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# SIP: A Mid-Course Assessment

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**The results of the latest SIP industry survey are in. SIP's role is becoming clearer, but controversy lingers like so many proprietary protocols.**

SIP is coming of age. Oh, it's still a long way from mature. But the industry seems closer than it was just a year ago to settling on what SIP (the Session Initiation Protocol) will be when it grows up.

That's the conclusion we draw from our second annual "State of SIP" assessment, based on ongoing research, hands-on reviews of the latest SIP-based wares and features, and a detailed survey of the vendor technical community, conducted by MierConsulting and Miercom in association with *Business Communications Review*.

Late last year we contacted all vendors of products based on or involving SIP, of which we were aware—about 90 in all. We emailed them a detailed survey, and by the deadline for replies, about 36 completed, validated surveys had been received. The responses tallied here represent a bit more than one-third of today's SIP-supporting vendor industry. These are the technical influences that drive the SIP standards, and then implement them in (hopefully) interoperable products.

One-third of the total population is a considerable sample size for any survey, and this tends to yield a fairly small margin of error. Indeed, we saw that the responses in many areas were statistically the same, plus or minus a few percent, as in last year's survey—despite the fact that many of the responding vendors were different.

But in a few areas there were noteworthy changes, and not all of them were positive. For example, SIP's completeness and adequacy for carrier services dropped, from a very positive rating of 4.1 last year, to 3.7 this year, a 10 percent decrease. (A 5 indicates total agreement; a 1 means general disagreement.) We didn't ask for elaboration, so we can only speculate as to why the difference. Perhaps as carriers and service providers attempt to deliver more and more via SIP, they are finding limitations, or perhaps are discovering new and more complex interoperability concerns.

There were several other such subtle indicators. For example, fewer respondents this year agreed with the statement that "most new phone systems in two years will be SIP-supporting." On the other hand, optimism hit an all-time high—a rating of 4.8 out of 5—in agreement that most new phone systems in *five* years would be SIP-supporting.

Like last year, 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.

Figure 1 shows how complete, clear, and stable SIP implementers regard the current volume of SIP standards, as embodied in a dozen RFCs and other specifications—ranging from implementation profiles to still-baking "Internet Drafts."

There is still not *complete* agreement that the specs even for implementing the "basic dozen telephony features" via SIP are totally nailed down. The current rating, 4.4, is exactly the same as a year ago.

Indeed, most of the ratings in Figure 1 are within a few percent of their comparable ratings last year. But there are some noteworthy differences as well:

- We asked this year, for the first time, about the adequacy of the SIP specs for firewall and NAT traversal by SIP control messages and media streams. The responses: a so-so average rating of 3.2 for firewall traversal and 3.3 for NAT (network address translation) traversal. In other words, some specs are out there, and improvement is seen, but there's still a way to go.

Jay Batson, managing director of the SIP Forum, agrees. He noted that getting SIP through

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**FIGURE 1 The State Of SIP Specs— 2007**

"Rate the state of current SIP specifications (from all sources; IETF SIP/SIPPING, SIP Forum, etc.) for implementing these features and functions, using a 1 to 5 scale (5=Complete, clear, solid, unambiguous, stable)"

