

Small-Medium IP-PBXs Morph Into High End

Betsy Yocom, Jonathan Taylor, Diane Poletti-Metzel and Randy Birdsall

The line between high-end and small-to-mid-range (SME) IP-PBXs is blending.

The features, management, endpoints and security that have been available for some time on high-end IP PBXs are now largely available on small-medium enterprise systems (100–999 lines). The servers and controllers used in each segment may be different, but the management applications, endpoints, security features, basic telephone features and most advanced features are the same.

As we noted in last month's test of large IP-PBX systems (*BCR*, January 2004, pp. 28–40), voice quality is now virtually assured on all enterprise IP-PBXs. The advanced features and depth and breadth of endpoints distinguish the products.

These include a number of highly sophisticated unified communications offerings, such as handling voice mail through email, instant messaging blended with email, “find me/follow me” applications, the ability to set distinctive ring tones, etc.

Management applications on the SME systems tend to be easier to use than those at the higher end due to support of installation wizards to ease the configuration process. But security is still an issue, as it was on the high end. In this test as in

last month's, all the IP-PBXs proved vulnerable to at least one of our denial of service (DoS) attacks.

The Review

Last October, Miercom invited vendors that offer IP-PBXs supporting 100 to 999 stations to participate in a competitive review (results summarized in Table 1). Five vendors came in. An overview of their architectures and basic functions is shown in Table 2 (pp. 30–31), and features are summarized in Table 3 (pp. 32–33). Participants were:

■ **Alcatel**, which showed its Alcatel OmniPCX Enterprise Express, R5.1Lx; OmniVista 4760 management system v2.1; and e-Reflexes IP phones (the e-Reflex 4035 was the phone under test).



■ **Avaya**, which came in with the S8300 Media Server and G700 Media Gateway, version 2.0 (beta); Avaya Integrated Management v2.0; and Avaya IP telephones (the Avaya 4620 was the phone under test).

■ **Cisco**, which sent its CallManager (tested in this round on the 7825 server), version 3.3.3 with SR1; CallManager Administration management application v3.3.3; and Cisco IP Phones (the 7960G was the phone under test).

■ **Mitel Networks**, which tested its 3300 Integrated Communications Platform (IC) v4.1 (beta); 3300 Group Administration Tool, 3300 Desktop Tool and 3300 System Administration (all v4.1);

Betsy Yocom is managing editor at Miercom, a network consultancy and product test center based in Princeton Junction, NJ (www.mier.com). Jonathan Taylor is lead test engineer; Diane Poletti-Metzel is manager of lab testing and Randall Birdsall is test engineer.

TABLE 1 Scorecard: IP PBXs <1,000 Stations

	Percent Weighting	Alcatel e-ND OmniPCX	Avaya S8300/G700 	Cisco Systems CallManager 	Mitel 3300 ICP	Pingtel SIPxchange
Architecture	15	85	83	83	78	85
Endpoints	15	80	85	87	85	80
Management and Administration	20	82	85	88	85	75
Features	25	82	89	85	87	80
Security	10	75	86	85	81	84
Performance	15	84	89	90	80	83
The Bottom Line	100	82	86	86	83	81

and Mitel IP phones (the Mitel 5240 IP Appliance was the phone under test).

■ **Pingtel**, which submitted its SIPxchange, an entirely SIP-based, software-based IP-PBX, R2.2; SIPxchange Configuration Server, v2.2, which is incorporated into SIPxchange. Pingtel xpressa IP phones and Pingtel instant xpressa IP softphones were under test.

The Scorecard

Our testing ended in a tie for “Best-In-Test” between the Cisco CallManager and the Avaya S8300 Media Server and G700 Media Gateway.

In last month’s large-systems review, the Cisco CallManager beat the Avaya 8700 by just a tad. What moved the Avaya S8300 into the winner’s circle along with the CallManager in this review was a 100-percent call completion rate (the S8700 tested in the last review delivered “fine nines”) and support of local survivability on the S8300, which added points to the system’s score in the Architecture category.

But all the products tested had overall scores of 80 or above, reflecting that they rank well competitively.

We rated each of 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 telephone desk sets, softphones, 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.
- **Security**, including the ability to ward off DoS attacks, support of security features such as encryption or automatic VLAN assignment, as well as any other features that ameliorate system vulnerability.
- **Performance**, in which we assessed call quality, latency, call completion rates, ability to fail over quickly, etc.

Architecture

The Alcatel OmniPCX Enterprise Express and the Pingtel SIPxchange tied for first place in this category with scores of 85.

The OmniPCX Enterprise Express call controller, the Communications Server (e-CS), won points for its compact, one-unit form factor that integrates both H.323 and SIP VOIP protocols and an H.323 gatekeeper.

