BCR BEST-IN-TEST

Which Large IP-PBX Rules?

Edwin E. Mier, David C. Mier and Robert B. Tarpley

IP telephony's big guns came loaded for bear in this year's IP PBX shoot-out. The hottest areas: mobility, applications and security.

Ed Mier is founder of Miercom, a network consultancy and product test center based in East Windsor, NJ. Dave Mier is manager of lab testing and Robert Tarpley is senior lab tester at Miercom.

January 2005….And IP telephony is rolling out at a rate of 25,000 to 30,000 IP endpoints per month, the market researchers say. Indeed, if you're not already involved in selecting the vendor for a high-end IP-PBX deployment, the chances mount that sometime this year, you will be.

You likely won't make your pick based solely on a hands-on comparative article like this one. But the odds are strong that your list of choices *will* include one or more of the five vendors reviewed in this, the latest *BCR* Best-in-Test review of large IP-PBXs.

As the latest in what has become a major industry event, Miercom last fall started testing for the sixth annual comparative review of IP-PBXs. Today's article features the results in the "high-end" IP-PBX class—defined as systems targeted at customers requiring more than 1,000 stations. Next month's issue highlights the test results for the "mid-range" class—IP-PBXs supporting typical deployments of 100–999 stations. The *BCR* April issue will conclude this year's testing with the results of the "low-end" IP-PBX review—systems typically supporting fewer than 100 IP stations.

Invitations were issued late last summer to all IP-PBX vendors known to us. In the high-end class, five vendors accepted this year's challenge and were tested:

TABLE 1: Scorecard: Large IP PBXs (1,000+ Stations)

	Max Points Possible	Alcatel OmniPCX Enterprise	BCR Best in Test Avaya S8710 Media Server, G650 Gateway	Cisco IP Communications System	ShoreTel ShoreTel5	Siemens HiPath 4000
Architecture (1)	15	12	14	12	10 1/2	11
Endpoints (2)	20	19	20	19 1/2	17 1/2	18
Management and Admin (3)	15	12 1/2	11	12	11 1/2	11 1/2
Features (4)	20	15	17 1/2	17 1/2	17	16 1/2
Security (5)	10	5	8 1/2	9	5	5 1/2
Performance (6)	20	20	19	17	16 1/2	13
Bottom Line	100	**83 1/2**	**90**	**87**	**78**	**75 1/2**

(1) Includes: Call-control survivability; redundancy and failover; IP-to-PSTN re-routing, VOIP bandwidth control; QOS support; scalability, standards support and interoperability; and distributed-system networking.

(2) Includes verification of endpoint support: IP hard phone, softphone, wireless, analog/fax support, Power-over-Ethernet; unique, specialty and advanced endpoint features, including video.

(3) Based on management task completion, interface navigability and intuitiveness, on-screen help, real-time monitoring, reports and reporting.

(4) Includes: Validation of 16 basic phone features; verification of common add-on subsystems (E911, voice mail, voice recognition/response, etc.), and evaluation of special and unique "advanced" features.

(5) Includes: Verification of encrypted call control, VOIP/RTP stream encryption, secure management access, vendor's security documentation, services, security-infrastructure offerings and affiliations.

(6) Includes: Traffic load/call-completion testing, interactive/MOS voice-quality tests, latency measurements, verification of high-availability features.

■ **Alcatel,** which submitted its OmniPCX Enterprise system.
■ **Avaya,** which sent a system featuring the S8710 Media Server and the G650 Media Gateway products.
■ **Cisco,** which provided its latest IP Communications System, featuring the MCS 7845H-based CallManager.
■ **ShoreTel,** which sent its modular ShoreTel5 system.
■ **Siemens,** which submitted its HiPath 4000.

Notably absent from the high-end players this year was Nortel Networks. Nortel was invited but respectfully declined, citing "lack of resources."

Updated Methodology

Miercom totally revamped its IP-PBX test methodology for this latest round, taking into account the suggestions and concerns of past participants. The new test plan was built on hard, objective task accomplishment, reducing the potential for testers' subjective grading. Procedures were also adopted to ensure consistency in portions of the review involving management, architecture and features.

As shown in the Scorecard (Table 1), congratulations go to Avaya, which emerged a clear winner in this year's high-end IP-PBX review. Running a beta version of its soon-to-be-released 3.0 system software, Avaya earned one of the highest scores so far achieved in Miercom's comparative IP-PBX testing, 90 out of a possible 100.

Right on Avaya's heels was archrival Cisco, which earned 87 points in a powerful showing of its latest CallManager version 4.1(2) system. Cisco was followed closely by third-place Alcatel, which fared very well with its OmniPCX Enterprise, earning 83 1/2 points overall. ShoreTel and Siemens rounded out this year's high-end IP-PBX quintet.

Having spent a long week in the lab with each system, the Miercom test team felt that a few of the high-end IP-PBXs deserve special noteworthy praise, which the point scores alone may not adequately reflect. So in addition to the Best-In-Test, three "best of" category awards are also announced as a result of this year's testing:

■ "Best Performing IP-PBX, Large Systems," is awarded to Alcatel e-ND for its OmniPCX Enterprise. With *all* performance metrics considered—latency, call-load handling, call and connection quality in all scenarios, survivability and fail-over times, and so on—Alcatel did exceedingly well across the board. And only Alcatel scored a perfect 20 for 20 in the Performance category.
■ "Easiest to Use IP-PBX, Large Systems," is awarded to ShoreTel for its ShoreTel5 system. Reflecting its relative ease of deployment, consolidated management and easy-to-use features and applications, ShoreTel deserves this tribute. You won't need nearly as much technical staff to deploy and run ShoreTel's IP-PBX as the others.
■ "Most Secure IP-PBX, Large Systems," is awarded to Cisco for its CallManager-based IP Communications System. Based on our security analysis in this year's testing, Cisco clearly has leveraged its unique position in LAN and WAN infrastructure and networking to offer as secure an IP telephony environment as your enterprise demands, and your budget allows.

The formal 100-point scale comprises six categories. Three of these offer a maximum of 20 points each: Endpoints, Features and Performance. Architecture and Management categories are each worth up to 15 points, and the Security category offers the last 10 points.

Well Architected

In the Architecture part of the review, all aspects of the IP-PBX's architecture were examined—including scalability, survivability, voice over IP (VOIP) capabilities and configurability, ranging from vocoders and bandwidth control to teleworker and road warrior support.

