

Large IP-PBXs: A Well-matched Quartet

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For the first time, all the vendors support an H.323-to-SIP gateway, and snazzy, productivity-enhancing features. But security and management issues remain.

IP-PBXs have come a long way since their introduction in the late 1990s, when the major concern was whether these systems could provide adequate voice quality for business applications. Today, voice quality is virtually assured on high-end IP-telephony systems, and advanced features are the major distinction among IP-PBXs today.

All the IP-PBXs reviewed in this article offer—or soon will offer—a host of unified communications functions that will allow users to handle voice, email, instant messages and faxes from one screen, or let calls “follow” the user from desktop to cell phone.

While it’s clear that the vendors have put major efforts into feature development and delivery, issues remain. For the most part, IP-PBX management systems still need work to make them

more intuitive and less difficult to master. And security is still an issue—though it’s being more actively addressed with new features such as voice encryption, automatic VLAN assignment of IP phones and physical locks on phones.

The Review

In September, Miercom invited vendors that offer IP-PBXs supporting 1,000+ stations to participate in a competitive review (results summarized in Table 1). Four vendors, representing the market leaders, agreed to compete. An overview of their architectures and basic functions is shown in Table 2, (pp. 28–29) and features are summarized in Table 3, (p. 35). The participating vendors were:

■ **Alcatel**, which showed its Alcatel OmniPCX Enterprise, R5.1Lx, OmniVista 4760 management system v2.1 and e-Reflexes IP phones (the e-Reflex 4020 was the phone under test).

■ **Avaya**, which came in with the S8700 Media Server and G650 Media Gateway, version 2.0 (beta), Communications Manager v2.0 and Avaya IP telephones (the Avaya 4620 was the phone under test).

■ **Cisco**, which sent its CallManager, version 3.3.3 with SR1, CallManager Administration management application v3.3.3 and Cisco IP Phones (the 7960G was the phone under test).

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TABLE 1 Scorecard: IP PBXs > 1,000 Stations

	Percent Weighting	Alcatel e-ND OmniPCX	Avaya S8700/G650	Cisco Systems CallManager	Nortel Networks Succession 1000M
Architecture	15	85	83	85	86
Endpoints	15	80	85	87	78
Management and Administration	20	82	85	88	75
Features	25	82	89	85	87
Security	10	75	86	85	82
Performance	15	85	87	90	83
The Bottom Line	100	82	86	87	82



Nortel led the way in redundancy

■ **Nortel Networks**, which submitted its Succession 1000M Release 3.0, Optivity Telephony Manager v2.10.56 and Nortel Networks i2000 Series IP phones (the i2004 was the phone under test).

IP-PBXs that support from 100 to 999 stations will be covered in the February issue of *BCR*, and IP telephony systems supporting 99 and fewer stations will be reviewed in April.

The Scorecard

The four IP-PBXs are well matched in terms of features, functionality and performance. Weighing the attributes of each product and scoring it accordingly was an arduous task.

What resulted was a “squeaker,” with Cisco’s CallManager inching into the “Best in Test” slot by one point over the Avaya S8700 Media Server and G650 Gateway, and a tie between the Nortel Succession 1000M and the Alcatel OmniPCX.

This year, we rated the products in six different categories:

■ **Architecture**, in which we assessed physical attributes, such as redundancy, hot swappability, ease of scalability, etc.

■ **Endpoints**, which included IP phones, soft-phones, conference phones, wireless phones, etc.

■ **Management and Administration**, incorporating management applications, including integral and extra-priced options. We rated robustness, ease of use, intuitiveness, real-time monitoring and reporting features, etc.

■ **Features**, including the basic features we would expect to find on an IP-PBX (such as call hold, call forward, etc.), other IP-PBX features (such as automated attendant, unified messaging, interactive voice response, etc.), as well as unique features that set the product apart.

■ **Performance**, in which we assessed call quality, latency, call completion rates, ability to fail over quickly, etc.

■ **Security**, which included an assessment of encryption capability (on call control and feature delivery messages and RTP stream), authentication mechanisms, management interface security (levels of admin access, passwords, etc.), recommended security policy settings and ability to deflect denial of service (DoS) attacks.

While Cisco had the best scores in three categories—Endpoints, Management and Performance—Avaya scored highest in Features, the heaviest-weighted area in this review. Avaya also had the best showing in the Security category.

Architecture

The Nortel Succession 1000M achieved top ranking in this category chiefly for its many redundancy features—key to ensuring survivability. Significantly, Nortel offers not only system redundancy, but also alternatives in the event both redundant systems become unavailable.

For example, in addition to redundant signaling servers, which can also load-share, if both

servers are down, one of the media cards in Succession can take over as the IP telephony node master, allowing IP telephone registrations to continue locally. (However, if both signaling servers are down, calls between sites cannot be processed.) In addition, virtual trunks (which connect sites to provide H.323 gateway functions) also are redundant, and if both trunks on a given link fail, the call-processing software can select an alternate route.

The gateways are controlled by a gatekeeper, which is also redundant (i.e., primary and alternate gatekeepers). In addition, each signaling server can run a Failsafe Gatekeeper in case both the primary and alternate gatekeepers go down.

The Succession 1000M’s media cards are a pooled resource, which means an individual card can be taken out of service for replacement or upgrading without affecting the system itself. Once the card is returned to service, it automatically joins the IP telephony node.

Nortel’s Succession supports H.323-to-SIP gateway functions on the MCS-5100, a separate server that can be located in front of or behind a PBX—either Nortel’s or another vendor’s—which allows Nortel to offer its unified communications features on other vendors’ systems.

Alcatel and Cisco tied for second place in this category. Alcatel’s OmniPCX Call Controller can be configured with a hot-standby secondary controller, which resides on an independent server at a remote location, but is linked to the primary controller via an IP network. Thus, a disaster at the main location would leave one controller unaffected, and operations would continue at the secondary location.

If a remote media gateway loses connection with the main call controller, Alcatel also supports a backup-signaling link over the PSTN. This eliminates the need for a standby processor at remote locations—a cost savings to the end user.

Alcatel’s IP Media Gateways (IPMGs) support no central switching matrix; all switching intelligence is built into the individual cards, eliminating the need for switch fabric redundancy.

Alcatel offers mid-span power insertion through PowerDsine, and its IP phones also may be powered locally via an adapter. Alcatel said it was developing a LAN switch that will provide standard 802.3af power to IP phones.

One of Cisco CallManager’s key architectural features is Survivable Remote Site Telephony (SRST), which resides on a Cisco router. SRST extends a subset of CallManager capabilities to the IP phones themselves, so if the phones cannot reach CallManager, they automatically re-register with the router to continue handling calls.