A single e-CS server supports up to 500 endpoints, which is the lowest maximum capacity among the products tested in this review. However, the e-CS may be easily upgraded to a higher end (IBM) server to accommodate greater capacity. Because features available via both servers are basically equivalent, end users don’t have to purchase a lot of extra applications to upgrade from one system to another.

Cards in the Alcatel IP Media Gateway (IPMG) multi-slot chassis are hot swappable—gaining Alcatel a few more points.

Alcatel’s call controller may be configured with a second, redundant hot-standby. The standby control server may be independent of the main controller, and this secondary controller may reside in another location. The call control servers are synchronized through the IP network; databases are, therefore, mirrored, so that either one can be active at any given time.

Pingtel’s SIPxchange was the only entirely software-based IP-PBX in this review. It is also the only product that supports SIP (Session Initiation Protocol) natively.

SIPxchange works with several different types of media gateways, and its Linux (Red Hat)-based call-control software will run on a variety of off-the-shelf hardware. According to Pingtel, SIPxchange has been certified to work with HP/Compaq, IBM, Dell and Sun machines.

The ability to use a variety of systems will be a definite plus for organizations that want to use existing hardware or don’t want to be locked into a particular vendor’s solution. The SIPxchange won some points for its flexibility in this regard. We tested the IP-PBX on both the Vegastream Vega 100 and the Cisco 5300 gateways (an older Cisco model that has been superseded by the Cisco 5350 and 5400).


Remote office survivability is available on the Pingtel system via redundant servers (one at headquarters and one at remote sites) connected by an IP/PSTN trunk gateway in each location. If IP network connectivity fails, a local proxy server routes calls from station to station at the remote location and provides failover to the local IP/PSTN gateway for trunk calls.

Over-the-counter SIPxchange server hardware can be purchased with redundant, hot-swappable components, including power supplies, fans and RAID drives to ensure full redundancy of critical components.

A major downside of the SIPxchange in terms of architecture is that it supports SIP phones (hard and softphones), analog phones only via FXS gateway and no digital telephones. This could be a limitation for end users that need to maintain their investments in existing equipment.

By the time you read this article, Pingtel will have announced its new pricing structure for SIPxchange, which marks a turning point in the IP-PBX industry.

Alcatel provided a compact form factor and good redundancy



**Pingtel plans
a major shift—
to an “open
source”
distribution model**

Sometime in mid-February, Pingtel will announce that it is offering its SIPxchange IP-PBX software as open source code to developers, which it hopes will accelerate technical developments in the SIP arena.

While the source code is free, Pingtel will “sell” customers various levels of software subscriptions (basic, standard and enterprise), providing stable software, software updates, certified testing, complete product documentation, installation and management tools, interoperability test results and other forms of support, depending upon the type of subscription purchased.

The price of \$344 per station shown in Table 2 (pp. 30–31), was based on the new pricing, which includes two SIPxchange IP-PBX subscriptions (at \$999 each), 400 Pingtel xpressa IP phones (at \$299 each), 100 Pingtel instant xpressa softphones at \$14.99 each and third-party hardware, including a server and trunk gateways.

Cisco and Avaya tied for second place in the Architecture category with scores of 83.

The Avaya S8300 won points for support of a local spare processor (LSP). S8300s can be configured in a redundant configuration with a primary call controller (called the Internal Call Controller, ICC) and a redundant LSP. The LSP backs up the ICC and assumes full control if there is a failure in the primary system. (Note: When S8300s are networked with S8700 or S8500 call controllers, the LSP can be used to back up those systems as well.)

Avaya also won points for integrating the S8300 and G700 Media Gateway into the same server, eliminating the need to house separate servers for each function.

With a capacity of 2,400 endpoints and 450 trunks, the Avaya S8300 is an ideal system for networks that might grow from mid-range to high end quickly.

The H.323-based S8300/G700 also supports a separate Avaya Converged Communication Server (a SIP proxy) that supports SIP interworking with third-party SIP products, such as the Pingtel SIP phones, and with Avaya’s own 4602 SIP phone.

Cisco’s CallManager supports a SIP trunk to a SIP proxy server, but the vendor said it plans to integrate a SIP proxy server into the CallManager in a future release.

The only difference between the CallManager tested in last month’s large IP-PBX review and this one was the capacity of its server. While the 7845 server supports a cluster of up to eight CallManagers in one discrete system, the 7825 server tested in this round supports up to four—with a maximum system capacity of 4,000 endpoints—the highest among the systems tested.

The ability to gracefully scale the Cisco system is a key architectural feature. CallManager also won points for its Survivable Remote Site Telephony (SRST) feature, 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.

A couple of downsides 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), and no redundancy on the 7825 server tested.