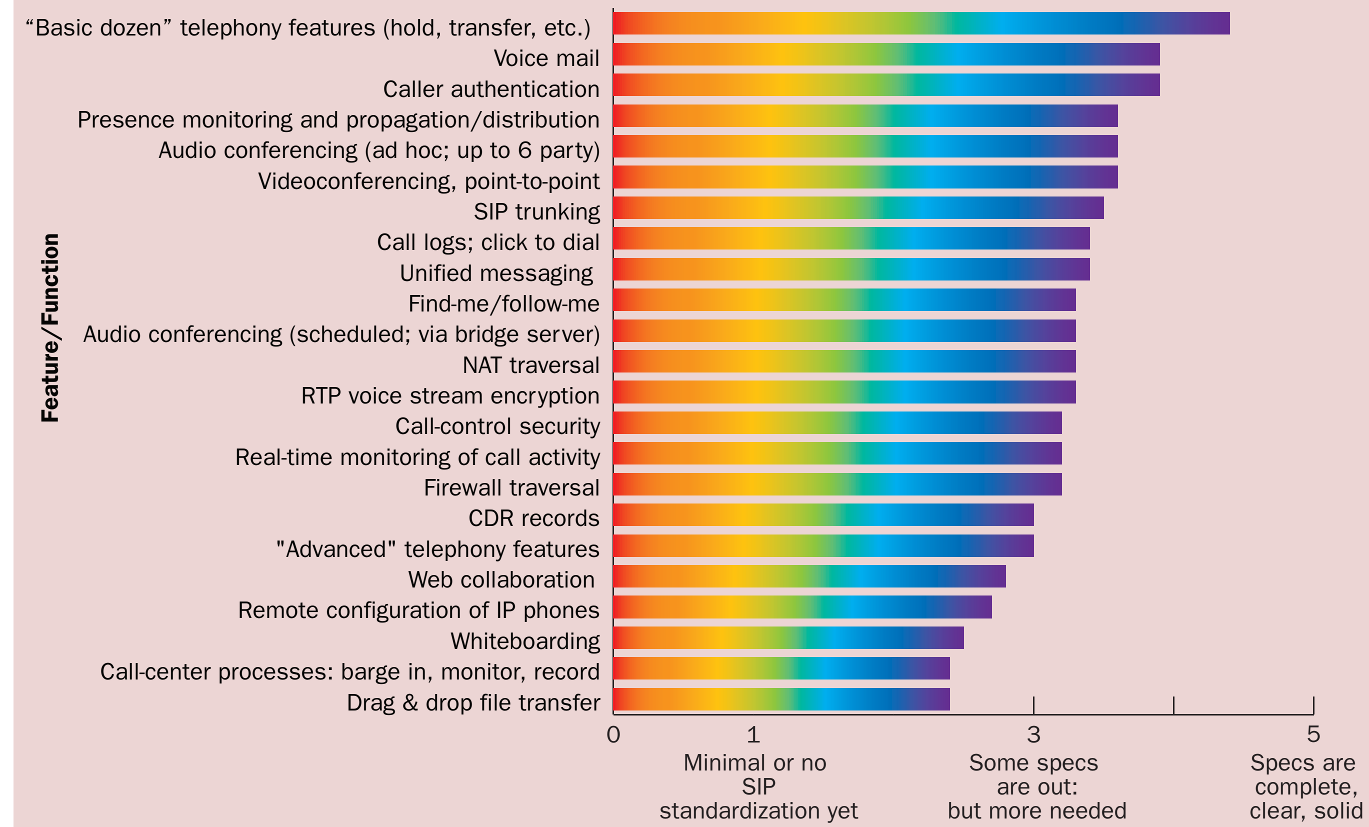
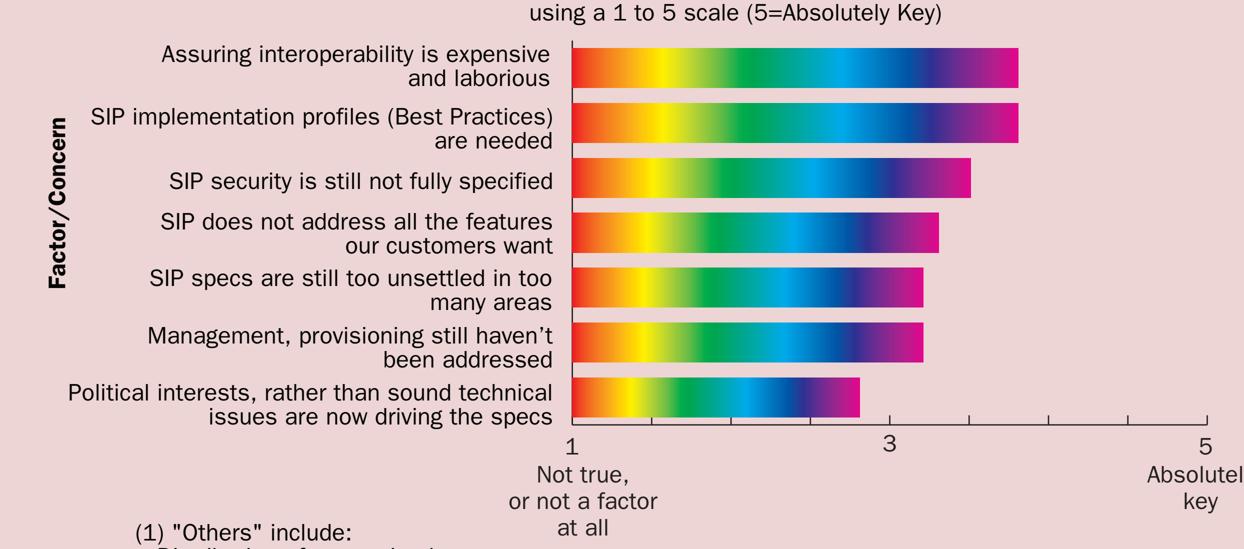



**FIGURE 2 Problems, Challenges Facing SIP—2007**

"How important, relatively speaking, are these factors to impeding SIP proliferation, using a 1 to 5 scale (5=Absolutely Key)




(1) "Others" include:
- Distribution of encryption keys;
- SBCs hurt end-to-end security;
- Inadequate SIP support in firewalls; and
- Lack of professional services for SIP migration

**Interoperability testing between vendors will remain an issue**

NATs and firewalls has been a particularly thorny bugaboo. But he said that a protocol called ICE (Interactive Connectivity Establishment) now represents the best prospect for improvement in this area.

■ In the area of SIP security, the respondents rated the adequacy of specs considerably lower than they did last year. The current 3.3 rating for RTP voice stream encryption represents more than a 10 percent drop over last year. The same is true for "call-control security."

Based on many comments received about this, it seems customer demand for better security in SIP products is increasing. And at the same time, vendors are finding issues in making SIP security work between different vendors' SIP products. The biggest issue, it seems, is agreement on how to exchange encryption keys.

"For secure RTP," said Batson, "key exchange is a mess." The SIP Forum hosts the popular SIPit events, at which vendors privately work out SIP implementation and interoperability issues. And according to Batson, attempts by participating vendors to meet and work out SIP security issues have increased dramatically.