CallManager supports the widest number of VOIP protocols, including H.323 and SIP, as well as MGCP (Media Gateway Control Protocol) and its own Cisco Skinny Client Control Protocol (SCCP).


TABLE 2: System Architectures Compared

	Alcatel e-ND Calabasas, CA www.alcatel.com/ enterprise	Avaya Inc. Basking Ridge, NJ www.avaya.com
System tested (version, general availability [GA] of release tested)	Alcatel OmniPCX Enterprise, R5.1Lx; GA in 10/03	S8700 Media Server and G650 Media Gateway, v2.0 (beta); GA in 12/03
Call control server	Communications Server (e-CS) running on IBM e-Server x-Series 305	S8700 Media Server running on dedicated appliance
Call control OS and processor	Linux; Intel/AMD processors	Hardened Linux Red Hat 8.0; Intel processors
Other key call-switching components	IP Media Gateway (IPMG) multi-slot chassis for PSTN trunking and digital and analog station interfaces	G650 Media Gateway multi-slot chassis for PSTN trunking and digital/analog station interfaces; contains CLAN/gateway signaling board and media gateway
H.323 gatekeeper	Integrated in Communication Server; also supported externally	Integrated in the S8700
H.323-to-SIP gateway support	Yes	Yes
Management system	OmniVista 4760, v2.1 (extra priced at \$24,828 plus \$4,000 for server)	Communication Manager, v2.0 (software license based on number of system users @ \$35/user up to 100 and \$175/user over 100); includes Avaya Site Administration (ASA), Native Configuration Manager (NCM), and VOIP Monitoring Manager, v2.0.7
Vocoders supported	G.711, G.729A, G.723.1; VAD on G.729A and G.723.1	G.711, G.729A&B, G.723.1; silence suppression on G.729
VOIP protocols supported	Native call control: H.323, v2; Intra-system: proprietary encapsulated in H.323 stack; H.323 and SIP; Intersystem: ABC/Q.Sig, H.323 and SIP	Native call control: H.323, v2; Intra-system: CCMS (proprietary extension to H.323); Inter-system: H.323, v2; Q.Sig over H.323
Station interfaces	Analog, digital, IP, wireless	Analog, digital, IP, wireless
Maximum overall station capacity*	5,000	36,000
Maximum capacity, all IP stations	4,000 if all IP phones; 3,000 if all IP softphones	12,000 (mix of IP phones and softphones)
Maximum number T1/E1 trunks and analog trunks*	66 T1/E1; 2,000 analog	8,000 T1/E1 or 8,000 analog based on two S8700s as the discrete system and approx. 34 G650s
IP phone tested (version, release date)	Alcatel Premium e-Reflexes 4020; version 2.18, GA in 7/03	Avaya 4620 IP Telephone; v2.0, GA in 12/03
Third-party IP phones supported	Pingtel SIP phones; H.323-based Polycom conference phones; Audiocodes analog/IP devices	Standard SIP phones via the Avaya Converged Communications Server proxy server
Power delivery options for IP phones	PowerDsine mid-span power; IP phones have local power adapter	1152 MidSpan power; Cajun P333T LAN powering per 802.3af; 110VAC plug-in; 1151B1/B2
Cost per station (US list price)**	\$676	\$596

*Based on a single, discrete, local system under a common call controller; assumes 4-to-1 station-to-trunk channel ratio and a mix of station types (IP, analog, digital, etc.) unless otherwise indicated.

**Based on typical large enterprise configuration of a system supporting 1,600 stations (consisting of 1,400 mid-range IP phones, each with a secondary switch port, at least six feature buttons, an LCD display and 200 softphone licenses), and including all tested components (management, voice mail, devices to power IP phones, etc.). It is assumed that all US list prices are subject to discount.

Cisco Systems, Inc. San Jose, CA www.cisco.com	Nortel Networks Brampton, Ontario, Canada www.nortelnetworks.com
Cisco CallManager, v3.3.3 with SR1; GA in 8/03	Succession 1000M, Release 3.0; GA in 10/03
CallManager running on Media Convergence Server (MCS) 7845 and MCS 7835	CSE 1000M Call Server resides on board inside chassis
Windows 2000 Server for MCS 7835H; Windows 2000 Advanced Server for MCS 7845	Vx-Works, v5.3.1; Motorola processor
Media gateway functions (for T1 and PRI trunking) via module on Cisco 3725 switch	Signaling Server for IP station and trunk interface; Media Gateway multi-slot chassis for TDM trunk interface
Cisco Multimedia Conference Manager (MCM) supports gatekeeper functions	Integrated in the Signaling Server
Yes (in release 4.0)	Supported on MCS 5100, which can be integrated into Succession 1000M
Via CallManager Administration, v3.3.3 application integrated in CallManager; includes tools for bulk admin, reporting, monitor, etc.; separate CiscoWorks IP Telephony Environment Monitor (ITEM), v2.0 for monitoring IP telephony network, extra priced at \$19,995 for Enterprise edition	Optivity Telephony Manager, V2.10.56 (extra-priced at \$7,130); separate Element Manager comes with the Succession system
G.711, G.729, G.723, wideband (IP phone to IP phone); VAD support on all codecs	G.711, G.711 clear channel, G.729, G.729A, G.723.1; VAD on G.729A
Native call control: Skinny Client Control Protocol (SCCP); Intra-system: SCCP for Cisco IP phones; H.323 for third-party devices; Inter-system: H.323, v2; Q.931 over PRI	Native call control: H.323, v3 and Unistim (Nortel proprietary); Intra-system: Unistim, H.323, v3 and H.323 with proprietary additions; Inter-system: Meridian Customer Defined Networking (MCDN) over H.323
Analog, IP, wireless	Analog, digital, IP, wireless
30,000 in a CallManager cluster of eight in load-sharing configuration	13,000
30,000 IP phones in CallManager cluster; 10,000 if all softphones	10,000 (all IP phones or all IP softphones)
3,200 T1/E1 or 3,200 analog in CallManager cluster	224 T1/E1; 8,000 analog
Cisco IP Phone 7960G, version 5.0(3); first released 10/03	Nortel Networks i2004, v1.59; first released 10/03
SCCP-based phones from Sanyo, ADTech, SpectraLink, Symbol, IP Blue, Telstrat and Polycom	No
Inline power via Catalyst LAN switch, Cisco External Power Patch Panel, local transformer module	Inline power via BayStack 460-24T-PWR, local power module
\$534	\$776



Several phones can display network statistics such as jitter and packet loss

Cisco also offers the most elegant scalability. Up to eight CallManagers can be combined in a cluster, providing clean, easily replicated call control to both local and geographically dispersed systems. One cluster can support up to 30,000 IP phones or 10,000 softphones.

One downside for the CallManager: no hot swapping of cards on the media gateway, which resides on a Cisco 3725 switch. The system must be taken down for upgrading.