Key points for Mitel in this category are its compact form factor and scalability. While system capacity maxes out at 700 endpoints, multiple 3300 ICPs (up to 60 call controllers) can be connected in a cluster, and clusters may be connected for a capacity of 42,000 endpoints.

The ability to configure call controllers for failover and load sharing is a plus (multiple controllers may be configured to operate in hot stand-by mode or in concurrently active load-sharing), but there’s a major hitch in Mitel’s redundancy architecture. Mitel’s PSTN gateway *doesn’t* speak IP; instead, it’s designed only to convert signaling and bearer traffic from a Mitel proprietary remote messaging protocol to TDM for transit over the public network. So VOIP traffic coming from the LAN must first hit the call controller, which converts the traffic from IP to this proprietary protocol, then sends it to the gateway for the second round of conversion.

Here’s how this kludginess affects redundancy: The PSTN gateway only supports connection to a single call controller at a time, so if you deploy redundant controllers and the primary one fails, you have to physically disconnect it from the PSTN gateway, and hook up the secondary controller to the gateway. You could try to solve this problem by deploying a redundant call controller *and* a redundant gateway, but then if there’s a call controller failover, you’d still have to manually move cables—in this case, the PSTN connection, since each gateway supports only a single PSTN circuit connection. The only way to achieve “touchless” failover would be to have redundant PSTN links.

Mitel representatives told us they’re aware that this is an unwieldy configuration, and they plan to address this issue in 2005.

The 3300 ICP supports SIP, but it’s not integrated into the system. H.323-to-SIP gateways are optional and available from third parties.

Mitel’s native VOIP call control protocol is proprietary, as is its intra-system protocol, MiNet, which is used to deliver phone features between systems. Mitel said that MiNet can interoperate with other vendors’ devices, citing support for Symbol’s NetVision wireless phone. But heavy reliance on proprietary protocols will be a downside in this market over the longer term.

Endpoints

All of the vendors offer a full complement of offerings in this category—from basic IP phones

to sophisticated, touch-screen color models that provide stock quotes and weather reports.

Cisco took the lead in this category with a score of 87, thanks to the breadth and depth of its offerings, excellent IP phone design, and some useful software applications that facilitate use. But Avaya and Mitel also scored well, with ratings of 85, reflecting their broad IP product lines.

Cisco offers one of the widest arrays of IP phones—from the 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.

In addition to the endpoints, Cisco CallManager supports integral software enhancements including WebDialer, which allows Cisco IP phone users to place calls from both Web and desktop applications. Attendant console software that runs on a desktop machine can be used to set up speed-dial groups, personal directory, etc.

Avaya's product line includes seven models of IP phones from \$195 to \$900 and three models of wireless/cordless phones. All Avaya IP phones support a full-duplex speakerphone that handles up to six parties.

Avaya also offers the sophisticated 4630 Screenphone—a color, touch-screen model with 120 speed dials, a “click to dial” feature and one-touch dialing. This phone now incorporates a 320 × 240 pixels window, a switch for PC connection and a wireless headset.

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.

Mitel also offers seven IP phone models, ranging in price from \$110 to \$650, plus a Symbol Netvision wireless phone. In addition to the 5200 Series IP phones, Mitel offers a unique iPAQ cradle and software application for docking a Compaq iPAQ directly into a 5230 IP Appliance (more on this under Features) and the 5303 Analog Conference Unit.

Mitel's digital display is a well-sized 320 × 240 pixels, but the vendor currently offers only black and white or orange and black displays, no color models.

Pingtel's SIPxchange will work with any SIP-based phone, and the vendor also offers its own Pingtel xpressa IP phones. We tested the SIPxchange with xpressa phones and also Cisco 7960 and 7940 IP SIP phones. Pingtel also supports a full-duplex speakerphone with excellent, “echo free” voice quality.

The xpressa phone is well designed, incorporating a 160 × 160 pixels display area, 11 soft keys, and a knob on the side for scrolling through the LCD menu.

At the time we tested, Pingtel offered the PX-1 xpressa phone that cost \$240, but this model was to be discontinued by the end of 2003 and

replaced by the PX-2 xpressa with an integrated 10/100 Ethernet switch. This was scheduled for introduction in the beginning of 2004 at a price of \$299 (not including a connection license fee).

Pingtel lost some points in this category due to its limited scope of offerings relative to the competition. As mentioned, the SIPxchange doesn't support digital phones, and analog phone support requires a third-party FXS gateway, such as the Mediatrix 1102, 1104 and 1124.

Also, Pingtel does not currently support an IP conference phone, a video application, which some of its competitors in this market do, and there are no call statistics on the xpressa phone.

Alcatel's score of 80 in this category reflects a solid endpoint product line including three models of Advanced e-Reflexes IP phones, ranging in price from \$195 to \$485. Alcatel supports third-party SIP phones, including Pingtel's xpressa.