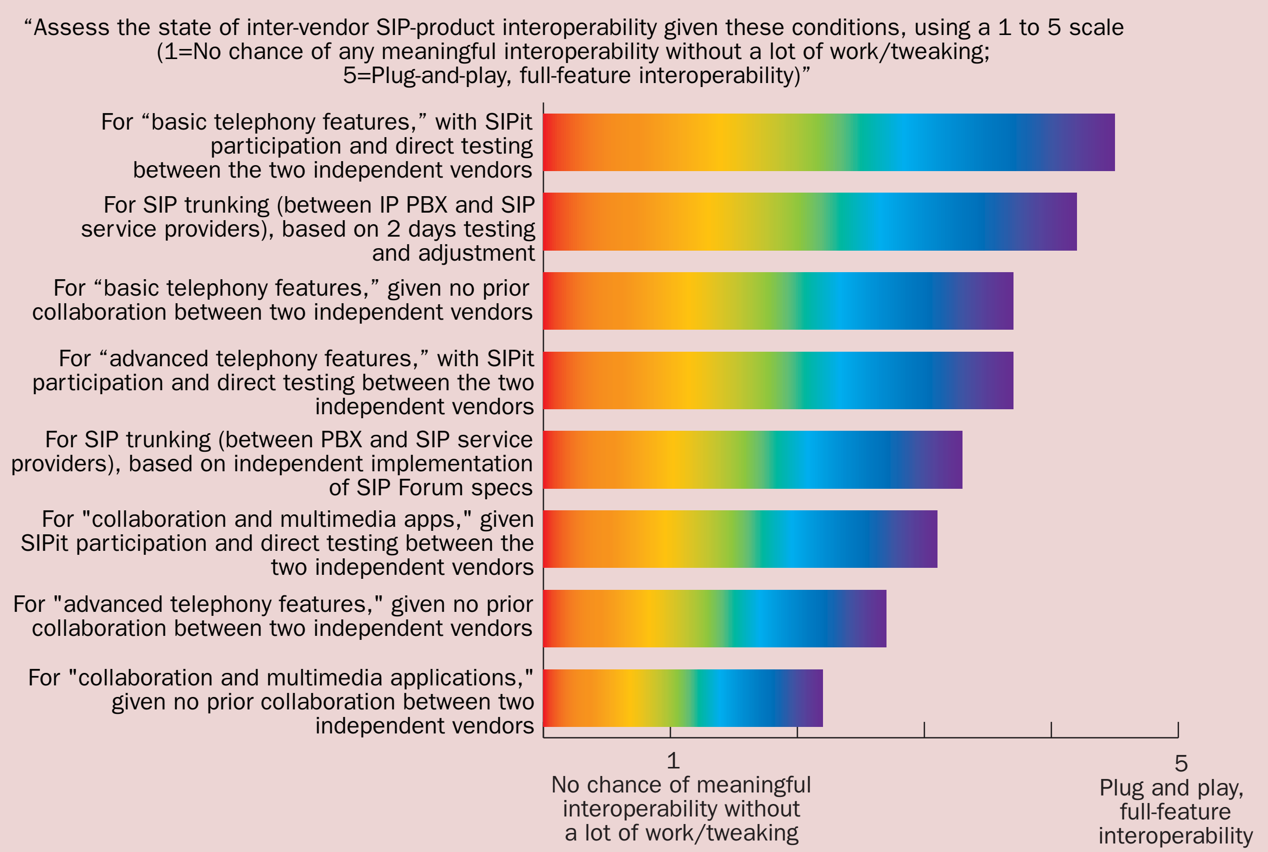
Batson said that as many as half the SIP products brought to SIPit events nowadays include Transport Layer Security (TLS). TLS is the accepted mechanism for ensuring that SIP call-control information is secure and encrypted. Key exchange for this encryption had been a problem, but it now seems that Public Key Encryption (PKE) is gaining ground as the least common denominator for TLS key exchange between vendors, until something better comes along.

Interoperable or secure Real-time Transport Protocol, (sRTP) isn't as far along. Batson estimated that 25 percent of SIP vendors that show up at SIPit events have implemented sRTP, adopting their own key-exchange protocol for their products internally. But again, with no industry-wide specs or consensus for doing sRTP encryption-key exchange, interoperable sRTP—the encryption of VOIP media streams—among different SIP vendors has not progressed.

Improvement was indicated in one other area of SIP specs, "remote configuration of SIP phones." The adequacy of SIP specs for this jumped 10 percent over last year. Some respondents commented that this is due to a bold new specification issued within the SIPPING working group of the IETF.

"The configuration framework work in the SIPPING working group is nearing completion," said Jason Fischl, chief technology officer of CounterPath Solutions. "The core draft is still being reworked for readability, but in general the mechanism is sound and is starting to be adopted by vendors. The actual format of the profile

**FIGURE 3 SIP Product Interoperability—2007**




## 3Com: SIP End-to-End

At 3Com, which pioneered industrial-strength SIP with its VCX IP-telephony system, SIP is now underlying multi-vendor and open-source application integration and several bold new initiatives.

"SIP is mature enough that we are now seeing actual, multi-vendor deployments," said Pat Rudolph, vice president of technology at 3Com. Indeed, 3Com credits the standards-based openness of VCX as the reason IBM last year tapped the package as a new application, delivering comprehensive IP-telephony to IBM's System i users.

In 3Com's case, SIP has become the universal glue for holding together applications, VOIP, multimedia and effective end-to-end management. True interoperability—via SIP—was apparent in various 3Com configurations we recently reviewed.

■ **End to end**—3Com's SIP-based product portfolio is unique in several ways. First, virtually all aspects of its IP-telephony, and the applications that run over it, are solidly SIP based. From unified messaging and conferencing services, to contact center, to attendant console, to trunking, it's all done via SIP—even the system management, which delivers end-to-end stats using the new extended reports (XR) capability of SIP's RTCP.