A key selling point for Avaya in this category is direct connection of S8700s through fiber interfaces to provide real-time memory mirroring. We tested this capability by simulating a failure in the primary S8700; failover to the secondary S8700 occurred within eight seconds—the shortest failover time observed.

Avaya's S8700 Media Servers do not support load sharing *per se*, but their principal IP resource modules (control LANs and media processor) do load share and are also redundant. In addition, Avaya's system supports alternate gatekeepers, whereby IP devices can re-home to up to eight alternate gatekeeper addresses.

Endpoints

Cisco took the lead in this category thanks to the breadth of its offerings, as well as some notable features and excellent IP phone design.

Cisco offers one of the widest arrays of IP phones—from the low-cost 7902G, which costs only \$130 plus \$40 for the software license—to the high-end 7935G, which sells for \$1,195. In between are eight models, including a touch-screen color phone, the 7970G.

We saw Cisco's new IP softphone in beta; it was scheduled for general availability this month. This new product includes Skinny, which allows the softphone to support all basic features found on a Cisco IP phone.

Audioconferencing is available via Polycom's IP Conference Phone, which Cisco OEMs and sells direct.

In addition to the endpoints, Cisco also supports several nifty software applications that enhance their usefulness. WebDialer, for example, which is used with CallManager, allows Cisco IP phone users to place calls from both Web and desktop applications. There is no extra charge for this feature.

Cisco also offers—again at no extra charge—attendant console software that runs on a desktop machine. This application can be used to set up speed-dial groups, personal directory, etc. Also available is IP Manager Assistant (IPMA) software to support call handling and grouping, allowing one person, for instance, to handle calls for multiple bosses in different locations.

An especially noteworthy feature on the Cisco IP Phone 7940 under test was the ability to view status messages and network statistics (such as jitter, packets lost, codec in use, etc.) on the phone's

LCD display. We also liked the phone's "help" button that allowed us to access information about selected features.

While Cisco supports a video application via H.323, we saw an upgrade that will be available in its 4.0 release, which was scheduled for late 4Q03 delivery. (It should be generally available by the time you read this.) In 4.0, video has been tightly integrated into CallManager, and the video endpoints run on Skinny, working just like Cisco's 7960 IP phones to support call forwarding, alternate routing, etc. What we saw represented fairly good audio and video quality.

The only notable downside for Cisco in this category was a lack of a full-duplex speakerphone.

Avaya placed second in this category. Its product line includes seven models of IP phones from \$195 to \$900, three models of wireless/cordless phones, the Avaya IP Softphone, an Avaya Soft-console (attendant console) and an Avaya IP Agent call center.

All its IP phones support a full-duplex speakerphone that handles up to six parties. The 4690 IP speakerphone, which is based on Polycom's IP Furby, is also supported. The 4690 provides a high-quality unit with speakers and microphones, so many speakers can participate in the conference. Via one dedicated number, a person can host a conference at any time without the need for a third-party to set up the call. The default for the maximum participants in a conference is 10, but this can be increased during the call through an auto-expand feature.

The Avaya IP phone under test was the 4620 IP Telephone. It incorporates a large screen (168 × 132 pixels) and an integrated switch for PC connection. Ten fixed-feature buttons for standard operations, such as conference, transfer, hold, redial, etc., are incorporated, as well as four soft application feature buttons.

Avaya won points in this category for its Avaya Terminal Configure (ATC) PC client application, which lets users easily assign features to soft buttons and configure the IP phone.

At the high end of Avaya's IP phones is the 4630 Screenphone, a 256-color touch screen model with 120 speed dials, a call log and statistics (codec in use, packet loss, etc.) shown on the phone, along with a new "click to dial" feature and one-touch dialing from a directory application.

Since we last evaluated the 4630 a year ago, Avaya has added a bigger browser window (320 × 240 pixels), replaced its hub with a switch for PC connection and incorporated a wireless headset. An HTML Web browser incorporated into the 4630 allows access to Web content (weather, news, entertainment, etc.) via the phone's display.

Both Avaya and Nortel support a Net6 gateway server application that strips a Web page of content such as banners and graphics, to include only pertinent information that can then be displayed on an IP phone screen.



Cisco has been doing its homework when it comes to management

Alcatel came in third place in this category. While Alcatel offers a solid product line with some good features, it doesn't yet offer any high-end, touch-screen color IP phones, as Cisco and Avaya do. It also relies more heavily on third parties to deliver products.

Alcatel's Advanced e-Reflexes IP phone product line includes only three models, from \$195–\$485. But Alcatel also supports third-party SIP phones, such as Pingtel's.

One of Alcatel's most notable features was IP Plugware, which lets users upgrade four models of Alcatel digital phones to an IP phone, preserving investment.

The vendor also offers a well-designed, intuitive softphone application, called Alcatel My Phone, with both Windows-based and Web-based client options. Powerful features—the ability to place a call via name or number, customize the phone set and integrate with Internet Explorer—are available via the softphone.

Alcatel does not sell conference phones directly, but recommends IP and analog Polycom sets, which are sold via Alcatel channel partners. Third-party video equipment—based on SIP or H.323—is also supported. Alcatel's speakerphone capability is currently only half-duplex.

The Alcatel IP phone under test was the Advanced e-Reflexes 4035. The phone lost some points for its small, 2 × 40 characters (two-line) display, compared to the much larger displays now available on its competitors' phones. The phone also still has paper labels, which are rapidly becoming a thing of the past on software-controlled phones. But the Alcatel 4035 gained points for the keyboard at the bottom of the phone, through which we could send messages.

Currently, no statistics are available through the e-Reflexes phones. Users must Telnet to the phone to obtain them.

Nortel's overall score of 78 in this category reflects several limitations compared to the competition. This includes lack of support for an IP conference phone, which all its competitors in this review offered—either directly or via third party. (Nortel said it will offer a Polycom phone in a future release.)

Nortel offers three IP phone models (the i2004, i2002 Internet Phone and i2001) and two softphones—the i2050 Softphone and the i2050 Mobile Voice Client for PDA. Symbol NetVision H.323 phones are also supported.

Nortel's endpoints score took a hit for lacking a call log (planned for 2Q04 delivery, according to Nortel) and a missed-call indicator. Further, while the i20XX Series phones have a larger display than the Alcatel IP phones (8 × 24 characters on the i2004 and i2050 and 4 × 24 on the i2002), they are still somewhat limited compared to Cisco's and Avaya's.

The Nortel IP phone under test was the i2004, which supports interactive access to HTML and

XML Web-based data. After it is stripped of graphics content (via the Net6 gateway mentioned previously), Web content is reformatted into primarily text-based information, which can be displayed on the i2004 phone.

Some limited graphics support is also built into the i20XX phones. For example, when a voice mail is playing back, a status bar on the display shows how much of the message has played.

