/mobile devices. Some SIP hard phones now cost less than $100, and pricing is definitely headed down.

Table 5, p. 24, concerns what SIP can do for end users. More than a dozen SIP-based application packages and servers are shown. The most common applications delivered by these systems are: presence; audio conferencing; desktop sharing; Web collaboration; buddy lists/workgroups; and unified messaging.

We asked the vendors to specify what third-party SIP PBXs or call controllers their systems worked with. Those are totaled in the last column of Table 5. IP Unity provided details on more than 20 SIP-supporting IP-telephony systems that its Mereon package works with.

Conclusion

A detailed survey of SIP-supporting vendors yielded about 40 responses, representing about half of the estimated 80 vendors of SIP-supporting

TABLE 4 SIP Phones (1)

Vendor	SIP hard phone(s)	SIP softphone(s)	Number of SIP features supported	Interop with third-party SIP vendors
3Com	Models range from entry-level to large-screen with Java support	Converged Center Client: includes IM, audio and video conferencing	65	32 cited
Avaya	4600 series models support SIP and H.323	One-X Desktop; also, hybrid SIP/H.323 softphone	Not specified	10 cited
Cisco	SIP endpoints include wireless, video; basic SIP firmware available for most models	Unified Personal Communicator: featuring presence and collaboration	170+	Most based on RFC 3261 and SIPPING
Dialexia	—	Dial-Console: for switchboard operations; uses drag and drop to manage calls	Most basic features	Not specified
FirstHand Technologies (formerly SIPquest	—	Mobile Assistant and Mobile Console SIP clients run on mobile platforms	25+	3 cited
Interactive Intelligence	Rebrands SIP hard phones, SIP wireless	Interaction Client: features presence, record	Not specified	1 cited
Mitel	5200 series SIP hard phones	Navigator; SIP soft phone	35+	5 cited
NEC Unified	Dterm series support SIP, includes wireless, video endpoint models	Dterm family softphone	500+ on vendor's PBX	9 cited
Nortel	IP Phone 2004; wireless and video SIP support	Multiple soft clients, PC and Web based versions; one integrates with Outlook, another runs on BlackBerry; a dual-mode mobile client	450+ on vendor's PBX	5 cited
Polycom	SoundPoint IP; SoundStation IP, VSX 7000; many SIP models, including video	—	Not specified	10+ cited
Siemens	optiPoint 420 S family of SIP phones and optiPoint WL2 S is a wireless SIP phone	optiClient 130 S is a full, SIP soft client	73, plus 23 more on vendor's PBX	12 cited
Swyx	—	SwyxIt!, a Windows-based softphone	30	30 service providers
Zultys	Various SIP models, including remote-office SIP phone	Not specified	Not specified	Various service providers

(1) Based on responses received by April 20, 2006 to a survey emailed to all SIP-supporting vendors known to Miercom.

TABLE 5 Open SIP Application Platforms (1)