The testers set up and exercised each system's call-control redundancy and fail-over, as well as the ability to re-route around a failed VOIP-PSTN gateway and a failed IP-WAN link. The abilities to set quality of service (QOS) values for VOIP traffic and to tune VOIP performance parameters were also checked.

Other architectural issues were reviewed in separate whiteboard and discussion sessions, including how multiple distributed systems are networked, which protocols and standards are supported, maximum capacity and expandability.

Table 2 (pp. 26–27) provides a thumbnail comparison, including some key architectural aspects. The vendors supplied the capacities shown for maximum call load and IP stations supported; we did not verify these in the testing. We did conduct call-completion load tests, delivering 50,000 calls per hour to each system tested. Those results are discussed under the Performance section (and shown in Table 3, pp. 28).

Avaya ended up at the top in the Architecture category, earning 14 out of a possible 15 points. Among its distinguishing points were: the fastest call-control fail-over time, exposing just a 6-second interruption; user-settable IP quality of service (QOS) thresholds—for latency and packet loss—for use in re-routing VOIP traffic between IP and PSTN paths (Table 4, p. 29). Avaya also showed off its rich new SIP support—for third-party endpoints as well as SIP trunking between systems—via its new add-on Converged Communication Server, or CCS.

Why not a perfect score for Avaya? A point was deducted because we found room for improvement in Avaya's customer documentation on how to deploy road warrior and teleworker environments.

Cisco and Alcatel tied for second, each with 12 points. Cisco now claims the highest *single-*

Products with the best performance, ease of use and security also deserve mention

Third parties still must use the Skinny protocol to link their phones to Cisco's server

system IP-station capacity of the vendors tested, 30,000, based on a CallManager cluster with at least five servers. As noted in Table 2, the others ranged from 5,000 IP stations (Alcatel), to 8,000 (ShoreTel), up to 12,000 (Avaya and Siemens). The protocols, capacities and feature-transparency aspects of networking multiple, distributed systems by these vendors varies considerably.

Cisco offers somewhat eclectic VOIP-protocol support. While SIP trunks are supported, as well as MGCP and H.323-based control of gateways, Cisco's native VOIP call-control protocol is still its proprietary SCCP, called "Skinny." That is the protocol that third parties have to adopt in order for their endpoints to connect to Cisco VOIP networks.

Cisco offers as survivable an architecture as any of its competitors, but fail-over times take a little longer. And unlike Avaya, IP QOS is not dynamically monitored for maximizing VOIP call quality. Siemens is the only other high-end IP-PBX vendor, of those tested, that also supports active IP QOS monitoring (of latency and packet loss) as a criterion for re-routing VOIP calls.

Alcatel is in the process of adding SIP to its currently H.323-based VOIP control-protocol support, but SIP trunking wasn't yet supported in the version tested. The native call control is proprietary, though it works through H.323 This means that H.323 is the underlying basis for the call-control message exchange, but H.323 doesn't address most vendors' complete feature/functionality set, so above a certain minimal set, it is effectively proprietary (in its product, Avaya calls this "proprietary extensions"). Still, the OmniPCX has integral H.323 and SIP gateways (as well as an

TABLE 2 Large IP PBXs Tested

Vendor US HQ, city, state Website URL	Alcatel e-ND Calabasas, CA www.alcatel.com/enterprise	Avaya Inc. Basking Ridge, NJ www.avaya.com
IP PBX tested	Alcatel OmniPCX Enterprise	S8700 Media Server, G650 Media Gateway
Version tested, availability	Beta R6.1, to be released Feb 2005	Beta 3.0, to be released Feb 2005
Max system call load, per vendor	300,000 BHCA, as tested	300,000 BHCA, as tested
Max system* IP-station capacity	5,000 if all IP hard phones; 3,000 if all IP softphones	12,000, a mix of IP hard and softphones
Call control server, platform(s) tested	Communications Server: running on Linux on IBM x-Series 305/306 e-Servers	S8710 Media Server: running on Red Hat Linux 8.0 on vendor's processor platform
Other key system components	IP Media Gateway (IPMG): a multi-slot chassis for PSTN trunking, IP station control	G650 Media Gateway: a multislot chassis for PSTN trunking, IP station control
Max expandability: How multiple distributed IP-PBX systems are networked	Proprietary ABC protocol, over IP (within H.323) or TDM trunks, or Q.sig over TDM; up to 256 systems with 90% feature transparency	Proprietary DCS, or SIP or H.323 trunking over IP, or DCS or Q.sig over TDM; up to 1,000 systems; feature transparency varies w/protocol
Native VOIP call control protocol	H.323, v2; with proprietary feature extensions	H.323, v2; with proprietary feature extensions; gateway control can be H.248
Other VOIP protocol support	H.323 gatekeeper and gateway, SIP gateway and proxy server, are integral in Communications Server	H.323 gatekeeper and gateway are integral; H.248 is supported; SIP requires separate CCS SIP server
SIP trunking	Planned, under test	Yes
3rd party IP phone, device support	Via H.323 and SIP; 6 vendors/products cited	Via H.323 or SIP; 14 vendors/products cited mostly SIP)
Vocoders supported; VAD (silence suppression) support	G.711, G.729a, G.723.1; VAD supported on G.729a and G.723.1	G.711, G.729 and G.729a, G.722, G.723.1, G.726; VAD supported on G.711, G.729 and G.729a
Options for 802.3af in-line power (PoE) to IP phones	Vendor's PoE switches or PowerDsine mid-span power-insertion units	Vendor's PoE switches or PowerDsine mid-span power-insertion units
US List price per station based on 1,500 IP stations**	$586; includes mid-span 802.3af powering 1,000 IP hard phones	$680; includes mid-span 802.3af powering 1,000 IP hard phones

* Based on a single, discrete "system" where all phones and gateways are under a common call controller.
** Based on a conventional, large enterprise configuration: a system supporting 1,500 IP stations—consisting of 1,000 "mid-range" IP hard phones (each providing a secondary switch port, at least six feature buttons, and an LCD display) and 500 softphone licenses—and including voice mail for all, full management, and power to IP phones. "US list" prices are subject to further discount.

H.323 gatekeeper and a SIP proxy server). And with these, the system today supports a half-dozen third-party endpoints.