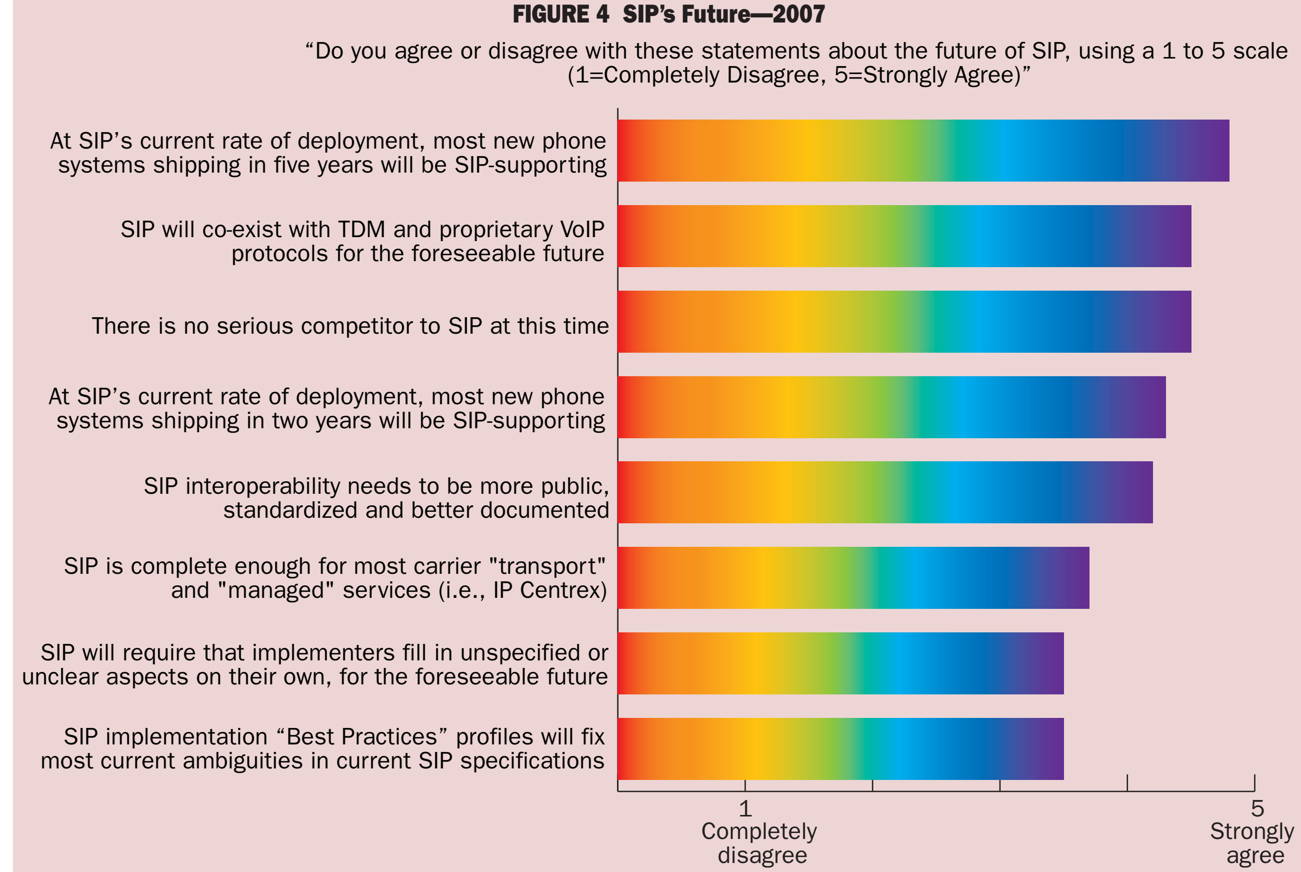
■ **NBX and VCX**—The NBX, 3Com's popular SMB-class IP-PBX, can now network cleanly as a subsystem in a broader network with 3Com's higher-end, native SIP-based VCX. The NBX call controller continues to run legacy VOIP call control internally, for backward compatibility, while at the same time interworking via SIP with one or more VCXs. In addition, the NBX and VCX can now both access and share the same SIP-based unified messaging and large-scale conferencing services, as well as gateways.

■ **Third-party plug-and play**—More and more third-party components are appearing in 3Com IP-telephony networks, thanks to SIP. In one running VCX network we noted Polycom SIP phones, Nokia WiFi SIP phones, even a MultiTech MultiVOIP gateway. Interestingly, the SIP-based MultiTech gateway was adapted to serve as a back-up, local branch call controller, able to handle SIP call processing if IP connectivity to a remote, centralized VCX call controller were lost.

What's next for SIP at 3Com? For one, a new program of interoperability and integration testing has just kicked off, the 3Com Open Network Program, which includes self-validation by third-party SIP vendors. And a new 3Com initiative, dubbed Open Services Networking, aims to spawn various third-party and open-source software modules, which integrate via common interfaces and APIs, including SIP. These will address various network-oriented chores, ranging from intrusion prevention to setting QOS in routers and switches.