Vendor	Application package	Key features	Interop with third-party SIP systems
3Com	3Com Convergence Application Suite; hardware and software; enterprise oriented	• Conferencing, audio and video (P-P) • Presence • Buddy lists/workgroups • Click to call • Desktop sharing • Web/desktop publishing	6 cited
Adomo	Adomo Voice Messaging; appliance, enterprise oriented	• Presence • Buddy lists • Click to call • Unified messaging and voice mail • Speech-based auto attendant, directory • Find me, follow me	2 cited
Antepo	OPN System; software, carrier-oriented	• Presence • Buddy lists • Click to call • Desktop sharing • Whiteboard • Instant messaging • Group chat forums	4 cited
Avaya	Meeting Exchange; hardware and software, enterprise oriented	• Conferencing, audio • Web collaboration • Presence • Whiteboard •Click to call • Desktop sharing • Web/desktop publishing	8 cited
Cisco	Cisco Unified Presence Server and Personal Communicator, enterprise oriented hardware and software	• Conferencing audio and video • Presence • Web collaboration • Click to call • Desktop sharing • Whiteboard	1 cited
Continuous Computing	Trillium SIP; carrier/ service provider oriented software	• Generic base-server software package for SIP-based applications	no specific third-party SIP systems cited
Dialexia	Dial-Office DXO Series; enterprise oriented, hardware and software	• Conferencing, audio and video (P-P) • Presence • Buddy lists • Unified messaging	5 cited
FirstHand Technioligies (formerly SIPquest)	Conferencing Server; enterprise oriented, software	• Conferencing, audio and video • Presence • Buddy lists • Click to call • Desktop sharing	1 cited
Genesys Labs	SIP Server; enterprise or carrier, software	• Contact Center application suite	6 cited
Interactive Intelligence	Customer Interaction Center (CIC); hardware and software, enterprise oriented	• Conferencing, audio and video (P-P) • Web collaboration • Presence • Buddy lists • Click to call • Contact Center (ACD, recording, IVR, etc.)	no specific third- party SIP systems cited
IP Unity	Mereon; carrier or enterprise, hardware and software	• Conferencing, audio and video (multipoint) • Web collaboration • Buddy lists • Click to call • Desktop sharing • Whiteboard • Web/desktop publishing	20+ cited
Mera Systems	Mera SIPrise; enterprise or carrier, software	• Conferencing, audio • Web collaboration • Presence • Buddy lists • Click to call • IVR applications/services	no specific third-party SIP systems cited
Microsoft	Microsoft Office Live Communications Server (LCS) 2005, and Office Communicator, carrier or enterprise, software	• Conferencing, audio and video • Presence • Web collaboration • Buddy lists/workgroups • Click to call • Desktop sharing • Whiteboard	8 cited
NEC Unified	Dterm SP30, enterprise oriented, software	• Conferencing, audio and video (P-P) • Web collaboration • Presence • Buddy lists • Click to call • Desktop sharing • Whiteboard • Web publishing • MS Outlook plug-in	no specific third-party SIP systems cited
Nortel	Multimedia Communications Server (MCS) 5100/5200, many soft client versions and options; carrier or enterprise, software	• Conferencing, audio and video • Presence • Web collaboration • Buddy lists/workgroups • Click to call • Desktop sharing • IM/Chat • Whiteboard • Web publishing • Recorded announcements	no specific third-party SIP systems cited; various third-party SIP endpoints
Personeta	TappS applications; carrier and service provider, software	• Conferencing, audio and video • Presence • Buddy lists • Click to call • Mobile office	no specific third-party SIP systems cited
Siemens	HiPath OpenScape, with Microsoft LCS 2005, enterprise oriented, software	• Conferencing, audio and video • Presence • Web collaboration • Buddy lists/workgroups • Click to call • Desktop sharing • Whiteboard • Web publishing • Unified messaging • Speech access to MS Exchange	no specific third-party SIP systems cited
Swyx	SwyxIt! Now; enterprise or carrier, software	• Conferencing, audio • Presence • Click to call	no specific service providers cited
Veraz Networks	Veraz ControlSwitch; carrier oriented, hardware and software	• Conferencing, audio and video (P-P) • Click to call	2 cited

(1) Based on responses received by April 20, 2006 to a survey emailed to all SIP-supporting vendors known to Miercom.

Microsoft: SIP-based LCS, Office Communicator

Instant Messaging in some enterprises is secure and internal only, and therefore isolated in terms of connectivity. With Microsoft's Live Communications Server (LCS), however, you can integrate users from public IM networks including AOL, MSN and Yahoo natively. Affiliated, or "federated," enterprises, each running LCS, can securely communicate and share presence information through the wilds of the Internet.

And all of this runs over SIP. In fact, since its first SIP implementation in Windows XP in 2001, Microsoft boasts it has delivered native SIP support to more than 200 million Windows desktops worldwide.

Miercom reviewed the broad, SIP-enabled capabilities of Microsoft's preferred SIP client—Office Communicator 2005, now less than a year old—and the corresponding server software, Live Communications Server 2005. The LCS-Communicator combo delivers: secure instant messaging; click to call; secure voice communications via integrated softphone; integration with Microsoft Office, Exchange Server and Live Meeting; and very configurable PC-to-PC video. Besides the desktop version of Communicator, there's also now the Communicator Mobile version for Windows Mobile devices, and Communicator Web Access version for browser access from anywhere.

But among the real innovations is Microsoft's presence implementation. Built into the system is a presence server, integrated with Active Directory, which determines the availability of all users—and displays the status using a handful of simple icons. An intricate set of rules, inputs and settings for presence determination include: Outlook calendar entries; user-settable custom notes; "idle" time away from PC; and whether currently on the phone or in a video call.

What's more, various PBX vendors use SIP and Microsoft APIs to integrate LCS with their PBX environments. This enables features including: remote call control (switching between the PC softphone and associated, external hard phones); rules-based call forwarding; pop-up notification of incoming PBX calls; PBX-based calling from the Communicator PC; and integration with corporate directory and Outlook contacts for reverse lookups.