A unique offering from Alcatel is IP Plugware, through which users with Alcatel digital phones can upgrade to an IP phone to maintain their investment in existing equipment.

Alcatel's product line does not include conference phones, but IP and analog Polycom sets are sold via Alcatel channel partners.

One of the trends that we've noted in both this and last month's large IP-PBX review is the increasing market acceptance of softphones. What's driving this development is the increase in teleworking and the many powerful new softphone enhancements that combine multiple communications applications into a modular, adjustable Windows-based interface—making communication between the home- and office-based worker far easier to maintain.

All of the IP-PBXs in this review support a softphone—some in more advanced stages of development than others.

We saw Cisco's new IP Softphone in beta; it was scheduled for general availability in late 2003 or early 2004. This new product includes Cisco's Skinny protocol, which allows the softphone to support all basic features found on a Cisco IP desk phone.

Alcatel's My Phone softphone is a well-designed, intuitive softphone application with both Windows-based and Web-based client options. Powerful features such as the ability to place a call via name or number, customize the phone set and integrate with Internet Explorer, are available via the product.

The Avaya IP Softphone integrates with Microsoft IE, which allows a number of useful features, such as the ability to pop up contact information on the screen about the caller while the call is in progress. It also allows recording of inbound and outbound call information in a journal that can be viewed later.

The Avaya IP Softphone also supports instant messaging in which all conversations are encrypted. And it offers the ability to “scrape” a Web page



Cisco offers a wide array of IP phones

TABLE 2 System Architectures Compared—IP PBXs <1,000 Stations

	Alcatel e-ND Calabasas, CA www.alcatel.com/enterprise	Avaya Inc. Basking Ridge, NJ www.avaya.com	Cisco Systems, Inc. San Jose, CA www.cisco.com
System tested (version, general availability)	Alcatel OmniPCX Enterprise Express, R5.1Lx; GA in 10/03	S8300 Media Server with Local Survivable Processor and G700 Media Gateway, v2.0 (beta); GA in 12/03	Cisco CallManger, v3.3.3 with SR1; GA in 8/03
Call control server	Communications Server (e-CS) running on Alcatel hardware	S8300 Media Server running on dedicated appliance	CallManager running on Media Convergence Server (MCS) 7825
Call control OS and processor	Linux; Intel/AMD processors	Hardened Linux Red Hat 8.0; Intel processors	Windows 2000 Server for MCS 7825
Other key call-switching components	IP Media Gateway (IPMG) multi-slot chassis for PSTN trunking and digital and analog station interfaces	G700 Media Gateway multi-slot chassis for PSTN trunking and digital/analog station interfaces	Media gateway functions (for T1 and PRI trunking) via module on Cisco 3725 switch
H.323 gatekeeper	Integrated in Communication Server (can be turned on or off); also supported externally	Integrated in the S8300	Cisco Multimedia Conference Manager (MCM) supports gatekeeper functions
H.323-to-SIP gateway support	Yes, integral	Yes	Yes (in release 4.0)
Management system	OmniVista 4760, v2.1 (extra priced at \$17,699 plus \$4,000 for server); 4760i Web-based configuration tool; OmniPCX Installation Wizard; Web Monitoring Tool GUI	Avaya Integrated Management, v2.0 (beta) includes Avaya Site Administration (ASA) and Integrated Management Solution with Avaya Installation Wizard, VOIP Monitoring Manager, Avaya Terminal Configurator (part of enhanced package that costs \$20,000), Native Configuration Manager, Maintenance Web Interface and Upgrade Wizard	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
Vocoders supported	G.711, G.729A, G.723.1; VAD on G.729A and G.723.1	G.711, G.729 A&B, G.723.1	G.711, G.729, G.723, wideband (IP phone to IP phone); VAD support on all codecs
VOIP protocols supported	Native call control: IP; Intra-system: proprietary encapsulated in H.323 stack; IPMG based on MEGACO, 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	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
Station interfaces	Analog, digital, IP, wireless	Analog, digital, IP, wireless	Analog, IP, wireless
Maximum overall station capacity*	500	2,400	4,000 in a CallManager cluster of four in load sharing configuration
Maximum capacity, all IP stations	500 if all IP phones; 500 if all IP softphones	2,400 (all IP phones)	4,000 IP phones in CallManager cluster; 2,000 if all softphones equally balanced among servers
Maximum number T1/E1 trunks and analog trunks*	24 T1/E1; 176 analog	450 T1/E1 or 450 analog	800 T1/E1 or 800 analog in CallManager cluster
IP phone tested (version, release date)	Alcatel Advanced e-Reflexes 4035; R3.0, GA in 10/02	Avaya 4620 IP Telephone; v2.0, GA in 12/03	Cisco IP Phone 7960G, v5.0(3); GA in 10/03
Softphone(s) supported	Alcatel My Phone softphone; Alcatel 4059 Multimedia Attendant Console	Avaya IP Softphone; Avaya Softconsole; Avaya IP Agent call center	Cisco IP Softphone; Cisco IP Communicator
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; Pingtel SIP phones	SCCP-based phones from Sanyo, ADTech, SpectraLink, Symbol, IP Blue, Telstrat and Polycom
Cost per station (U.S. list price)**	\$611	\$395	\$630