Nortel's i2050 Softphone supports a useful diagnostic utility that displays information about the last call. Operational measurements, including counts of registrations attempted, registrations confirmed, average jitter, percentage of lost packets, etc. are gathered hourly.

Management And Administration

Cisco's CallManager took the gold in this category. It's clear that the vendor has been doing its homework, and its management application has become one to beat in this market.

The CallManager Administration application, which is integrated into CallManager, is well organized, Web-based (via Netscape and IE), intuitive, menu-driven and—as high-end IP-PBXs go—fairly easy to use. A CLI is also available.

CallManager Administration includes several plug-in management tools that come with the price of the system. We were especially impressed with the Bulk Administration Tool (BAT) that allows bulk transactions, such as adding, updating or deleting a large number of phones, users and ports—an excellent timesaving tool for large deployments.

Other plug-ins include a CallManager Serviceability CDR Analysis and Reporting (CAR) tool that generates reports on quality of service, traffic, user call volume, billing and gateway activity; a Real-Time Monitoring Tool (RTMT) that shows performance information of all CallManagers in a cluster; and a Quality Report Tool (QRT) that monitors voice quality and problems on Cisco phones. QRT extends to the IP phones as a Windows NT service.

Up to eight CallManagers can be clustered together to form one discrete system, and each cluster can be managed from a single CallManager Administration application. However, a separate CallManager Administration is needed for each cluster—you can't centrally manage all clusters, which is one of the few notable downsides of this application.

In addition to its integral management application, CallManager also supports an extra-priced application called IP Telephony Environment Monitor (ITEM), which is part of the CiscoWorks management application. ITEM provides health and fault monitoring information of the IP network and IP telephony implementation. ITEM is supported on an Apache Web server; a Java plug-in version 1.3 or higher is required. It is also accessible via Web browser.

All the vendors offered innovative “advanced” features, with Avaya in the lead

Coming in second place is the Avaya S8700/G650, which supports several applications under the Communication Manager umbrella. Included are the browser-based Native Configuration Manager, Windows-based Avaya Site Administration (ASA) for PBX and telephony applications, and the VOIP Monitoring Manager, which monitors active calls, station status of endpoints, etc. An Avaya Installation Wizard and Upgrade Wizard come with the package.

Also supported are two CLIs accessed via the ASA: Emulation and Graphically Enhanced (called GEDI). Emulation is a traditional, Telnet-type window, while GEDI offers enhancements that make the CLI easier to work with. For example, typing a “?” brings up a list of commands that will take you where you want to go, in most cases without a lot of digging around. Also, help is readily available to describe what will happen once a command is invoked.

While the information within each of Avaya’s management applications is well organized, we found it difficult to contend with so many different applications. It can be confusing and requires a learning curve to get used to.

On the plus side, Avaya’s management comes with some good animated tutorials, online and context-sensitive help for all “task” tab items. Help is command-line driven under GEDI, and the mouse-over help in the wizards is excellent.

Ranking third in management was Alcatel’s OmniVista 4760 “all in one” management application for the OmniPCX. OmniVista is an extra-priced application that costs \$24,828.

Included within OmniVista 4760 is 4760i, a Web-based configuration tool for moves, adds and changes that has an OmniVista-like interface; an OmniPCX Wizard that simplifies the installation process for initial configurations; and a Web Monitoring Tool, which provides a GUI to access real-time information on system parameters.

A plus: OmniVista 4760 supports central management of up to 100 systems—a significant advantage in large networks. It also supports a very comprehensive directory tree structure.

But with such a robust system supporting so many tasks and features, it’s easy to lose your way quickly in this application. A better organization of the directory tree—perhaps the option to view it alphabetically—would increase navigability.

Alcatel also lost some points for its non-intuitiveness. In several instances, we had to drill way down into the system to find information or make changes. For example, it was difficult to figure out what trunk group was associated with which card in the system. We had to drill far down into the trunk area to find that information.

OmniVista also was slightly “buggy” here and there. For example, when we launched help, the index tab failed to respond.

The Nortel Succession 1000M supports several management interfaces: a CLI for configura-

tion; Element Manager, a Web-based configuration interface supported on the Signaling Server; and the optional, extra-priced (\$7,130) Optivity Telephony Manager (OTM), a Web-based/Windows interface providing higher-level management capabilities, such as navigating multiple systems, adding sets via templates, billing and alarm consolidation.

Nortel told us that a month of training is required to work with the management system, but we believe it could take much longer to become proficient on it. Of the four management applications we evaluated, Nortel’s would require the steepest learning curve.

Both Element Manager and the OTM provide straightforward access to all information via menus, a content outline tree and point-and-click navigation. The Element Manager’s tree is collapsible and hierarchical.

But what makes Nortel’s application difficult to work with is understanding the content on the screens and also navigating through multiple screens to complete a task. This is the same criticism we had of Nortel’s management last year, and it seems nothing has changed to make the experience any better.

Nortel’s management also lost some points for its slowness. The vendor said that in our test, the OTM hardware was not set at the recommended level (512 Mbytes), which hampered performance—though we used the settings pre-configured by the vendor.

Features

The Features category—the most heavily weighted area in this review—is the one in which all vendors showed the most development during the past year. Support for video, text-to-speech, integrated voice mail/email and other features that take full advantage of convergence are either available or coming soon.

Support for basic features, such as call forwarding, music on hold, call transfer, multiple line appearances, etc. are virtually ensured on IP-PBXs today. Further, all products also support most of the traditional PBX system features and add-ons, such as alternate routing, automated attendant, speed dialing, recorded announcements, etc. (Table 3).

What distinguishes one IP-PBX from another in this category is “advanced” features—an area in which all the vendors displayed a number of innovative and useful offerings.

Avaya, first in this category with an 89, showed its Avaya IP Softphone, which supports a tight integration with Microsoft Explorer and Outlook over a built-in Windows TAPI. When a call comes in on the softphone, a profile of the caller pops up on the screen. The softphone also records inbound and outbound call information, including instant messages, in a journal so that users can go back and review them.