3Com offers a new software development kit to facilitate this, and recently announced the first round of third-party applications based on it. These will run on generic Linux processor modules, which plug into 3Com's multislot routers and switches □

**FIGURE 4 SIP's Future—2007**



**Vendors offer different levels of interoperability assurances, depending on the partner**

datasets themselves is still far from being standardized and will be much more contentious. At this point, the IETF is focused on the mechanism to retrieve the [configuration] profiles, rather than the format of the profiles themselves."

Figure 2 shows SIP implementers' current thinking about SIP's weaknesses and issues that could threaten SIP's long-term prospects.

Emerging at the top of the list this year is the statement that "Assuring [SIP product] interoperability [between vendors] is expensive and laborious." This was rated 3.7 in overall significance, about the same as last year.

Interoperability testing between different vendors' SIP products seems destined to remain an issue, and an increasing administrative burden on vendors, as more and more vendors join the SIP-product community and as the complexity and functional diversity of SIP increases.

Indeed, a multilevel structure for third-party SIP-product interoperability is emerging at several major SIP-supporting vendors, including Cisco and Avaya. At the top level are "partners," where the vendors' products complement each other and there is usually a contractual joint-marketing agreement.

At the second level are "certified interoperable" products, where comprehensive lab testing is done, by one or both parties, and both vendors attest to, and stand behind, their products' ability to interoperate with each other on a fairly complete basis.

And at the lowest level is "basic function" interoperability, which one vendor usually undertakes on its own, or perhaps using an online, limited-function test suite implementation, which some vendors offer to any third parties. The larger vendor usually takes no responsibility, and provides no support or certification, for the interop claims of the smaller vendor.

One other result shown in Figure 2 is a bit disconcerting. Two criteria have grown in significance by at least 10 percent over last year. One is the statement that, "SIP does not address all the features our customers want," now a 3.2 rating. The other is that "SIP specs are still too unsettled in too many areas," rising to 3.1. These changes could reflect the reality of implementing SIP, especially by newcomers. Or it could mean growing disenchantment with SIP, acknowledging that it is not all things to all people. In either case, this sentiment will bear close watching.

Figure 3 reflects this year's assessment of relative SIP-product interop scenarios. The results are very close to last year's—in most cases statistically the same.

One area, though—the level of interoperability confidence between IP-PBXs and carrier services via SIP trunks—has climbed somewhat over last year. This may be the result of a detailed SIP Forum implementation spec for SIP trunks released last year, as well as much more actual hands-on experience gained by SIP equipment vendors, working out interop details with SIP-based carriers and service providers.

**About 35 or 40 features are solidified in SIP RFCs**

Figure 4 reflects the same generally optimistic outlook that we saw in last year's results. Two minor departures from last year have already been noted:

- The respondents have pared back expectations that "most phone systems" will be SIP-based in two years. Agreement with this statement dropped from 4.6 last year to 4.3. Again, expectations for SIP to take over within five years was at an all-time high—with near unanimous agreement at 4.8 out of 5.
- The adequacy rating of SIP for carrier and service-provider transport and "managed services," such as IP Centrex, has dropped off somewhat. This could just be that initial, very high expectations for SIP in the carrier space have been tempered somewhat with the reality of actual deployment. (The survey was not sent to SIP service providers, just to equipment vendors.)

### SIP Products Side-by-Side

Another part of our survey asked for comparative details about the respondent's SIP-based product(s)—including specifics about claimed interoperability with other vendors' products. We distilled and compared key SIP aspects of their responses in four areas—IP-PBXs and call controllers, SIP-based gateways, SBCs and SIP-security packages, and SIP-based application servers and suites.

Table 1 profiles 14 IP-PBXs from among the responses. Not quite half of these are native, SIP-only systems, while the rest have adopted SIP in a multistack role, usually alongside the vendor's legacy, and proprietary, call-control protocol.

What is especially revealing is the number and breakdown of features implemented within SIP by these IP-PBX vendors. In cases where more than 200 features are cited, those vendors—including Mitel and Nortel—readily acknowledge that these are implemented in a proprietary manner.

Such proprietary methods—employing feature codes or proprietary headers—are legitimate and authorized within SIP. Indeed, a basic tenet of SIP is that a device receiving a SIP message needs only to accept and process feature instructions it recognizes and understands. Any instructions not recognized can simply be ignored.

This means that SIP devices and endpoints can speak and understand several dialects of SIP features. There is the set of SIP features that are standard and which most all SIP devices recognize and support—shown in the chart as "per SIP RFCs."

And there are features defined in Internet Draft specifications. These are up-and-coming standards, but are still in development. Some vendors implement these, some do not. And implementations by different vendors won't necessarily interoperate. These are shown in the chart as "per SIP drafts." Most vendors interviewed for this article acknowledge that features implemented per draft specs will likely need to be reworked