Alcatel's call-control fail-over time, 19 seconds, was just behind Avaya's. The 5,000 max-IP-station capacity of Alcatel's system could prove an architectural limitation, especially as more and more IP-telephony deployments are announced that far exceed 10,000 stations. Also, like Avaya, Alcatel could improve its "how-to" customer documentation for VOIP environments including road warriors and teleworking.

Siemens HiPath 4000 is essentially a multislot chassis, into which all manner of gateway, IP-station and other cards are plugged. The system was difficult to provision for IP—that is, to readily figure out how many cards of what type are needed for a particular configuration, or how to plan for redundancy or scalable growth. Also, during our testing, we encountered some problems due to different, and in some cases incompatible, versions of software and firmware in the different cards, and even in different IP phones.

Still, with enough engineering, the HiPath 4000 could be set-up properly, including for redundancy failover. Indeed, reflecting Siemens' European heritage, some survivability mechanisms rely on ISDN, such as a Basic Rate Interface for remote-branch back-up signaling. There is no current support for SIP; the system is H.323 based and there are no protocol gateways. Siemens cites just one compatible third-party endpoint, the SpectraLink 802.11b VOIP wireless phone, which Siemens resells.

All vendors tested in this round, except for ShoreTel, support similar redundancy, failover

Most of the vendors support similar redundancy, failover and backup schemes

Cisco Systems, Inc. **San Jose, CA** **www.cisco.com**	**ShoreTel, Inc.** **Sunnyvale, CA** **www.shoretel.com**	**Siemens Communications, Inc.** **Boca Raton, FL** **http://enterprise.usa.siemens.com**
Cisco IP Communications System	ShoreTel5	HiPath 4000
Version 4.1(2), shipped Oct 2004	ShoreTel5, Rel.1.2, shipped Oct 2004	Version 2.0, released June 2004
250,000 BHCA (125,000 as tested)	50,000 BHCA	112,500 BHCA
30,000 IP hard or softphones, with a 5+ server cluster	8,000 IP hard or softphones, with 2,000 trunk channels	12,000, a mix of IP hard and softphones
CallManager: running on Win 2000 Advanced Server on custom HP/Compaq servers	ShoreGear-120/24: VxWorks-based switch modules, and ShoreWare Director, running on Win 2003	Call Controller: running on Unix on vendor's processor card (Intel Pentium 3, 800 MHz)
Many Cisco switch and router platforms can be configured for PSTN trunking, IP station control	ShoreGear-T1, a single-T1, self-contained gateway module for PSTN trunking	HG 3530 gateway and other cards, in multislot chassis, for PSTN trunking, IP station control
H.323 w/ gatekeeper or SIP trunking over IP; or Q.sig tunneled in H.323; up to 100 clusters; most features do not extend across clusters	Maximum distributed configuration is a single system, spanning up to 200 sites, and supporting up to 10,000 stations and trunk channels	Proprietary CorNet protocol provides 100% feature transparency across distributed systems, up to 100,000 max total users supported
Proprietary 'Skinny' (SCCP) for phones; H.323 or MGCP for gateway control	Proprietary SIP-like for call control; MGCP to endpoints	H.323, for endpoints and for gateway control
Integral gateways for H.323, MGCP and SIP trunking. SIP phone support requires an external proxy server	MGCP is used for native and 3rd party endpoint control	H.323 gatekeeper and gateway are integral
Yes	No	No
Via proprietary SCCP; 7 vendors/products cited	Via MGCP; 3 vendors/products cited	Via H.323; 1 vendor/product cited
G.711, G.729, G.723.1, G.722, wideband, GSM; VAD supported on all vocoders	G.711, G.729a, G.726 (ADPCM), and broadband (128 and 256 kbps); VAD is not supported	G.711, G.729, G.723.1; VAD supported on all vocoders
Vendor's PoE switches and switch modules, or mid-span Power Patch Panel	Netgear PoE switches or PowerDsine mid-span power-insertion units	Vendor resells PoE switches and PowerDsine mid-span power-insertion units
$715; includes 10/100 switch ports for powering 1,000 IP hard phones	$601; includes 10/100 switch ports for powering 1,000 IP hard phones	$572; includes 1,000 AC adapters for powering IP hard phones

TABLE 3 Heavy-Load Call-Completion Rate

	Alcatel	Avaya	Cisco	ShoreTel	Siemens
Percent call completion, 50,000 BHCA x 9 hours	99.99+	100	99.99+	99.6+	99.7+
Call load included voice prompts	✓	✓		✓	
Call load included DTMF tones	✓	✓			
Points awarded (8 maximum)	8	8	6	3	3

Notes: All call-completion tests required that the IP-telephony system perform basic call routing, call completion and verification. Two additional components added to the load complexity: voice prompts and DTMF tones. To get extra points for these, each call had to convey recorded-voice prompts and/or DTMF tones in each direction to be considered successful. Handling one or both of these reflects the enhanced robustness of the IP-telephony system. The chart shows the call-completion rate, and whether voice prompts and/or DTMF tones were additionally handled as part of the call load.

and backup schemes to assure survivability of IP-telephony operations.

For example, all—except ShoreTel—support a hot-standby, redundant call controller, which mirrors or synchronizes the entire system database—call-states, status and configuration—in what is called an active-passive relationship. The vendors vary subtly in how much bandwidth is needed between these two nodes, and subsequently how far apart they can be situated.

Then there are back-up schemes for:

a.) assuring that branch offices that get cut-off from a remote call controller can continue to make calls, and

b.) an off-site call controller, that's more loosely coupled with the primary call controller, and which can take over call control for the whole system if the primary fails.

We exercised these fail-overs for all the systems and found that, while some operational details are different and costs vary, they worked reasonably well.

The exception to these failover models is ShoreTel, which offers essentially two types of small call-control modules (see Table 2). One is the ShoreGear-120/24, which handles up to 120 IP phones, or 24 directly-connected analog stations. This is a self-contained switch, and there will be a dozen or more of these if you need a system with 1,000-plus IP stations. The other unit is the ShoreGear-T1, which is a classic VOIP-to-T1 gateway.

The ShoreTel system is clean and refreshingly simple to deploy, but it is quite different from the other high-end IP-PBX competitors. ShoreTel's modules all constantly update each other, and back each other up in active-active relationships. It is an "n + 1" fail-over architecture, meaning that a single active unit, with sufficient excess capacity, can provide backup to any number (n) of other active units.