TABLE 3 Features Compared

	Alcatel OmniPCX Enterprise	Avaya S8700 Media Server/ G650 Media Gateway	Cisco Systems CallManager	Nortel Networks Succession 1000M
Percent of basic features supported*	100	94**	100	94**
Automated alternate routing	Optional	Integral	Integral	Integral
Automatic route selection	Integral	Integral	Integral	Integral
Automated attendant	Integral, but requires license fee	Integral (can be IVR and DTMF; additional server adds speech recognition)	Integral	Optional via CallPilot*** or Integrated Call Assistant
Hunt groups	Integral	Integral	Integral	Integral
Least cost routing	Integral	Integral	Integral (via AAR Groups)	Integral
Night service (time of day)	Integral, but requires license fee	Integral	Integral	Integral
Recorded announcements	Integral	Integral	Planned for 4.0 release	Integral
System speed dialing	Integral	Integral	Integral	Integral
User directory	Integral with access to LDAP directories	Proprietary, but can be synchronized with LDAP and other databases	Integral (via embedded LDAP directory)	Integral
Billing/accounting	Optional within OmniVista 4760	Optional third party	Optional via third party	Optional via Optivity Telephony Manager
Voice mail	Via Alcatel 4645; runs on Communication Server or on separate server	Optional via Avaya Modular Messaging 1.1 with MS Exchange integration	Optional via Cisco Unity Voicemail or third party application	Optional via CallPilot
Call Center/ACD support	Integral, but requires license fee	Integral with license fee	Optional via Cisco IPCC Express Edition	Integral with basic ACD; optional via Symposium Call Center Server for increased functionality
911/E-911	Optional via third party	Integral	Optional via Cisco Emergency Responder	Integral
Interactive Voice Response (IVR)	Via standalone Alcatel CC-IVR or third party	Integral with license fee; optional add-on for higher capacity and advanced features	Optional via Cisco IP IVR	Optional via CallPilot or Periphonics portfolio
Unified Messaging (UM)	Basic UM via IMPA connector; Advanced UM via "My Messaging" in OmniTouch Unified Communications app	Optional voice mail system feature	Optional via Cisco Unity	Optional via CallPilot
Other add-on applications, special features	Integrated wireless support; third-party integration for 802.11x wireless phones	Audio conferencing; Multi-media Contact Centers; EC500 Extension to Cellular	Cisco Personal Assistant, Standard XML applications, Cisco WebDialer	Integrated Call Director (Find-Me/Follow-Me); Integrated Voice Services (hospitality app); Integrated Call Assistant

*Analysis is based on 16 features that Miercom considers necessary to support business environments. For a complete list of the 16 features tested, see the sidebar "Basic Telephony Features Tested" that accompanies this article.

**Avaya and Nortel each lost a point for the Missed Call Indicator feature. Avaya supports this in a call log, but does not offer any visual indication of a missed call on the phone. On the Nortel product, this feature is not currently supported, but planned future release.

***The Nortel CallPilot option costs \$99,180 for CallPilot hardware, software and user licenses.

Several applications integrate with Microsoft Exchange and Outlook

One of the impressive things we saw on the Avaya IP Softphone was the ability to go to a Web page, find a phone number on that page and click on it to dial the number. The Avaya IP Softphone can also be configured to search for contacts against an LDAP directory, an Outlook “contacts” list or a local database. The softphone can also be configured for detailed event logging.

Avaya won additional points for its Speech Access for Communication Manager (version 1.0) voice recognition, which runs on a Windows 2000 server. Using this application, we set up a speed-dial directory and spoke the name of a person to place a call. Via voice commands, we accessed the system and forwarded calls, dialed by number, set up a conference call and retrieved voice mail. (Voice mail requires a separate package for speech access to Unified Messenger.)

Avaya Access Phone Status for the visually impaired was also demonstrated. This client application provides the status of the phone—who is calling, whether there is a line open to make a call, etc.—via voice through the PC.

Nortel also had an impressive showing in this category. Some of Nortel’s many features, including IVR, voice mail, unified messaging and automated attendant, are available through the vendor’s extra-priced (\$99,180 for hardware, software and user licenses) CallPilot feature.

The vendor’s new SIP-based Meet Me Conferencing application, which one of our testers

dubbed “NetMeeting on steroids,” was what propelled Nortel to second place in Features. Meet Me Conferencing greatly expands the ability of employees to work with one another remotely—exchanging files, whiteboarding, instant messaging—even conducting peer-to-peer video calling via laptop. A future release will support multipoint videoconferencing.

Meet Me Conferencing requires Nortel’s Multimedia PC Client, a SIP client that runs on the vendor’s MCS-5100 SIP gateway. The MCS-5100 provides all of the features cited above to any phone on the user’s desk—digital, analog, IP—and supports any other vendor’s PBX. The MCS-5100 can reside in front of or behind a PBX, and it also supports Centrex.

Nortel also showed us its i2050 Mobile Voice Client, which runs on a PDA (Toshiba, Dell or Hewlett-Packard), allowing it to act as a softphone. The client can be integrated with a Microsoft Outlook database from which we placed calls from our local directory.

Cisco also demonstrated several advanced features available now, as well as others that will be delivered in CallManager’s next release. (Our evaluation and scoring took into consideration only features shipping during the time we tested.)

First up was Cisco’s Unity unified messaging (we evaluated version 4.0, Build 3), which resides on a separate Cisco MCS 7845I server. Unity is an extra-priced option: the server costs \$20,995, a per-user license is \$65 and a license for the server costs \$5,000.

A tight integration between Microsoft Exchange and Unity allows voice mail to be read in Outlook (all voice mail and email messages are in the same mailbox). Messages marked urgent came with an audible alert, and we could respond to any message via phone or laptop.

An extremely useful Unity feature is the ability to listen to a whole message or a summary. Also, a text-to-speech engine is available as an option.

A Cisco Personal Communications Assistant, included with Unity, provides a portal to the user’s mailbox to change the phone password or greetings, set up notification options, screen calls, etc.

Basic Telephony Features

Below is a list of 16 telephony features that Miercom deems “must haves” in any IP system used in a business environment. Our analysis of basic features on the IP-PBX systems tested in this review was based on this list.

- **Call conference:** Audio path for multiple parties on a single call, established via user key-strokes with no outside intervention.
- **Call drop:** Terminate a call (usually via a button) without hanging up the receiver (e.g., “release” or “end call” button).
- **Call forward all:** User can readily redirect all calls to another station or location.
- **Call hold:** A hard or soft key on the phone that places incoming call on hold and retrieves call.
- **Caller ID:** Phone displays the name and/or number of the calling party.
- **Call park/retrieve:** A user can put a call on “hold” and pick up from another phone within the system.
- **Call transfer:** User can redirect an answered call to another user; this includes restrictions on trunk-to-trunk transfers.
- **Group pickup:** A user can answer a call ringing on another extension, usually via a “star code” (i.e., * + code).
- **Last number redial:** Button on phone causes last call to be re-dialed.
- **Message waiting light/indicator:** A visual indicator of a voice mail message (e.g., blinking lamp)
- **Missed call indicator:** Lists missed calls.
- **Multiple line appearance:** A single phone can have multiple, repeated instances of a single phone extension.
- **Music on hold:** A caller hears music when placed on hold.
- **Mute:** Phone supports mute button to disable its microphone.
- **One-button speed dial:** Programmable one-touch speed dial on the phone set.
- **Volume control:** User can control volume of speaker, handset and ringer□

Cisco's extension mobility feature, which is standard on CallManager, allows a user to "move" his/her profile to another phone—as long as it is in the same phone cluster. For security, an auto logout feature can set the phone back to a default profile once the user leaves the phone.