**TABLE 1 Comparison Of SIP-supporting IP PBXs (1)— 2007**

| Vendor/IP-PBX | SIP Role/Other Protocol Support | Number Of SIP Features | Per SIP RFCs | Per SIP Drafts | Proprietary | SIP Security (2) | Third-party SIP Endpoint Support | SIP Trunking | Carrier Interop Via SIP Trunks |
|---|---|---|---|---|---|---|---|---|---|
| 3Com/VCX | Native SIP only | 175 | 55% | 42% | 3% | Authent | 12 vendors | Yes | 1 cited |
| Adtran/NetVanta | Native SIP, and TDM (T1, PRI) | 50 | 50% | 40% | 10% | Authent, IPsec VPNs | 1 vendor | No | — |
| Alcatel-Lucent/OmniPCX | Multistack/H.323, Q.sig, others | 16 | 100% | 0 | 0 | Authent, IPsec, some encryption | 4 vendors | Yes | 17 cited |
| Avaya/SIP Enablem't Services | Multistack/H.323, Q.sig, others | 60 | 10% | 0 | 90% | Authent, TLS | 17 vendors | Yes | 5 cited |
| BroadSoft/BroadWorks | Native SIP only | 321 | 100% | 0 | 0 | Authent | 15 vendors | Yes | 7 cited |
| Centile/IntraSwitch | Multistack/MGCP, Cisco prop (SCCP) | 85 | 76% | 12% | 12% | Authent | 10 vendors | Yes | 1 cited |
| Cisco/CallManager | Multistack/MGCP, H.323, Cisco prop | 176 | 50% | 10% | 40% | Authent, TLS, sRTP | None specified | Yes | — |
| Escaux/net.PBX | Multistack/H.323, Asterisk prop (IAX) | 50 | 30% | 20% | 50% | Authent | 10 vendors | Yes | 2 cited |
| Mera Systems | Multistack/H.323 | 10 | 100% | 0 | 0 | Authent | 3 vendors | No | — |
| Mitel/3300 ICP | Multistack/Mitel prop (MiNet) | 300+ | 3% | 0 | 97% | Authent | 9 vendors | Yes | 5 cited |
| Nortel/MCS 5100 & CS 1000 | Multistack/H.323, Nortel prop (Unistim) | 450+ | 10% | 30% | 60% | TLS, sRTP | 9 vendors | Yes | 2 cited |
| pbxnsip/IP-PBX | Native SIP only | 30 | 90% | 10% | 0 | Authent, TLS, sRTP | 10 vendors | Yes | 3 cited |
| Pingtel/SIPxchange | Native SIP only | 174 | 90% | 10% | 0 | Authent | 7 vendors | Yes | 5 cited |
| Siemens/HiPath 8000 | Multistack/MGCP | 100 | 40% | 45% | 15% | Authent, TLS | 8 vendors | Yes | — |

(1) Based on responses received to a survey emailed in December 2006 to all SIP-supporting vendors known to us. Some vendors who may offer applicable products did not respond to the survey and subsequently are not represented here.
(2) TLS = Transport Layer Security (encrypted SIP call control); sRTP = secure Real-time Transport Protocol (encrypted voice streams). Authentication is typically via Digest Authentication and MD5 (Message Digest 5) algorithm, protects call-control message integrity and verifies source identity.

when the specs are finalized as RFCs.

"If there's an RFC tomorrow," said Venky Raman, product manager of 3Com's VCX IP-telephony system, "we all have to change to it." For updating the SIP code in 3Com's products, Raman said the company is committed to staying as current as possible: "If we don't do it in the next software release, it'll definitely be in the one after."

In interviews, we asked vendors: How many SIP-defined features are there, really? And it turns out there is some agreement. Most agree that, per solid SIP RFCs, there are about 35 to 40 features defined today. And these can be expected to work between different vendors' products at a 90-percent confidence level (that is, 90 percent of these features will work, to a 90-percent extent, 90 percent of the time). That's not bad, practically speaking.

How many total, useful features are defined in the myriad SIP specifications today, putting aside interoperability prospects? That number seems to be between about 125 and 175. Most of these are derived from drafts, and so they are not likely to be interoperable, or even necessarily implemented, across different vendors' products. 3Com, Cisco and Pingtel, for example, all agree with this feature headcount.

Table 1 also shows the security mechanisms

**TABLE 2 Comparison Of SIP Gateways (1)—2007**

| Vendor/Gateway | Connects SIP Environment With: | Number 'Telephony' Features Supported | Interoperable IP PBXs, Soft Switches, Call Controllers | SIP Security (2) | Max Gateway Capacity (Concurrent Calls) |
|---|---|---|---|---|---|
| 3Com/VCX VOIP Gateways | PSTN (T1, E1, FXO) | 106 | Generic; none specified | TLS, sRTP | 480 (16 x E1) |
| Adtran/Total Access 900 Series | PSTN, various services | 10 | 8 systems cited | IPsec VPNs (for signaling & media) | 100 |
| Cantata/IMG 1010 | PSTN (SS7, PRI), other VOIP (peering, transcoding) | 12+ | 10 systems cited | RADIUS authentication and accounting | 96 to 768 |
| Cisco/IOS ver 12.4(11)T | PSTN | Not specified | Cisco and other unspecified | TLS, and SBC functions | 450 |
| Escaux/net.PBX | PSTN | Not specified | 3 systems cited | MD5 (Message Digest for authent) | 120 |
| FirstHand/Enterprise Mobility Solution | Cellular networks, PDAs and WiFi | 15 | 3 systems cited | TLS, sRTP, 3DES (BlackBerry) | 1,000+ (via signaling gway) |
| Grandstream/GXW Series | PSTN and analog (FXO and FXS) | 24 | Generic; none specified | sRTP | 8 |
| Mitel/3300 ICP | PSTN (TDM) | Not specified | 18 systems cited | — | 120 (4 x E1) |
| Multi-Tech/MultiVOIP (MVP) | PSTN (analog or digital) | 20 | 3 systems cited | None | 30 |
| Nortel/MG 1000, 3200 and 9000 | PSTN, legacy systems, SIP trunks | Depends, up to 450+ | 9 systems cited | TLS, sRTP | 192 (MG 1000) to 3,000 (MG 9000) |
| Siemens/RG8700 | PSTN (T1, E1) | 100+ (less in survivability mode) | Testing now with third party systems | TLS | 2,000 (model RG8716) |