There's nothing inherently flawed with ShoreTel's architecture that we could tell. It's just different. We found in testing that ShoreTel provides the same degree of resiliency and survivability as its competitors, and we successfully conducted all the same fail-over tests. We note, however, that call control fail-over takes a little longer with ShoreTel than with the others, about 35 seconds. But any such unit that fails affects just the 100 or so IP stations registered to that particular, standalone switch module.

ShoreTel's call control is proprietary—which is true for all the systems tested to a large extent. ShoreTel has chosen to implement a fairly obscure VOIP standard, MGCP, for endpoint control, and does support several third-party endpoints via this protocol.

Endpoints

Twenty percent of this evaluation, 20 points out of 100, was based on Endpoints. Up to 7 points were awarded based on demonstrated support for: analog/fax, IP hard- and softphone, power-over-Ethernet (PoE), IP speakerphone and wireless IP phones or endpoint devices. All the vendors garnered full points here. The only issue was ShoreTel, which does not sell or resell wireless phones, but does support several models via plain-old analog connections. ShoreTel even supports SpectraLink wireless IP phones in this manner, using SpectraLink's VOIP-to-analog gateway unit.

Then up to 4 points were awarded based on a hands-on review of the vendor's high-end IP hard phone. Another 4 points were awarded based on a hands-on review of the vendor's softphone, or else its "best price performing" IP hard phone. Finally, up to 5 1/2 points were awarded based on the overall breadth and depth of the vendor's endpoint offerings, along with consideration of any unique or special endpoint features, including, for example, video capabilities.

Avaya offers nearly a dozen IP phone endpoints, including softphone and re-branded Polycom IP speakerphone. There are two models of IP wireless phone (office and warehouse), and several flavors of softphone (including attendant console and call center agent). Avaya's minimal 4601 IP phone lists for $139. The high-end, full-featured 4630SW is costly at $965 list, but features a very nice color touch screen, which can even display a keyboard for text input. There are copi-

TABLE 4 High-Availability Features Compared

	Alcatel	Avaya	Cisco	ShoreTel	Siemens
Call controller fail-over time; redundant set-up	19 seconds; local active-passive	6 seconds; local active-passive	30 seconds; local active-passive	23 seconds; local active-active	35 seconds; local active-passive
PSTN-gateway fail-over time	Immediate; no loss of service	Immediate; no loss of service	Immediate; no loss of service	Immediate; no loss of service	Immediate; no loss of service
IP-to-PSTN fail-over criteria (for automatic traffic re-routing)	• IP link loss; • IP bandwidth overflow	• IP link loss; •IP bandwidth overflow; • latency; • packet loss	• IP link loss; • IP bandwidth overflow	• IP link loss; • IP bandwidth overflow	• IP link loss; • IP bandwidth overflow; • latency; • packet loss
Fail-over time, (for IP link loss)	16 seconds	20 seconds	85 seconds	4.2 seconds	6 seconds
Auto-recovery when IP link restored	60 seconds	15 seconds	6 seconds	45 seconds	10 seconds
Other validated high-availability features	• Remote back-up call controller (Silent Call Server) • Hot-swap T1 modules on media gateway(s) • Signaling via back-up link for remote sites	• Remote back-up call controller (Enterprise Survivable Server), • Redundant IP-phone (CLAN) and gateway (MedPro) cards • Dual NICs on some key nodes	• Remote back-up call controller (Survivable Remote Site Telephony) • Hot-swap power supplies, dual NICs in call controller(s) • Redundant controllers in gateways (Catalysts)	• Users dynamically re-assigned if any switch fails • Redundancy options for servers • Dual NICs on gateways • Power-fail transfer analog port	• Remote back-up call controller (Survivability Processor) • IP line card fail-over • Redundant chassis power supply

ous softkey options via the touch screen. A small nit: only RJ-22 headsets are accepted; we think adding a USB port or sound-card jacks would be a plus.

Splashy or unique features? Avaya has a bunch. Point-to-point video is now supported as an adjunct to its affordable softphone package. There is the "click to dial" software feature, which lets you click on and dial any phone number that appears in Web content, and there are really well thought-out configuration options for the softphone (for teleworkers, road warriors, and so on). Mobility-wise, there's Avaya's new "dual mode" wireless phone: an 802.11a-based wireless phone for on-premises use, which is also a GSM-based cell phone for off-premises connectivity—in either scenario using a display that mimics the interface on Avaya desk sets.

Cisco earned a near-perfect score here, garnering 19 points. The vendor earned the full 4 points for its featured IP hard phone, the $695 (list) 7970G. There's a very nice, backlit color touch screen, easy access to call logs, as well as network statistics. Cisco submitted its 7940G for our consideration as a "Best Price Performing" IP hard phone. It's a very useable IP phone, with a small monochrome display, two switch ports, and a good speakerphone. But given the $315 price tag, we withheld 1/2 point from full credit for "value."

Cisco special endpoint features? We think the vendor's VT Advantage video package—$190 complete, for USB camera and all software—is likely to help make desktop video telephony ubiquitous. Displaying XML data on the IP hard phones is especially well supported. And then there's Cisco's new high-end 7970G-GE IP phone, featuring Gigabit Ethernet connections for PCs or laptops that connect to the LAN via the phone set's internal switch.

Alcatel earned 19 points for endpoints, just one point behind first-place Avaya. Alcatel showed off two of its new IP-Touch hard phones, with list prices at $495 and $695, augmenting its older e-Reflexes line of IP hard phones. Despite the name, the color screens of its new IP-Touch phones are not touch screens. The screens were not as clear and legible as competitors' and so lost 1/2 point for display quality. The vendor's softphone showed very well and earned Alcatel the full 4-point credit.

Some special Alcatel endpoint features were shown, such as slick support for Bluetooth wireless headsets on the IP-Touch hard phone, and integral keyboards with many of its phone models. Also, via integral gateways, third party SIP and H.323 endpoints can concurrently connect. The vendor offers no videoconferencing support, however.