Cisco also showed us Cisco Emergency Responder (CER), version 1.2 (build 1), an application to handle 911 calls more effectively by dynamically, automatically updating physical IP phone locations. CER can be used with analog, IP and 802.11b wireless IP phones.

Here's how it works: Through SNMP, CER gets the MAC address of any new devices, which are then matched with extensions within CallManager. An audit trail shows who made the changes and what was modified.

CER provides multiple mechanisms for internal alerting of 911 calls, including a Web-alert tool. It also supports multiple levels of failover, including redundant servers. CER is an extra-priced option, which would cost about \$30,995 for a system supporting 2,000 IP phones and the requisite software, server hardware and user licenses.

The highlight of Alcatel's advanced features offering was OmniTouch Unified Communications, parts of which we reviewed, though the full offering will not be available until next month.

The OmniTouch suite—based on SIP, XML and vXML applications—includes My Phone, a softphone supporting Microsoft client-based or Web-based access to OmniPCX Enterprise's features, My Messaging software-based UM, and My Assistant, a personal call routing application. All of these functions are presented on one Web page, which the user can customize.

The My Messaging area displayed voice mails, emails and faxes, requiring a manual refresh to update the display. We could reply to voice mail via email, and messages could be integrated with an LDAP directory and/or Microsoft Exchange Active Directory.

All OmniTouch applications can be ported to a PDA, and eventually to a cell phone.

Alcatel also showed us a Cellular Client application, which is embedded into the OmniPCX Enterprise Communications Server. It allows an external phone, such as a cell phone, to be configured in a "twin set" mode with a desktop phone, enabling both to ring simultaneously and to share a mailbox and programming options. This allows users to take calls from any location, and calls in progress may be passed from the cell phone to the desktop phone. When Alcatel demonstrated this feature, we detected no audible indication that a transfer had occurred.

Also noteworthy was the IMAP Connector feature, which allows access to the Alcatel 4645 and 4635 messaging systems via IMAP-4-compliant applications, such as Netscape Messenger,

Microsoft Outlook and Lotus Notes. Users can access voice mail, email and fax from a single client; all can be forwarded via email or stored on the PC.

Security

The Avaya S8700 took the honors in this category, chiefly for its support of IP phone-to-IP phone encryption on voice RTP streams. In addition to the proprietary Avaya Encryption Algorithm (AEA), the vendor now supports standard AES 128-bit encryption.

We tested Avaya's IP phone to determine how encryption affected voice quality and latency. Not much, it turns out. With AES encryption and G.711 vocoding, the voice quality rating was 4.83 (out of a possible 5.0). With AEA encryption and G.711 vocoding, voice quality dropped only slightly, to 4.33.

While the Cisco CallManager was evaluated based on the currently shipping software version, we were able to take a peek at some new security features that will be available on 4.0.

In 4.0, Cisco supports AES 128-bit encryption on both the media stream and call control. The call controller generates a new key for every call, and each call has its own key code. All encrypted phones will have a digital certificate burned into them, according to Cisco. The CallManager will authenticate each call to avoid "man-in-the-middle" attacks.

A nifty feature on Cisco's high-end, touch-screen, color phone is a cable with a lock on the end, much like a bike lock. We wrapped it around a table leg to secure the phone to the table.

CallManager lost a couple of points in this category for lack of SSL support, but the vendor told us that this would be available in 4.0.

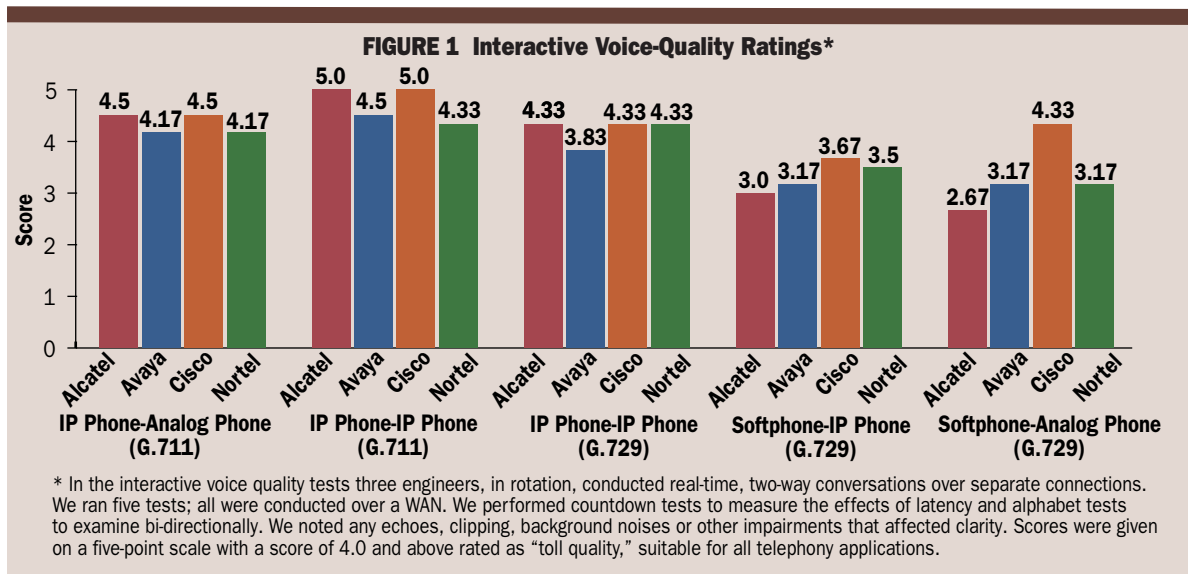
All IP-PBXs proved susceptible to at least one type of denial of service (DoS) attack, and it was not the same one on all products. On the Avaya G650 a couple of DoS attacks caused its cards to go out of service. Nortel's Media Gateway TDM trunk interface and gatekeeper were vulnerable to a particular DoS attack, causing both to reset. On the Alcatel OmniPCX, one type of DoS attack caused the IP Media Gateway to reboot.

As we've noted previously, correct use of VPNs to segment the voice and data networks, firewalls, access control lists and other security measures are necessary to provide sufficient protection against security threats in an IP-PBX environment (see *BCR*, February 2003, p. 40.)

Alcatel lost some points in this category because its OmniPCX does not support SSL or SSH. Also, the product supports no encryption, and Alcatel did not reveal any plans to address that in the near future. On the plus side, the OmniPCX supports automatic VLAN assignment of its phones.

The Nortel Succession 1000M supports no SSL on its Element Manager and no encryption.