Mitel Networks Kanata, Ontario www.mitel.com	Pingtel Corp. Woburn, MA www.pingtel.com
3300 Integrated Communications Platform (ICP), R4.1 (beta); GA in 12/03	Pingtel SIPxchange, R2.2, GA 11/10/03
Mitel 3300 ICP Controllers (one remote and two at headquarters)	IP PBX is fully SIP-compliant and software based
Proprietary call control application runs on top of VxWorks OS	Linux Red Hat 7.3 running on HP/Compaq, IBM, Dell and Sun servers
3300 Universal Network Service Units for T1/E1 trunking; 3300 Universal Analog Service Unit (ASU)	System tested with VegaStream 100 and Cisco 5300 gateways
Yes, via an optional device available from VCON	Not applicable; SIPxchange uses SIP proxy server
Yes, via optional gateways	No
3300 Group Administration Tool, 3300 Desktop Tool and 3300 System Administration Tool, 3300 OPS Manager, all R4.1 6110 Contact Center Management included in price of 6110 Contact Center application (\$520 per agent)	SIPxchange Configuration Server, v2.2 (with integrated Linux utilities for configuration changes); settings for T1 trunks done via associated gateway
G.711, G.729A	G.711, GIPS enhanced G.711 and G.729A
Native call control: proprietary; Intrasystem: MiNet; Intersystem: IP MSDN, and DPNSS	All SIP
Analog, digital, IP, wireless (Symbol NetVision)	IP; SIP to analog gateways that are fully SIP compliant
700	Limiting factor is BHCC per call server; however, system has been tested with up to 1,000 endpoints
700	Same as above
384 T1/480 E1; 16 analog trunks per ASU with maximum of 768 analog trunks via expansion cabinets	Test with up to 250 DSOs
Mitel 5240 IP Appliance, R4.1; GA in 12/03	Pingtel xpressa IP phones; Pingtel instant xpressa IP softphone; R 2.1.11; GA in 8/03
Mitel Your Assistant Pro and IP Console	Pingtel instant xpressa IP softphone
Symbol Netvision wireless	Cisco 7960 and 7940 SIP phones; any SIP-compliant phone***
\$498	\$344

Notes:

*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 small-to-medium sized enterprise configuration of a system supporting 500 stations (consisting of 400 mid-range IP phones, each with a secondary switch port, at least six feature buttons, an LCD display and 100 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.

***Note that some IP-PBX features may be lost if the phone does not support a full suite of SIP Service Examples as standardized by the IETF.

to highlight a phone number through which a call can be placed directly from the page.

Mitel's Your Assistant Professional (Pro) soft-phone is a sophisticated application that helped

the vendor tie for second place in this category. Significantly, in addition to integrated email and voice mail, integration with Microsoft Outlook and Exchange and Lotus Notes is supported.

Table 3 Features Compared—IP PBXs <1,000 Stations

	Alcatel OmniPCX Enterprise	Avaya S8300 Media Server/G700 Media Gateway	Cisco Systems CallManager
Percent of basic features supported*	100	94**	100
Automated alternate routing	Optional via My Assistant in OmniTouch Unified Communications	Integral	Integral
Automatic route selection	Integral	Integral	Integral
Automated attendant	Integral, but requires license fee	Integral (can be IVR and DTMF; additional server adds speech recognition)	Integral
Hunt groups	Integral	Integral	Integral
Least cost routing	Integral, but requires license fee	Integral	Integral (via AAR Groups)
Night service (time of day)	Integral, but requires license fee	Integral	Integral
Recorded announcements	Integral, but requires license fee	Integral	In 4.0 release
System speed dialing	Integral	Integral	Integral
User directory	Integral with access to LDAP directories	Proprietary, but can be synchronized with LDAP and other databases; Avaya IP Softphone can interact with MS Outlook contact database or external LDAP server	Integral (via embedded LDAP directory)
Billing/accounting	Optional within OmniVista 4760	Optional third party	Optional via third party
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
Call Center/ACD support	Integral, but requires license fee	Integral with license fee	Optional via Cisco IPCC Express Edition
911/E-911	Optional via third party	Integral	Optional via Cisco Emergency Responder
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
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
Other add-on applications, special features	Integrated wireless support; third-party integration for 802.11x wireless phones	Audio conferencing; Multimedia Contact Centers; EC500 Extension to Cellular	Cisco Personal Assistant, Standard XML applications, Cisco WebDialer

*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" in "Large IP-PBX's: A Well Matched Quartet," *BCR*, January 2004.