Siemens showed off its new optiPoint 420 Advance phone, which it calls "self-labeling" because soft keys and line appearances show in LCD displays. That's what competitors have long done. What's remarkable is that many of Siemens' IP phone models still require paper labels. That aside, Siemens offers a half-dozen IP hard phone models, in a broad range of features and prices. Siemens phone displays are oriented towards Wireless Application Protocol (WAP) format and

Though management packages were generally good, all had their tedious, non-intuitive elements

feature "sidecar" add-on modules to expand buttons and line appearances. Even though this isn't a wireless environment, WAP is used because of its ability to modify Web and HTML data so it can be displayed on limited screens—like cell phones and many IP phones—which cannot handle full-screen Web pages and HTML data.

Siemens optiClient 130 softphone earned nearly full credit, and offers notably excellent on-screen help. We thought the softphone's latency was a bit high, however. The only video Siemens currently offers is as part of the company's OpenScape conferencing package. That was not included in the review, since OpenScape is a PC client application, totally independent of the HiPath 4000 IP-PBX.

ShoreTel offers three models of its ShorePhone, featuring lackluster monochrome displays, but one of the best-quality full-duplex speakerphones of the IP phones we have tested. The vendor also offers a Polycom IP speakerphone. A real competitive strength of ShoreTel is a rich CTI package, called Call Manager, which earned ShoreTel full 4-point softphone credit. The vendor says it expects customers to use the Call Manager software, which is tightly integrated with the ShoreTel phone system, rather than make much use of the phone set's display, and we agree. The Call Manager suite, which offers the softphone as an option, comes in various versions, and features an extremely easy-to-use interface for finding contacts, placing calls and scanning call logs. ShoreTel does not offer or support any desktop video capability.

Management And Administration

The Management and Administration portion of the evaluation offered vendors up to 15 points. There were two parts to this evaluation: First, a series of specific management tasks was posed—such as displaying and sorting a directory listing of phone-system users by name, extension and phone type. And points were awarded for whether or not the tasks could be satisfactorily accomplished.

Secondly, the IP-PBX management interface(s) of each vendor was reviewed, and comparatively assessed in six areas: configuration, including moves, adds and changes; on-screen help; real-time monitoring; navigability of the interface(s) and reporting.

Every vendor's management offered some unique or useful aspects, along with other aspects that were non-intuitive or tedious to use. Not surprisingly, then, the scores awarded in the management category were not far apart. At the top was Alcatel, which garnered 12 1/2 points out of 15. At the other end, just 1 1/2 points away, was Avaya, with 11 points.

Alcatel offers the OmniVista package for much of its system management. This is a set of mainly client/server Windows applications, plus a Configuration application that's directly accessed on the call controller. We also used Alcatel's command-line interface quite a bit, as well as another separate Windows application, called the "Alcatel VOIP Assessment Tool," and the late beta version of a new Web-based maintenance tool.

Strengths of Alcatel's management: Did well producing the management data we sought, but a lot needed to be retrieved using some fairly detailed commands on their CLI. Good job on configuration, on-line help and reports. Shortcomings: Too many interfaces, and management of softphones is not integrated.

Cisco's management was next best, close behind Alcatel's, earning 12 points out of 15. We used the native CallManager admin interface, but there are now more than a dozen additional plug-in, supplementary applications to augment it. It may be tough to keep track of these, but most of those we reviewed—like the Bulk Admin Tool, for quickly applying mass changes—do add to management effectiveness. A wholly separate, but very worthwhile management piece, is the IP Telephony Environment Manager, or ITEM, which is a CiscoWorks package. We think the beefy additional price for this ($20,000+) is worth it for large VOIP networks. Strengths of Cisco's management: Good real-time monitoring, thanks to ITEM, although most screens require manual updating; good on-line help; and reports. Drawbacks: Still too many management interfaces for the user to learn.

ShoreTel's ShoreWare Director, running on a Windows server, provides the network manager's view of the ShoreTel system via a clean and efficient interface. Adding a new ShoreTel user, for example, is all done on a single screen, even for voice mail. It takes a half-dozen or more screens for the other vendors. ShoreTel's management is among the few that earned full credit for interface ease-of-use and navigability. The management offered by ShoreTel is not as functionally rich as packages from, say, Cisco, but there's nothing extra to buy, and there aren't a multitude of different interfaces to learn.

Siemens' management is largely vested in the HiPath 4000 Assistant, which embodies a number of discrete tools, plus some additional applications and interfaces. We were especially impressed, however, with the configuration and monitoring capabilities of a totally separate Windows application, the Deployment Tool. It's a shame all these pieces aren't integrated. Amassing and learning them all is tedious. There is pretty good monitoring of IP QOS conditions, under the Diagnostics tab of the Assistant. Overall: Siemens offers all the tools to do an effective management job. They could be better integrated, and their interfaces more consistent.

Management is not Avaya's strongest suit. But don't misunderstand; its tools are more than adequate to do a good management job. The primary interface to Avaya's IP-PBX world is the Avaya Site Administration (ASA), a dressed-up command-line. Avaya admits that ASA's interface "aspires someday to be a GUI." And Avaya has been slowly embellishing ASA with wizard-type applets for frequently performed tasks; these are effective, well organized and invoked from the high-level ASA interface. An extra must-have for IP-telephony managers is Avaya's separate "VOIP Monitoring Manager," a Windows Java application. It monitors only IP stations and activity, and it does that very well, but it is disjointed from ASA. And users need to toggle between the two.

Features

Avaya and Cisco ended up tied for first in the Features category. Out of 20 possible points, 5 came from our validation that a set of basic phone features (call conference, forward, hold and so on) were all supported by the vendor's IP hard phones. In the test, the vendors each earned all 5 of those points. Another 5 points were awarded if it could be verified that the vendor offered popular add-on packages to customers, including: call centers, interactive voice response/voice recognition (IVR), accounting/billing software and E911 support. In some cases, these were offered through third party affiliations, which was acceptable. Again, the vendors each earned all 5 of those points.

Where the features part of the evaluation became exciting was in "advanced features." This is where the vendor could showcase up to 5 aspects of its system that it considered unique, innovative or in some other way a differentiator. Up to 2 points could be earned per advanced feature, up to 10 total—in other words, 10 percent of the whole evaluation.

Four clear-cut criteria were applied to the scoring of each advanced feature: uniqueness; the percent of users that would likely benefit from the feature; the ease of employing the feature; and the overall value or significance of the feature.

Avaya and Cisco each garnered 7 1/2 points for their advanced features. Avaya's splashiest advanced features included: Extension to Cellular; Seamless Communications; Converged Conferencing; and Multimedia Client.