Alcatel's feature strengths included applications based on SIP, XML and VXML



But Nortel's IP phones have no open TCP ports, so they are not susceptible to attacks that require a port number.

The Nortel Succession won points for its Equipment LAN (ELAN). In this arrangement, the call server, media gateway and signaling server communicate with one another across the ELAN, which resides behind the PBX and is, therefore, protected from the public network.

Performance

Delivering toll-quality voice via IP phones is no longer an issue on high-end IP-PBXs. All the interactive voice rating scores conducted on the IP phones under test were well within "toll quality" range—4.0 or above on our scale (Figure 1).

Voice quality on tests conducted from IP softphones to an IP phone or analog phone, however, tended to be lower overall, but, for the most part, still within acceptable ranges for most business applications (i.e., 3.0 and above).

The reason for this lower quality: Voice quality on a softphone can be affected by the underlying hardware—laptop, headset, etc.—something to consider when deploying these devices.

One-way measurements conducted over the WAN between IP phones revealed that latency on all the products remained below 100 milliseconds (ms) on baseline tests (Figure 2). Average baseline latencies recorded for Alcatel and Cisco between an IP phone and an analog phone were 108 ms and 112 ms, respectively. That's on the high side, but it didn't affect voice quality, which remained above 4.0 in both cases.

We also conducted several tests with various types of impairments, such as added latency, jitter and packet loss. In the first "campus" test between IP phones, we introduced 10 ms of added latency. All latencies increased, but none substantially.

In the next test, we added 30 ms of latency and 10 ms of jitter. This pushed the latencies on all systems above 100 ms, but, again, none rose so high as to seriously affect voice quality.

This was not the case on our third "Internet" test, which added 100 ms of latency, 1-percent packet loss (based on eight consecutive bit-error bursts) and 20 ms of jitter in each direction. Latencies on all products soared—to 212 ms on the Alcatel OmniPCX and the Cisco CallManager, 195 ms on the Avaya S8700 and 183 ms on the Nortel Succession 1000M.

Due to time constraints during testing and some test equipment glitches, we could not obtain a full set of valid voice quality ratings with the impairments on all products. However, we did complete those on the CallManager and OmniPCX, and results showed that, with the impairments, both products maintained acceptable quality on IP phone-to-IP phone tests.

Even on the "Internet" tests in which we introduced the most significant impairments, call quality was a 3.0 on CallManager and 3.17 on the OmniPCX—both acceptable scores for most business applications.

What pushed the Cisco CallManager to the top slot in Performance was not only its excellent showing in voice quality, but its 100-percent call completion rate.

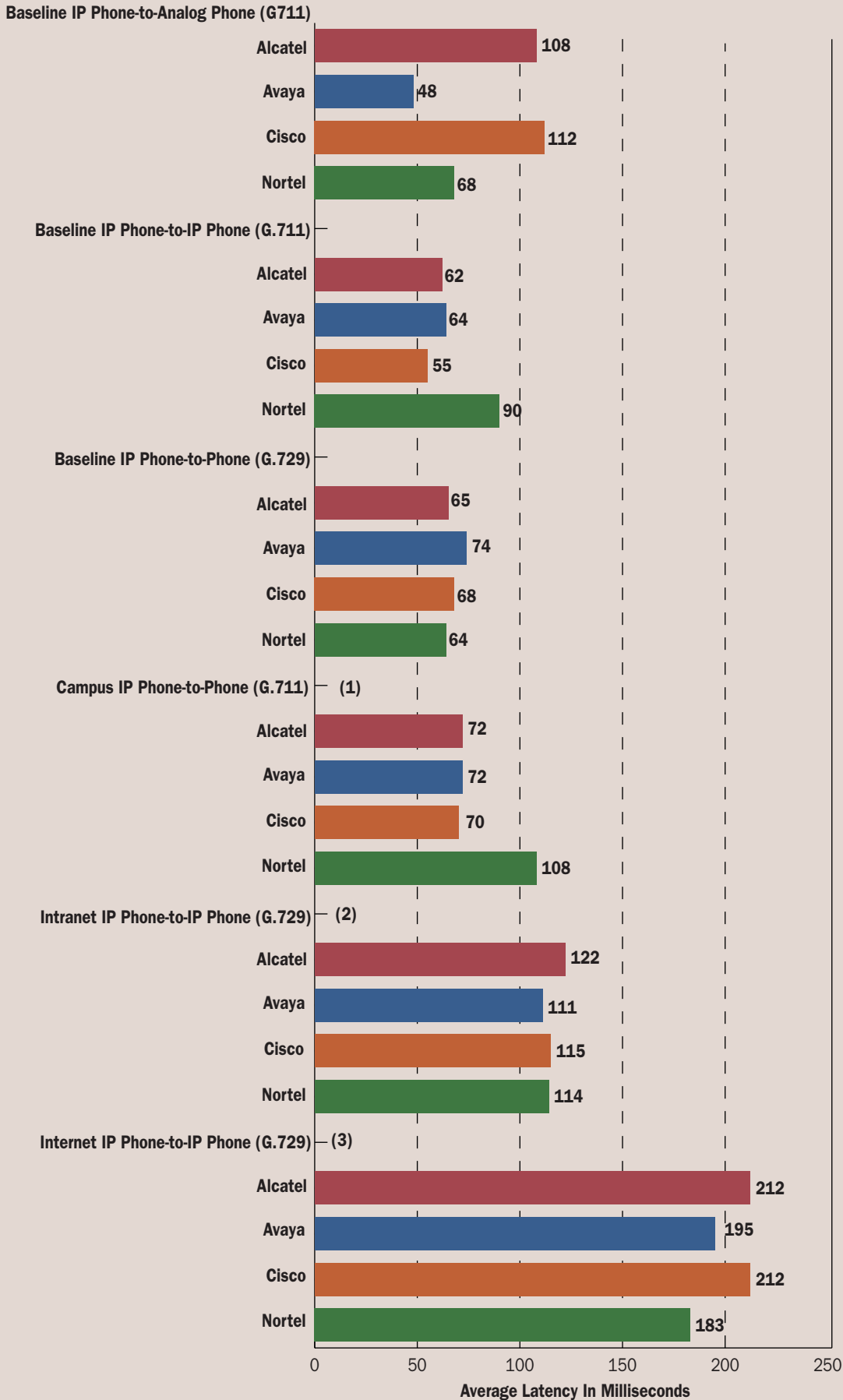
To measure call completion, we conducted tests over a nine-hour period in which each IP-PBX handled thousands of repeated call setups and teardowns. Calls were administered via a Hammer LoadBlaster.

Avaya's S8700/G650 had the next-best call completion—a "five-nines" rate of 99.9995.