At last count, Microsoft had landed more than a dozen IP-telephony partners, who were integrating IP-PBXs with LCS and Communicator; several multiparty audio- and videoconferencing vendors; multiple server vendors, for load-balancing and high-availability topologies; and dozens of other hardware vendors, system integrators and vertical-industry suppliers. And the list is growing daily□

Polycom: Innovating SIP Endpoints

Milpitas, CA-based Polycom Inc. is a pioneering provider of IP endpoints that deliver real-time communications, voice as well as video, to user desktops. The vendor is probably best known for its distinctive three-legged SoundStation speakerphones, found in practically every conference room. The SIP version of that phone shipped nearly 18 months ago. Today, the overwhelming majority of its new phone shipments are SIP, the company said.

From a SIP perspective, Polycom plays an atypical role. While the company makes IP endpoints, it doesn't make call controllers—the IP-PBXs, proxy servers, and such—that drive the endpoints. So the vendor works closely with SIP-supporting IP-PBX vendors—such as Interactive Intelligence, Vonexus, Nortel MCS and Adtran; with leading vendors of application and call servers for service providers—including Sylantro Systems and BroadSoft; and even with Digium and Pingtel, who offer enterprise versions of SIP open-source packages.

It is a delicate balancing act for Polycom: Creating innovative endpoints that provide the richest end-user experience possible; doing as much of that within the scope of solid SIP specifications as technically possible; building in additional features and capabilities that partners seek; and ensuring their endpoints and all these vendors' call servers work together.

How's it working out? Pretty well, based on the latest Polycom innovations we reviewed. A new low-end SIP phone, the SoundPoint IP 430, delivers a full-duplex speakerphone, integral power over Ethernet (PoE) support, graphical display—the works—for $239. We reviewed the vendor's latest SoundPoint IP 601 phone, featuring a "microbrowser" for converged applications. And we exercised the vendor's new plug-and-play Expansion Modules, which expand the IP 601 phone with up to 42 additional line appearances and soft buttons. The modules slide on and connect via a slick infrared link; no additional power or cables needed.

SIP has permeated the video side of Polycom's business, too, but not yet to the same degree as voice phones. That's because SIP doesn't yet formally support some features key to videoconferencing—like enabling one endpoint to control the pan, tilt and zoom of the other endpoint's camera—that H.323 does. Still, responding to demand, Polycom's video endpoints are now also SIP-enabled. Furthermore, Polycom is actively working in the IETF to bring needed videoconferencing extensions to SIP. And the vendor ships a family of special bridge units that let users videoconference concurrently between SIP, H.323, even ISDN-based endpoints.

More new and useful SIP capabilities are being prepared for upcoming Polycom products and new releases. Some of what's ahead: SIMPLE presence-based features, full Microsoft LCS 2005 integration; and new SIP-based security via TLS-encrypted call control□

Citel: Migrating From TDM To SIP

Migrating to IP-telephony doesn't necessarily mean buying all new IP phones.

Citel Technologies, based in Nottingham, UK, has an answer for those who ask: "Why do I have to throw out my old TDM phones? They ain't broke. They're paid for. And everyone knows how to use them." Indeed, there's an estimated 300 million TDM phones out there, representing billions of dollars in telephony equipment.

Enter the CitelLink SIP Handset Gateway, a thin, 1-U appliance that lets users plug in up to 12 or 24 TDM phone sets, and go anywhere in the SIP world. At last count Citel supports over 70 leading TDM phone models—all the classic Nortel Meridian 1 and Norstar models, Avaya Definity sets, TDM phones from Ericsson, NEC, Toshiba and more.

"When that phone plugs into our box, it's no longer a TDM phone, it becomes a SIP phone," said Leigh Fatzinger, vice president of marketing in Citel's U.S. headquarters in Seattle. In the wiring closet, the user unplugs the TDM phones from the legacy PBX and, via the same RJ-21X connector, plugs them into an appropriately configured Citel Handset Gateway.

On the SIP side, the Handset Gateway translates all the proprietary TDM-phone-set signals into standard-SIP equivalents. The unit powers the TDM phones, and can even bridge multiple streams for conference calling by TDM-phone users.

Miercom reviewed the Citel gateway in a lab set-up where classic Nortel M2616 sets were working off a classic Meridian 1, Option 11, running classic Release 22 code. The phones were unplugged from the Meridian line card, and re-plugged into a Citel Handset Gateway. After a 30-second reboot, the phones came back up and—quite imperceptibly to users—were working on an Asterisk public-domain SIP-based IP-PBX. Call hold, transfer, even low-latency call quality all worked the same. A teeny difference we saw was that the date-time was on the right side of the phones' LCD display, instead of the left. And even that can be adjusted, we noted, via a very clean Web management interface to the Citel gateway.