Extension to Cellular is a set of advanced routing capabilities, where a user's phone-system extension and capabilities are mapped and delivered to off-premises phones, including cell and home phones. Seamless Communications entails use of the Avaya "dual phone," the new 802.11a and GSM combo wireless phone. Avaya is now wooing GSM-based cellular service providers to sign on to participate with this offering. Avaya Converged Conferencing refers to the impressive conferencing package, now called Meeting Exchange, which Avaya gained through its SpecTel acquisition. This SIP-based package features nice Web and audio conferencing; picture quality with the add-on, point-to-point video capability could be better, though. The video quality is much better with the "multimedia client" capabilities that Avaya offers for use with its softphone—very easy-to-use, point-to-point video, with instant messaging and presence.

Cisco fared equally as well with its hottest advanced features, which included: VT Advantage; the latest 5.3 version of MeetingPlace; the Personal Assistant package; and its new mobility application.

VT Advantage is Cisco's $190-per-seat videoconferencing offering, which interfaces through Cisco's IP hard phones and works just like setting up an audio phone call, even for multiparty videoconferencing. Cisco's MeetingPlace, already one of the leading conference/collaboration packages, has been embellished in the release we reviewed, with the addition of video. Well done, good quality, easy to use; it will keep MeetingPlace positioned as the conference package to beat. The Personal Assistant is akin to others' find-me/follow-me, personal call routing, with the addition of speech recognition and text-to-speech (TTS) capability. Cisco's mobility application is akin to Avaya's Extension-to-Cellular, wherein calls can simultaneously ring at the office and home, or cellular, and the user can readily transfer and pick up the call between these sites.

ShoreTel's most impressive advanced features are embodied in desktop applications which tap the capabilities of the ShoreTel5 IP-telephony system. These included: the Operator Call Manager, a softphone with attendant-console capabilities, including real-time presence indication for hundreds of extensions; and drag-and-drop document sharing, which lets you easily send any file to anyone you are on the phone with.

Siemens' advanced features were much more down-to-earth, telephony oriented. The vendor showed off: its wideband G.722-based vocoding, for high-quality voice; the modularity of its add-on "sidecars," expansion units for adding line appearances, soft keys and buttons to its IP phones; and its optiPoint 600 Office, the only phone model we have seen that supports TDM operation as a digital phone with the capacity to switch over to operation as a full IP phone.

Alcatel showcased its OmniTouch Unified Communications package, release 3.0, as its premiere advanced feature. Besides the vendor's rich downloadable softphone module, other pieces are accessed via Web portal on a server. These include: My Assistant, which provides find-me/follow-me, personal call handling; My Messaging, for unified voice mail and email handling (for Exchange/Outlook or Domino/Notes environments); and My Teamwork, a conferencing and collaboration package. Alcatel also showed its Cellular Extension capability, where customers can extend the features, and even the interface, of their desk phones to GSM-based cell phones.

Avaya and Cisco both offered up video as an advanced feature

TABLE 5 Call Quality Ratings

Based on a MOS-equivalent 1-to-5 scale, where 5=Excellent; see Notes below.

Scenario	Alcatel	Avaya	Cisco	ShoreTel	Siemens
Local Office IP hard phone G.711 Campus LAN	4.9	4.6	4.9	4.6	4.9
Warehouse Wireless phone G.729 Campus LAN	4.5	4.3	4.7	Not tested; supports 3rd party wireless via analog links	4.2
Remote Office—Secure IP hard phone G.729 – Encrypted Internet impairments	4.2 pre-release version of encryption	4.3	3.2	Encryption not supported	Encryption not supported
Road Warrior Softphone G.729 Internet impairments	4.1	3.9	3.1	4.2	Not tested; equipment unavailable when tested
Teleworker IP hard phone G.729 Internet impairments	4.2	4.2	3.4	4.3	3.4

Notes:
1. All test calls were placed to an IP hard phone.
2. A team of three experienced Miercom VOIP testers conducted the ratings, placing multiple, round-robin calls in each scenario, and then rating each connection for clarity, voice quality, bi-directionality, and the effects of latency. The resulting ratings, on a MOS (mean opinion score) scale of 1 to 5, have been averaged here.
3. Generally, MOS-type ratings can be interpreted as follows:
   5.0 = perfect; exceptional clarity, the best telephone call quality that can be achieved.
   4.0 = very good; considered toll quality
   3.0 = fair; minimally acceptable for business phone communications; regarded by many today as similar to cell-phone quality.
   2.9 and below = unacceptable for business voice communications.
4. The Internet impairment profile consists of: 100 ms of added one-way latency (200 ms roundtrip), 20 ms of latency, applied uniformly; and 1 percent packet loss, applied in bursts of 4 consecutive packets.

Security

Some of the key components comprising the scores in the Security category of the evaluation are shown in Table 7 p. 36. Specifically, 6 points out of 10 were awarded for verified support for security features including: encrypted call control; encrypted VOIP (RTP) streams; and secure management access.

Cisco and Avaya both garnered 5 of 6 points here, but for different reasons. Avaya was given full credit for VOIP encryption because all the Avaya-made phones, including its softphone, can now secure VOIP streams with 128-bit AES encryption. Avaya lost 1 point because it does not encrypt its call control. The vendor added new software in its latest 3.0 release, which provides encryption of user data and authentication, but does not, strictly speaking, encrypt H.323 call control.

Cisco does fully encrypt and authenticate its call-control messages. And it supports AES encryption on many, but not all of its phones. Its lower-end IP phones and softphones, for example, do not support encryption. Alcatel, too, earned partial credit for VOIP encryption it is adding to several of its phone models. We evaluated an early release of this software.

Secure management access entails username/password control, as well as encrypted management sessions. These are typically addressed by secure-shell tunnels (ssh) for command-line-like interfaces, and secure-HTTP, HTTPS, or SSL, for Web-based interfaces. All the vendors except Alcatel earned full credit (2 points) for secure management access; Alcatel earned partial credit.

The remaining 4 points were based on our assessment of overall attention to security, in terms of: documentation offered customers to help them secure their IP-telephony environment; services offered through the vendor, including by third-party agencies, for on-site security planning or audits; affiliations with third-party vendors that offer security equipment, software and appliances; documented enhancements, improvements or security alerts/fixes pertaining to the vendor's equipment (call controller, gateways, IP phones, other servers).