(1) Based on responses received to a survey emailed in December 2006 to all SIP-supporting vendors known to us. Shown here are products described as "generic SIP gateways" (not interop-limited to the vendor or specific IP-PBX or SIP switching system). Some vendors who we know offer applicable products did not respond to the survey and subsequently are not represented here.
(2) TLS = Transport-Layer Security (encrypted SIP call control); sRTP = Secure Real-time Transport Protocol (encrypted voice streams)

**TABLE 3 Comparison of SBCs/SIP Security Packages (1) —2007**

| Vendor/SBC, Security Product | Package; Main Application | Firewall/ NAT Traversal | Protocol Conversion | Hides Internal Topology | Denial of Service Protect'n | Caller Verification | Other Key Firewall Protections, Security Features (2) |
|---|---|---|---|---|---|---|---|
| 3Com/IP Telecommuting Module | Appliance; designed to work with vendor's IP-PBX, SIP environment | Far-end traversal | | ✓ | ✓ | ✓ | Up to 600 SIP sessions |
| Alcatel-Lucent/VPN Firewall | Appliance; mediates between carriers and/or enterprises | Near-end traversal | | ✓ | ✓ | | IPsec VPN services; bandwidth management; intrusion detection; SIP-specific filters; tunneling options; full data firewall |
| Cisco/Multiservice IP-to-IP Gateway | Appliance and/or software; for enterprise-to-carrier | Near-end traversal | SIP-H.323 and transcoding | ✓ | ✓ | ✓ | TLS (signaling encryption) proxy; distributed DoS protection, QOS control, call admission control (CAC) |
| Covergence/Eclipse | Appliance; mediates between carriers and/or enterprises | Near- and far-end traversal | SIP-H.323 and transcoding | ✓ | ✓ | ✓ | Stateful firewall for VPN traffic; sRTP media encryption; bandwidth management, virus scanning |
| Data Connection/DC-SBC | Portable software | Near- and far-end traversal | SIP-H.323, transcoding | ✓ | ✓ | ✓ | Encryption/authentication; routing policies, bandwidth management |
| Ingate/Firewall and SIParator | Appliance; mediates between an enterprise and carrier service | Near- and far-end traversal | | ✓ | ✓ | ✓ | Full data firewall, spoofing protection, QOS and VOIP survival, applies sRTP and TLS, SIP trunking |
| Intertex/IX67 and IX68 | Appliance; mediates between SIP domains | Near-end traversal | | ✓ | ✓ | | Full data firewall, built-in SIP proxy server and registrar |
| Mera Systems/MVTS II | Appliance or software; mediates between carriers or enterprises | Near- and far-end traversal | SIP-H.323 and transcoding | ✓ | ✓ | ✓ | |
| Nortel/MCS 5100 Border Control Point | Software; designed to work with vendor's IP-telephony environment | Near- and far-end traversal | SIP-H.323 and transcoding | ✓ | ✓ | ✓ | Media anchor for address translation; BCP 7200 also offered, for mediation between any SIP domains |
| Sipera/IPCS | Appliance; mediates between an enterprise and carrier service | Near- and far-end traversal | | ✓ | ✓ | ✓ | SIP signaling and media firewalls, call routing policies, call admission control (CAC), VOIP VPN service |

(1) Based on responses received to a survey emailed in December 2006 to all SIP-supporting vendors known to us. Other vendors who may offer applicable products did not respond to the survey and subsequently are not represented here.
(2) Additional security features that were cited by the vendors in their survey responses; these have *not* been independently tested or reviewed.

that SIP-supporting IP-PBX vendors say they implement. Authentication, which assures the integrity of SIP control messages as well as the sender's identity, is now implemented universally, the responses indicate. TLS, providing encrypted call control, is growing in deployment popularity, and some sRTP—voice-stream encryption—is appearing in products.

We asked, too, about third-party endpoint interoperability and SIP-trunk support. As Table 1 shows, some vendors are much more aggressively pursuing third-party endpoint interoperability than others. Similarly, SIP trunking interoperability is being worked out between IP-PBX vendors and a growing number of carrier services.

Table 2 shows 11 vendor respondents, who offer "generic SIP gateways." In most cases these provide access to the PSTN from SIP environments, although they look to the SIP network as endpoints in themselves.

Because the PSTN posesses a smaller feature set than SIP-based networks, the number of telephony features that are supported by the gateways in Table 2. is less than in all-IP environments. Features in the PSTN world are reduced to what can be conveyed by the D channel of ISDN PRI links and Q.Sig-based signaling.

Showing that SIP circuitry, in terms of digital signal processing (DSP) density is making impressive strides, several of the gateways in Table 2 can support hundreds, even thousands, of concurrent SIP calls out to the PSTN.

Table 3 shows 10 security offerings designed to mediate call control and traffic between SIP domains, such as an enterprise and a carrier's SIP-based VOIP service. Most of these are security-hardened, packaged appliances, though a few are software only. A couple are oriented toward carriers and service providers, but most say their products address both enterprise and carrier environments.