**Avaya and Mitel each lost some points for the Missed Call Indicator feature. They both support this in a call log, but do not offer any visual indication of a missed call on the phone.

***At the time we tested, music on hold was not supported; however, it should be available by the time this article publishes.

Pingtel's instant xpressa IP Soft-phone rounds out the offerings in this area. The product is supported on Windows 98, 2000 and XP systems, but since it is built on a cross-platform development foundation, it could be readily adapted to support Linux, MacOS and Solaris.

Perhaps one of the most significant things about the instant xpressa is its price—only \$59. When Pingtel's new pricing structure goes into effect, the instant expressa will be only \$14.99 within Pingtel's new subscription pricing.

Mitel Networks 3300 ICP	Pingtel SIPxchange
94**	81***
Integral	Integral
Integral	Integral
Optional	Integral
Integral	Integral
Integral	Planned for 2004
Integral	Integral; requires manual switchover
Integral	Not supported
Integral	Integral
Integral; Telephone Directory updated by an LDAP-compliant directory via OPS Manager	Integral, but requires corporate LDAP server
Optional via third party	Integral (by YE 2003)
Integral in 3300 ICP, but optional licenses required to activate service	Integral
Optional	Not supported
Integral	911 supported; no E911
Optional	Integral
Optional via 6510 Unified Messenger	Integral (email notification of voice mail)
5700 Voice First Video Conferencing, Integrated 802.11 Wireless Gateway; Your Assistant and Your Assistant Professional, 6500 speech enablement platform, 6510 unified messaging	Dialing by SIP URL, MS Outlook integration with auto-speakerphone; custom ring tones

Management And Administration

As noted earlier, products in the SME IP-PBX segment are beginning to look much like their high-end counterparts. Especially in the management area, the features and functions available on IP-PBXs targeted for both market segments are basically identical.

Cisco's CallManager took management gold in last month's large IP-PBX review and maintained its position in this one. The Web-based CallManager Administration application, which is integrated into CallManager, is still the most well organized, intuitive, menu-driven and easiest to use among the products tested.

A key factor in CallManager Administration's win in this category is the availability of several plug-in management tools that come with the price of the system.


These tools include a Bulk Administration Tool (BAT) that allows bulk transactions, such as adding, updating or deleting a large number of phones, users and ports; 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 telephones.

An extra-priced application (\$19,995), IP Telephony Environment Monitor (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.

Avaya maintained its second-place ranking in the management area with a score of 85, although in this review it shares this position with Mitel, whose 3300 ICP management tools are also well designed and intuitive.

Avaya's S8300 is managed via the Windows-based Avaya Site Administration (ASA) for PBX and telephony applications, and the VOIP Monitoring Manager for active calls, station status of endpoints, etc. The VOIP Monitoring Manager is part of Avaya's Integrated Management Solution that includes the Avaya Installation Wizard and Upgrade Wizard, as well as Native Configuration Manager and a Maintenance Web Interface.

The Avaya Installation Wizard is an especially useful application that facilitates a straightforward, intuitive installation process. Through a series of wizards, installers are prompted to add system parameters as determined on a "pre-installation"



Avaya offers cool features on a hot screen phone

worksheet. Embedded, online help addresses questions.

While Avaya's management application is rich, it comes with a lot of different pieces, so the learning curve to master them all will take a while. But Avaya's management comes with some good animated tutorials, online and context-sensitive help for all "task" tab items. The mouse-over help available in the installation wizards is excellent.

Mitel's 3300 ICP management application is entirely Web-browser based. It consists of four basic applications: the 3300 Desktop tool through which the end user can manage his or her own personal configuration; the 3300 Group Administration tool, used to complete moves, adds and changes, etc.; the 3300 System Administration Tool for configuring the IP-PBX itself; and the 3300 OPS Manager for monitoring the network.

We found the 3300 ICP management application well designed and fairly easy to maneuver. Notably, in the System Administration application we had the option of viewing the tree of parameters both logically and alphabetically—the latter being an option that helps one find configurable parameters more readily than drilling down through a long list of options.

Making moves, adds and changes via the Group Administration tool was straightforward, and we liked the fact that this function was separated from the System Administration Tool. Separating the more difficult system admin functions from moves, adds and changes makes sense. Less-technical staff can perform the latter without fear of seriously affecting the system itself.

In Group Administration, templates are supported for adding users. We just pulled one up and filled in the blanks. A plug-in to Active Directory allowed us to create an end user account directly from this database.