Based on our assessment here, points awarded (a maximum of 4) were: Cisco, 4; Avaya, 3 1/2; Alcatel, 3 1/2; ShoreTel, 3; and Siemens, 2 1/2.

Performance

We've already addressed a few of our hands-on tests in the architecture section above—specifical-

TABLE 6: One-Way Latency

Environment	Alcatel	Avaya	Cisco	ShoreTel	Siemens
IP hard phone G.711	57 ms	67 ms	54 ms	47 ms	54 ms
IP hard phone G.729	42 ms	76 ms	71 ms	55 ms	81 ms
IP hard phone G.711 - Encrypted	61 ms	62 ms	63 ms	Encryption not supported	Encryption not supported
IP hard phone G.729 - Encrypted	56 ms	67 ms	66 ms	Encryption not supported	Encryption not supported
External analog phone (inbound PSTN call) G.711	76 ms	80 ms	60 ms	51 ms	77 ms
External analog phone (inbound PSTN call) G.729	81 ms	75 ms	69 ms	58 ms	72 ms
IP Softphone (7) G.711	64 ms	75 ms	85 ms	89 ms	Softphone not measured (6)
IP Softphone (7) G.729	89 ms	85 ms	104 ms	89 ms	Softphone not measured (6)
Wireless IP phone G.711	81 ms	92 ms	90 ms	Wireless not measured (5)	Wireless not measured (6)
Wireless IP phone G.729	87 ms	89 ms	92 ms	Wireless not measured (5)	Wireless not measured (6)

Notes:
1. Measurements are from these endpoints/environments to the vendor's same IP hard phone.
2. All results are the average of three measurements, rounded to the nearest millisecond (ms).
3. There were no added impairments in any environments.
4. Network transit delay was less than 2 ms.
5. ShoreTel does not directly sell or offer any wireless phone sets, but supports many via direct analog connections (in the same manner as residential cordless phones).
6. Siemens does offer a softphone, as well as an IP wireless phone (SpectraLink). Latency was not measured in either of these cases due to unavailability of equipment.
7. The vendors' softphones in all cases were individually loaded and tested on the same Compaq Presario 2500 laptop, featuring a 2.4-GHz Pentium 4 CPU and 448 MB RAM.
8. Generally, one-way latencies under 73 ms are not discernible by users (per Bell Labs research), latencies greater than about 85 to 90 ms become noticeable, and latencies beyond 140 to 150 ms become annoying (per Miercom research).

ly, those involving survivability and fail-over of call controllers and gateways.

The majority of the performance metrics, though, were included in the Performance category, worth 20 percent of the evaluation, 20 points out of 100. There were 4 main test sections:

1. Heavy-load call-completion testing
2. Other high-availability features
3. Call-quality ratings
4. Latency measurements

As noted earlier, the results of the heavy-load call-completion testing are shown in Table 3. This is a grueling test, which runs for 9 hours and entails delivery of 50,000 calls per hour to the IP-PBX. Calls are delivered via 8 T1s, at a rate of about 14 calls per second. The vendor's single IP-PBX call controller has to process each call, and route them one at a time from an entry gateway, via IP, to an exit gateway, where they are delivered over 8 other T1s, back to the call-generation test equipment.

This volume of calls equates roughly to an IP-telephony system size of about 3,500 IP stations, given the traditional metric of a 4-minute voice-call holding time. This test does not just exercise the call controller, but all other facets of the IP-telephony environment--including the VOIP-T1 gateways, and the LAN/WAN infrastructure. A minor hiccup at any point, even transient, can result in many failed call attempts.

As explained in Table 3, vendors could simplify the composition of the call load, and increase the prospects for a higher call-completion rate, by excluding voice prompts and DTMF tones from the calls. Obviously, this has the drawback of making the system less user-friendly for callers. So of 8 points that can be earned for this test, 1 point is deducted if voice prompts are excluded, and another point if DTMF tones are excluded.

Congratulations go to Alcatel and Avaya, both earning the full 8 points. Avaya finished with 100 percent call completion. And Alcatel completed the full load of calls within the very small number of allowed call failures (a relative handful, out of nearly a half-million call attempts, still well over 99.99 percent call completion).

Cisco elected to proceed without voice prompts or DTMF calls, and also finished with 99.99+ call completion, earning a maximum of six points in that scenario. ShoreTel achieved a call-completion rate of 99.6+ percent, processing voice prompts but not DTMF tones. This earned the vendor 3 points (no points are awarded if call completion drops below 99.5 percent). ShoreTel confirmed that some minor issues were identified as a result of this testing, and the vendor agreed that these would be addressed in their product in short order.

TABLE 7 Support for Key Security Features

	Alcatel	Avaya	Cisco	ShoreTel	Siemens
Encrypted call control	No	Limited (1)	Yes	No	Limited (1)
Encrypted RTP/ VOIP streams	Limited; an early version was reviewed; to be supported on certain high-end phone models in Q1 2005.	Yes; most complete of the IP-PBX vendors evaluated. Even vendor's softphone supports AES encryption.	Supported on specific phone models; not supported on low-end phone models or on softphone.	No	No
Secure management access	Access to most, but not all, key nodes can be secured.	Yes; access to all key nodes can be secured.	Yes; access to all key nodes can be secured.	Yes; access to all key nodes can be secured.	Yes; access to all key nodes can be secured.
Security score: out of 10 possible points	5	8 1/2	9	5	5 1/2

(1) Strong H.323-based authentication has recently been added. In Avaya's case this will ship with the vendor's next Release 3.0 software in February 2005, and includes encryption of many user values. H.323 call control is not encrypted, however.

Testing Large IP-PBXs

The "Large IP-PBX" test bed consisted of two simulated sites—a "headquarters" site and a "remote office" site, connected by an IP WAN link. At the "headquarters" site, the network infrastructure consisted of Extreme Networks Summit 48 switches and a Cisco 7200 LAN/WAN router. The same Extreme and Cisco infrastructure was deployed at the "remote office" site.

The IP-PBX vendors had the option of connecting to our network infrastructure, or else providing their own switch and router (Layer 2/3) equipment (using IP subnets, VLANs, etc., that we specified). The vendors all needed to also provide their own Power Over Ethernet (PoE) equipment to power their IP hard phones.