The vendor's SIP environment is growing fast. Besides Asterisk, supported IP-PBXs now also include Pingtel, Avaya's SIP subsystem, 3Com and Zultys, as well as the call controllers and application servers of Sylantro and BroadSoft, which drive many service providers' "managed" IP-telephony offerings. The gateways cost $130 per port list, for 12- or 24-port models.

The vendor also offers the EXTender, a family of specialized gateways that let TDM phones connect with legacy TDM PBXs over an IP network—taking advantage of IP's reach and distance insensitivity. The latest EXTender IP6000 model can be deployed in this role initially, and then flash-upgraded to become a Handset Gateway when the legacy PBX is finally replaced with SIP-based IP-telephony□

FirstHand: Mobilizing SIP Apps

What do SIP and cell phones have in common? Up until now, not much. But Kanata, Ontario-based FirstHand Technologies (formerly SIPquest) is changing that. The Canadian software upstart has developed a way of extending the services and applications of SIP-based PBXs—things like presence, conferencing, consolidated call logs and voice mail, PBX-originated toll-call set-up—out over public cellular networks to "smart phone" and "dual-mode" (WiFi/cellular) users.

FirstHand's products are specialized software modules. Miercom reviewed several of these, running on a plethora of cell phones and other mobile devices, in a variety of network environments.

There's the Mobile Assistant, a Java application that runs on wireless platforms including BlackBerry, Symbian and Windows Mobile. This efficient software, just 300 Kbytes, runs on the smart phone and communicates over the cellular data network, then through a special FirstHand gateway in the enterprise network; this gateway links into the enterprise's SIP-based environment. A special, secure SIP-like protocol is used, which makes efficient use of the network bandwidth. Cellular phones, even smart phones, lack the processor power or memory, as well as sufficient battery power, to run as full SIP endpoints.

What does the FirstHand software buy you? A lot. It makes the cellular smart phone a go-anywhere extension to the enterprise's IP-PBX. Via a very usable interface on the smart phone, we invoked a spectrum of features driven by the remote enterprise IP-PBX and associated SIP servers. These included: call screening, PBX call initiation (a tremendous saver of cellular call minutes), corporate directory access, presence for all enterprise users, instant messaging, click to conference, and voice mail message waiting indication (MWI).

Then there's FirstHand's Mobile Console, a 1-Mbyte SIP softphone that runs on Windows Mobile platforms. The software delivers to WiFi and dual-mode users all of the same features and capabilities as the Java-based Mobile Assistant, plus support on the mobile device for PBX features including call hold, transfer and redial.

Miercom reviewed the Mobile Console running with the Windows clients of several notable SIP-based call controllers and application servers, including Nortel's MCS, BroadSoft and 3Com's VCX. Others are on the way, FirstHand said. By extending the SIP environment out to mobile users, this gateway does the same thing for SIP calls and services that the BlackBerry Enterprise Server does for email□

products. The survey asked about SIP trends, and about the respondents' products.

SIP is definitely spiraling upwards, but the responses reveal issues in the adequacy and completeness of the current SIP standards and specifications. Also, interoperability for advanced SIP features and applications is still problematic. To assure a clean deployment, enterprises should be sure two SIP vendors have worked out interoperability issues between themselves.

Also, users pursuing SIP will find the "features" issue confusing. Comparing SIP systems based on supported SIP features can be frustrating and misleading. Features are implemented in various ways and with varying prospects for third-party interoperability□

Companies Mentioned In This Article

3Com (www.3com.com)
Acme Packet (www.acmepacket.com)
Adomo (www.adomo.com)
Adtran (www.adtran.com)
Antepo (www.antepo.com)
Avaya (www.avaya.com)
Cisco (www.cisco.com)
Citel (www.citel.com)
Continuous Computing (www.continuouscomputing.com)
Dialexia (www.dialexia.com)
Digium (www.digium.com)
Genesys Labs (www.genesyslab.com)
Ingate (www.ingate.com)
Interactive Intelligence (www.inin.com)
Intertex (www.intertex.se)
IP Unity (www.ipunity.com)
Mera Systems (www.mera-systems.com)
Microsoft (www.microsoft.com)
Mitel (www.mitel.com)
Multi-Tech Systems (www.multitech.com)
NEC Unified Solutions (www.necunified.com)
Nortel (www.nortel.com)
Personeta (www.personeta.com)
Pingtel (www.pingtel.com)
Polycom (www.polycom.com)
Ranch Networks (www.ranchnetworks.com)
Siemens (www.siemens.com)
Sipera Systems (www.sipera.com)
SIPquest (www.sipquest.com)
Swyx (www.swyx.com)
Veraz Networks (www.veraznetworks.com)
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