Session border controllers, or SBCs, are emerging as a key—almost necessary—component of SIP networks today. Their key functions and responsibilities, summarized in Table 3, include: NAT/firewall traversal, protection from denial-of-service attacks and caller verification. We note, too, that most of these vendors claim their SBCs are equipped to support the "lawful interception" of SIP traffic, as required in such government edicts as CALEA.

Finally, Table 4 profiles 18 packages that are designed to exploit the inherent capabilities of a SIP-based infrastructure.

The chart shows varied support for a dozen or so features common to SIP-based applications suites today—from videoconferencing to presence to click-to-call directories and call logs. But these are not the only ones. Instant messaging support, for example, is usually also a component part of SIP-based application packages.

Another key criterion that enterprises will want to understand is the capacity to integrate desktops, via these packages, with new and upcoming Microsoft environments, which are increasingly SIP based. Integration with Microsoft Outlook, as far as access to contacts and unified messaging, is critical. It is clear that IP-PBX vendors are working hard to network their packages, via SIP, with Microsoft Live Communications Server (LCS), LiveMeeting and other, upcoming Microsoft server environments.

**Conclusion**

A detailed survey of SIP-supporting vendors yielded 36 complete responses, representing an estimated one-third of the estimated 90 to 100 vendors of SIP-supporting products today.

As we saw in last year's survey, SIP continues to spread, permeating more and more vendors and products. But as with last year, the responses indicate issues in the adequacy and completeness of the current SIP standards and specifications. Also, interoperability for advanced SIP features and applications remains problematic.

Enterprises should make sure SIP vendors that are expected to interoperate have worked out interoperability issues between themselves. This may entail a written "pre-tested" certification from one or both vendors, which details the extent to which interoperability has been checked□

**TABLE 4 Comparison Of SIP Application Suites, Servers (1)—2007**

| Vendor/application server, suite | Product package; main application | Softphone | Audio Conferencing | Video Conferencing | Web Collaboration | Presence | Buddy Lists (2) | Click To Call (3) | Desktop Sharing | Whiteboard | Web publishing (4) | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 3Com/Various packages | Linux software, available on IBM System i | X | X | X | X | X | X | X | X | X | X | Unified messaging; fat PC clients; includes contact center |
| Alcatel-Lucent/OmniTouch Unified | Linux or Windows software | X | X | X | X | X | X | X | X | | X | Fat and thin PC clients; offered in four discrete software modules |
| Avaya/Meeting Exchange | Conferencing server and Modular Messaging | | X | X | X | X | | | X | X | X | Modular Messaging does unified messag'g; mostly thin client access |
| BroadSoft/BroadWorks | Software for carriers, runs on Sun Solaris, IBM Linux | X | X | X | | | | X | | | | Fat and thin PC client access; mobile and residential applications |
| Centile/IntraSwitch | Linux package for IBM or HP servers | X | X | X | | X | X | X | | | | For enterprises or services providers; fat and thin PC client access |
| Cisco/Various packages | 10+ discrete packages on Linux or Windows servers | X | X | X | X | X | X | X | X | X | X | Fat or thin PC client access, depending on package |
| Continuous Computing/Trillium SIP Server | Carrier-oriented software, runs on various servers | X | X | X | X | X | X | X | X | X | X | Fat and thin PC client access |
| Escaux/net.PBX suite | Linux software and server packages | | X | | | X | X | X | | | | Fat and thin PC client access, depending on application |
| FirstHand/Enterprise Multimedia Solution | Linux software, offered to SIP-based PBX vendors | X | X | X | X | X | X | X | X | | | Mostly fat PC client access |
| Genesys/Meeting Center | Web-browser-based multimedia server package | X | X | X | X | | X | X | X | X | X | Thin (Web) client access, with IP hard or softphone |
| Genesys Labs/SIP Server | Windows and Linux software, for contact centers | | | | | X | | X | X | | | For managing customer interactions across a SIP infrastructure |
| IP Unity/Mereon | Linux package on custom Solaris server | | X | X | X | | X | X | X | X | X | Fat and thin PC client access, depending on application |
| IVR/Talking SIP | Carrier-oriented software, runs on Windows servers | | | | | | | X | | | | Web callback, re-origination, reminder/notification services |
| Mitel/NuPoint Messenger and QuickConference | Two Linux software packages | | X | | | | | | | | | Thin PC client access; unified messaging via NuPoint Messenger |
| Personeta/TappS Applications | Carrier-oriented software, | | X | X | | X | X | | | | | Thin PC client access |
| Pingtel/SIPxchange Servers | Three server applications, run on generic PCs | | | | | X | | | | | | ACD Server, Media Server and Presence Server; thin client access |
| Nortel/MCS 5100 and others | Various SIP-enabled server packages | X | X | X | X | X | X | X | X | X | X | Contact Center and MPS servers included; fat and thin client access |
| Siemens/Open-Scape, Expressions | Windows-based software for enterprises | | X | X | X | X | X | X | X | X | X | Features based on inclusion of Microsoft LCS '05 and LiveMeeting |

(1) Based on responses received to a survey emailed in December 2006 to all SIP-supporting vendors known to us. Other vendors who may offer applicable products did not respond to the survey and subsequently are not represented here. Features shown here were cited by the vendors in their survey responses; these have not been independently tested or verified.
(2) Indicates support for "buddy lists, workgroups, and custom contact listings."
(3) Indicates support for "click-to-call directory and/or call logs."
(4) Indicates support for Web/desktop publishing; that is, one-way presentations.