The Alcatel OmniPCX came in at 99.9958, and the Nortel Succession at 99.9866. Both vendors lost a few points for not reaching "five-nines" or higher—the goal for PBXs.

Our performance measurements also included the time it took to automatically switch over to the PSTN when there was a failure in the IP network. While most of the products completed these failovers and fallbacks in less than a minute, Nortel's Succession 1000M had the best overall showing, failing over and back instantaneously to

FIGURE 2 Average One-Way Latency—Baseline And With Impairments



Notes:

(1) Impairments included 10 milliseconds of added latency in each direction.

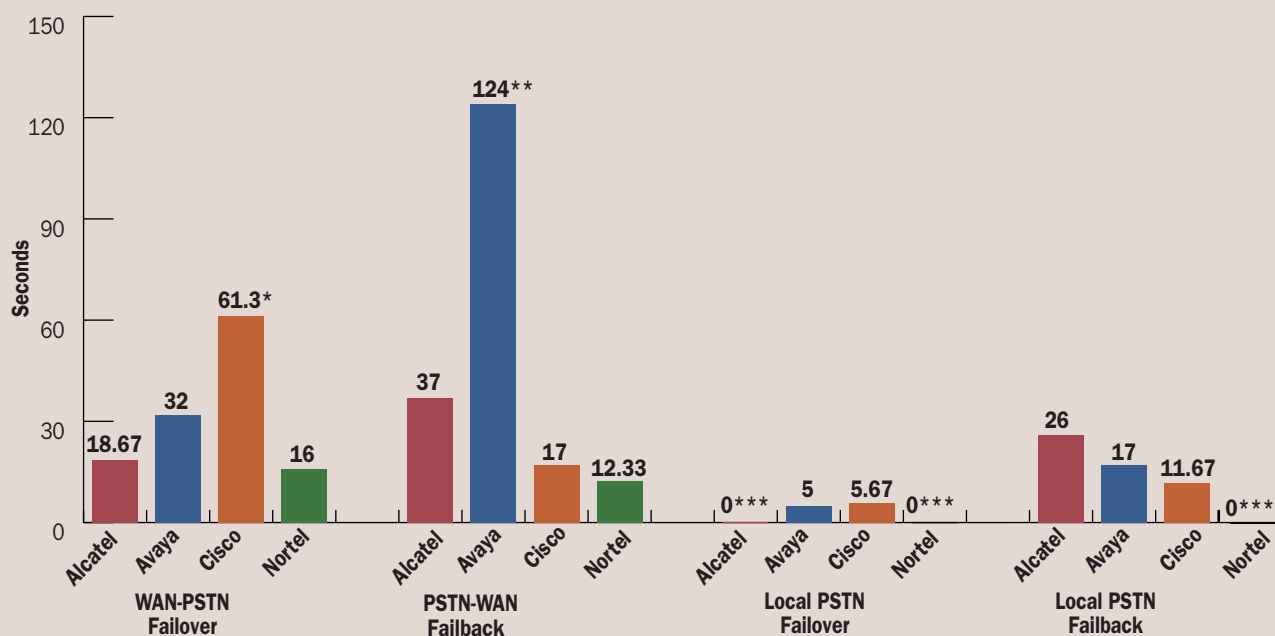
(2) Impairments included 30 milliseconds of added latency and 10 milliseconds of jitter.

(3) Impairments included 100 milliseconds of added latency in each direction, 1 percent packet loss (based on eight consecutive bit-error bursts) and 20 milliseconds of jitter.

After placing a VOIP call between two phones, we used a Telephone Handset Audio Tap-2 (THAT-2) adapter to inject a laptop-generated .wav file from the Line Out 1/8 stereo-jack connector of the laptop into the handset connection of the first phone. The left channel of the .wav file was diverted back into the "Line In" stereo-jack connector on the laptop and the right channel of the .wav file was transmitted across the phone connection to another THAT-2 box where the right channel was diverted back to the "Line In" jack. On the laptop, we ran Syntrillium Software Corp.'s Cool Edit 2000 audio recording, editing and mixing application for capturing and displaying both the left and right channels. We then measured the change between the two signals.

We measured one-way latency three times and averaged the results. All calls were placed over a wide area network (WAN). Impaired calls were run without any quality of service (QoS) enabled on the system under test.

FIGURE 3 Automatic Failover and Failback Times From IP Network To PSTN



Notes:

*This test was performed at the CallManager's default setting. Failover time on CallManager is configurable.

**This failback time did not affect any calls in progress.

***The "0" indicates an instantaneous failover/failback time.

To complete this test, the IP network was rendered unusable for placing calls, forcing a switchover to PSTN backup.

a local PSTN, and in 16 seconds (failover) and 12.33 seconds (failback) over a WAN. (Figure 3)

Avaya's S8700 had a high, 124-second failback time from the PSTN over the WAN to the IP network, but this did not affect any calls in progress.

We also tested the amount of time a call controller could fail over and back from a primary to secondary unit when a failure was simulated in the primary system. All the products completed failover in under a minute—with the Avaya S8700 posting the low time of eight seconds, and the Alcatel OmniPCX next with 19 seconds.

The Cisco CallManager and Nortel Succession failed over in 52.75 and 55 seconds, respectively.

Conclusion

As noted, these products are well matched and worthy of serious consideration for large networks. Which one you choose will depend heavily on the types of features, functions and performance you need in a particular environment.

Many of the features we evaluated in this review are in early or beta stages—some not yet released. Look for 2004 to be a pivotal year for IP-PBX features delivery.

Most new features we've seen are useful and productivity enhancing, but also portend some ethical questions and privacy issues. For example, being able to "find me/follow me" 24/7 could be good—and bad. Ultimately, companies will have to address these issues.

Security remains a sensitive issue. All the prod-

ucts were vulnerable to at least one of our DoS attacks. Look for IP-PBX vendors to address security more aggressively in the months ahead □

Note: Miercom would like to acknowledge the test tools and other network equipment used in this review: Adtran Atlas 800, Brix Networks 100, Extreme Networks Summit 48 switches, Hammer LoadBlaster from Empirix, Syntrillium Software Corp. Cool Edit 2000, JK Audio THAT-2 and the PacketStorm 1800E.

Companies Mentioned In This Article

- Alcatel-ND (www.alcatel.com/enterprise)
- Apache (www.apache.org)
- Avaya (www.avaya.com)
- Cisco (www.cisco.com)
- Dell (www.dell.com)
- Hewlett-Packard (www.hp.com)
- Microsoft (www.microsoft.com)
- Net6 (www.net6.com)
- Nortel Networks (www.nortelnetworks.com)
- Pingtel (www.pingtel.com)
- Polycom (www.polycom.com)
- PowerDsine (www.powerdsine.com)
- Symbol (www.symbol.com)
- Toshiba (www.toshiba.com)