The "headquarters" and "remote" sites were connected by a simulated IP WAN. A PacketStorm Hurricane 1800E Network Emulator was used to apply impairments that would be encountered in a typical IP LAN ("campus") environment, and for an IP WAN link over the Internet. In many tests, such as for latency measurements, no impairments were applied. For the VOIP connection-quality tests, though, the PacketStorm applied latency, packet loss and jitter to simulate the various test scenario environments.

The "headquarters" and "remote" sites were also connected via T1s, through an Adtran Atlas 800 central-office switch simulator, to test T1-gateway fail-over and IP-to-PSTN re-routing scenarios. The "simulated CO" also employed a channel bank—an Access Bank II from Carrier Access Corp.—to test fax support and analog connectivity.

For consistency, we ran all the IP-PBX vendors' softphones on the same laptop: a Compaq Presario 2500, with 2.4-GHz Pentium-4 CPU and 448 MB RAM, and employing a Plantronics DSP-400 USB headset.

Latencies were measured with an audio-analysis software tool, Cool Edit, developed by Syntrillium Software (now owned by Adobe). All latencies were measured between the vendor's same IP hard phone on the one end, and all other endpoints, one at a time, at the other end.

One-way latency (see Table 6, p. 35) was measured in the direction from the endpoint (softphone, wireless phone, etc.) to the IP hard phone. To measure the latencies, an audio tap was inserted directly into the phone handsets (or equivalent headset jacks) using THAT-2 units (Telephone Handset Audio Tap), from JK Audio.

A pair of Empirix Hammer systems—a Hammer FX and a Hammer LoadBlaster 500—generated the call loads in our test bed. The IP-PBXs were all initially tested at a load of 50,000 BHCA (busy hour call attempts), which was run continuously for nine hours, usually overnight. The call-generating Hammer units delivered the load from the "PSTN" equally into VOIP-T1 gateways at the "headquarters" and "remote" sites, via a total of 16 T1 spans. All calls were routed across the IP WAN between sites, then back out onto the PSTN, through VOIP-T1 gateways, to the originating Hammer systems, which "answered" each call and verified successful call completion.

Siemens achieved a call-completion in excess of 99.7 percent, without voice prompts or DTMF tones, which likewise earned Siemens 3 points.

Up to 3 points were awarded in the performance section for additional, demonstrated high-availability features, which contribute to sustained survivability of IP-telephony operations. We did not further elaborate, and left it up to the vendor to propose such features, which we then exercised and confirmed. These ranged from redundant power supplies to hot-swappable gateway modules. The added high-availability features that were tested are shown at the bottom of Table 4.

All vendors but ShoreTel earned the full 3 points credit here. ShoreTel earned 2 1/2. (One of ShoreTel's features was added redundancy options that users can buy for their Windows servers, but ShoreTel does not sell server hardware itself. Also, the Windows servers are not key to the survivability of ShoreTel's IP telephony. Phone calls continue even if one or more of the Windows servers fail). So these options aren't as valuable as those offered by the other vendors.

We also conducted call-quality measurements. Here, our lab testers performed round-robin calling between various endpoints in five different scenarios (Table 5, p. 32) and rated each call using a standard MOS (Mean Opinion Score) 1-to-5 scale. In 3 of the 5 scenarios, fairly substantial impairments were applied to the calls, simulating calls being placed over the public Internet.

A maximum of 3 points could be awarded. Averaged ratings that came in under a 4.0 MOS-score threshold resulted in point loss. Alcatel and ShoreTel each earned full 3-points credit. Avaya, Cisco and Siemens were awarded 2 points each.

A final performance metric was one-way latency, which we measured for various endpoints and vocoders (Table 6, p. 35). As long as six of the scenarios showed one-way latency under 100 ms, full credit of 3 points was awarded. All the vendors met this and earned 3 points, except Siemens, which was awarded 2 points. Circumstances prevented us from testing or confirming the latency of Siemens' wireless phone or softphone.

We note again that only one vendor, Alcatel, managed a perfect score (20 points) in this performance section. And while Avaya emerged the overall winner in the Best in Test evaluation, we offer Alcatel the category award of "Best-Performing IP-PBX, Large Systems."

Conclusion

IP-PBXs have proven they can deliver reliability and voice quality equal to predecessor TDM systems, and at competitive prices. This review, our sixth such annual testing of IP-PBXs, finds these systems continuing to improve in call quality and reliability, while continuing to add features that predecessor TDM PBXs cannot support□

In this sixth year of testing, systems continue to improve in reliability, call quality and features

Reasonable requests from the vendors to modify slightly the connectivity and call-generation behavior of the Hammerswere usually accommodated, as long as the 50,000 BHCA call-delivery load was not reduced. Vendors were offered the option of having "DTMF tones" and "voice prompts" enabled or disabled in the generated call load. If enabled, each call first had to pass DTMF tones bidirectionally, and then exchange voice prompts (.wav files saying "hello" and "goodbye") in both directions after call set-up, to be considered "successful." These added to the complexity of the call-load processing, and earned extra points for vendors who delivered high call-completion success rates *with* voice prompts and/or DTMF tones. As Table 3, p. 30, shows, most vendors achieved 99.99+ or 100 percent success rates. A 99.99+ percent rate equates to just a relative handful of failed calls, out of nearly a half-million delivered, and should be considered essentially perfect, given test-bed variability.

Also during the testing, various monitoring systems were used to verify network traffic and other VOIP operational characteristics. These included: Fluke Networks' Optiview Protocol Expert; Ethereal; and the BrixMon application with the Brix 100 Verifier from Brix Networks□

Companies Mentioned In This Article

Alcatel (www.alcatel.com)
Avaya (www.avaya.com)
Brix Networks (www.brixnetworks.com)
Carrier Access Corp. (www.carrieraccess.com)
Cisco Systems (www.cisco.com)
Compaq (www.compaq.com)
Empirix (www.empirix.com)
Ethereal (www.ethereal.com)
Extreme Networks (www.extremenetworks.com)
Fluke Networks (www.fluke.com)
JK Audio (www.jkaudio.com)
PacketStorm Communications (www.packetstorm.com)
Plantronics (www.plantronics.com)
Polycom (www.Polycom.com)
PowerDsine (www.powerdsine.com)
ShoreTel (www.shoretel.com)
Siemens (www.siemens.com)
SpectraLink (www.spectralink.com)
Syntrillium Software (now Adobe) (www.syntrillium.com)