The 3300 ICP management suite includes an excellent real-time monitoring tool, called IP Phone Analyzer, that provides statistics on packet loss, jitter, codec, etc. on a per-call basis. However, this information is only displayed once a call has ended. It would be great to view these stats while the call is in progress.

One criticism of the Mitel application: Some of the terminology used in the parameter tree is obtuse—for example, referring to a PBX as a "network element" and a codec/vocoder as a compressor.

Alcatel's management application is the OmniVista 4760, which is an extra-priced option (\$17,699 plus \$4,000 for the server). Included within OmniVista 4760 is 4760i, a Web-based configuration tool that facilitates moves, adds and changes; an OmniPCX Wizard that simplifies the installation process for initial configurations (for more on this application, see the Features section below); and a Web Monitoring Tool, which provides a GUI to access real-time information on system parameters.

OmniVista supports a very comprehensive directory tree structure that provides many options for configuring tasks and features. But we found it easy to lose our way in this management application and think a better organization of the directory tree, and the ability to view it alphabetically, would be helpful.

Alcatel's OmniVista is not as intuitive as most of its competitors. In several instances, we had to drill way down into the system to find information or make changes. The application also supports multiple interfaces, so we had to go to different places to perform various tasks. This jumping around hampers navigability.

Pingtel's management application, SIPxchange Configuration Server, supports a logical design, was easy to navigate and includes informative, context-sensitive help.

But Pingtel lost points in this category for lacking many things its competitors support and end users will expect to have on an IP-PBX management system that's deployed in a mid-sized enterprise network.

This includes the lack of real-time statistics, no call-detail reporting, no searchable online help, no filtering for alarms or traps, blank default fields and lack of central management of multiple devices.

Features

Voice quality, support for basic telephony features and traditional PBX system features and add-ons are now widely ensured on IP-PBXs (Table 3). The focus has, therefore, shifted to features that provide productivity enhancements and unified communications.

These new features include the ability to get voice mails via email, collaborate with colleagues in real time through a softphone, set up phone conferences via a desktop phone, participate in videoconferences on a laptop and work more efficiently from home.

It's in the area of "advanced" features that all of the vendors displayed the most innovative and useful offerings, and these features edged Avaya's S8300/G700 into first place in this category with a score of 89.

Most of the features supported on the Alcatel, Avaya and Cisco high-end IP PBXs we reviewed last month are also available on the mid-range systems reviewed here.

What we've highlighted below are features that the vendor showed us in this round of testing. But scoring in this category was based on these features, as well as those covered in the previous review.

Avaya's high-end 4630 IP Screen Phone is a standout. It includes a touch-tone color screen, 120 speed dials and a smart dialing feature that eliminates the need to add prefixes, such as a 1 or 9, to phone numbers. The phone also supports a screen lock for security.



Mitel offers unique full multiparty video conferencing

Through a Net6 server that strips down banners and other unnecessary information from a Web page, Web content can be viewed on the 4630 screen. Users can access stock quotes, weather and travel information, etc. They can also view call statistics, such as codec in use, packet loss, jitter, etc., and access a call log.

Another useful feature on the S8300 is Extension to Cellular 500 (EC500), which works with a digital cell phone from any supplier and may also be used with an analog phone in a home office.

With EC500, calls ring through on a user's cell phone and desktop phone, and even if answered on the cell phone, will appear to the caller as if they were answered via the desk phone. Calls placed from the cell phone will also appear as if they are being made from the desktop.

If the call is answered on the cell phone and one wants to continue the conversation on his/her office phone, the call may be extended to the desktop phone by pressing a line appearance button on the office phone to continue the conversation.

To prevent others from bridging onto calls at the desktop phone and listening in to conversations taken via the cell phone, Avaya provides an "Exclusion" feature, which disables the capability for shifting calls back and forth between a given desktop and outside phones. The Exclusion feature can either be invoked automatically by the network administrator, or it can be enabled/disabled on a per-call basis, via a feature button on the desktop phone or by dialing a specific code, called a Feature Name Extension (FNE), from the outside phone.

Avaya also showed us its 3626 and 3616 WiFi handsets that work with the S8300. These products are multi-line sets with three extensions that work with any access point that supports the 802.11b standard. The 3626 also can be used as a walkie-talkie providing four hours of talk time and 80 hours of standby.

Mitel's 3300 ICP also made a strong showing in the Features category, garnering a score of 87.

Mitel's Your Assistant Professional (YA Pro) softphone application is a major highlight of the vendor's product line. It combines multiple desktop communications applications into a modular, user-adjustable Windows interface.

YA Pro supports on-demand (*ad hoc*) conferencing. Calls are placed automatically and concurrently to all members in the desired group. A bridged conference is underway as soon as users answer the phone. (This feature will be fully integrated into YA Pro and generally available in release 3.0.)

As noted in the Endpoints section, YA Pro is the only softphone application we've seen to date that supports full multiparty videoconferencing (up to four end-user sites can be displayed together on a screen), which when we reviewed it was fully functioning but not fully integrated in YA Pro. It will be in release 3.0. Setting up a video-

conference uses the same process as establishing an audio conference—eliminating any major learning curve to use the feature.

Mitel's 6150 Multimedia Contact Center Solution is another powerful feature. It combines voice mail, email, fax and Web chat routing, reporting and real-time monitoring (through the 6110 Contact Center Management application) to handle customer contacts consistently across any medium in which they choose to communicate.

The 6150 integrates with Microsoft Exchange 2000 and Microsoft Outlook to provide a number of capabilities. These include automated routing of contacts to the longest-idle agent and a unique ticket number assignment to each incoming and outgoing contact that links the customer's entire email history together in one file. This information "pops" on the screen when a call comes into the center.

Mitel's 6110 Contact Center Management software application provides a rich set of real-time monitoring tools to track agent events within a 6150 Multimedia Contact Center. This extra-priced option (\$520 per agent up 100 agents) provides detailed reports on contact center activity. More than 160 canned reports are supported; these can be generated on a specific agent's performance for a specific time period, groups of agents, etc. and displayed in colored pie and bar charts, tables and spreadsheets.

Events in the call center can also be monitored in real time, and visual and audible alarms are available to alert supervisors of critical events.

Mitel also won points for its 5230 IP Appliance integration with PDAs through a docking cradle on the phone. Users can transfer their personal PDA profiles and preferences to any 5230 IP Appliance on the corporate network by simply dropping their PDA into the docking cradle.

Once the PDA is docked, applications available via the corporate IP network are available through the 5230, including visual call transfers, visual conferencing, visual call forwarding, etc., dialing by contact name, visual voice mail, 150 feature keys, speed dialing, etc.

Cisco placed third in the features category with a score of 85. The vendor demonstrated several advanced features available now, as well as others that will be delivered in CallManager's next release, which was not generally available when we tested the product.

Cisco's Unity unified messaging (we evaluated version 4.0, Build 3), resides on a separate Cisco MCS 7845I server, offering a tight integration with Microsoft Exchange. This allows voice mail to be read in Outlook with all voice mail and email messages in the same mailbox. Urgent messages are marked with an audible alert, and we could respond to any message via phone or laptop.

Unity provides the ability to listen to a whole message or just a summary. Also, a text-to-speech engine is available as an option.



As in last month's tests, the current crop showed security vulnerabilities

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.

Many models of Cisco IP Phones (7970, 7960G, 7905G and 7912G) support a 133 × 65 pixel-based LCD display and standard XML applications. Pressing a special button on any of these phones initiates a menu of services available through the phone, including conference room scheduler, email and voice mail messages list, daily and weekly schedule and appointments, personal address book entries, weather reports, company information, stock information, etc.

A software developers' kit is available to format and deliver content to the phone by providing Web server components for LDAP directory access, Web proxy and graphics conversion. It also contains sample applications that show how to use the XML tags that the phone supports.

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.

Many of Alcatel's advanced features are incorporated in its OmniTouch Unified Communications application, some of which, such as MyPhone softphone application and MyMessaging software-based unified messaging, weren't due for availability until this month. Alcatel lost some points for this tardiness relative to its competitors, who were already delivering some of the things OmniTouch had yet to offer.

The OmniTouch suite is based on SIP, XML and vXML applications. It includes the My Phone softphone that supports 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 user-customizable Web page.

All of the OmniTouch features can be ported to a PDA, and eventually to a cell phone, according to Alcatel.

One of the features unique to the mid-range OmniPCX Enterprise Express is the free OmniPCX Wizard for initial system installation. It is available to Alcatel's business partners through the OmniVista 4760i Web interface.

The Wizard provides a step-by-step process for installing the OmniPCX. It includes pull-down menus that facilitate configuration, and a template is available to create system hardware, such as the media gateway. The Wizard also provides a screen for configuring a dialing plan, changing phone profiles, setting up voice mail and user lists, etc.

Once a configuration is complete, it is downloaded to the OmniPCX server. At that time, errors show up in red so that they are easily identified. These must be corrected before the system accepts the configuration.

Alcatel also showed us its OmniTouch Contact Center Distributor/Contact Center Supervisor

(CCD/CCS), a subset of its full-blown contact center application that is targeted to mid-range networks. CCD is the contact center software itself, and CCS is the client software used to manage the system.

This application shows a graphic of which agents are set up to answer incoming calls and the order in which calls should be handled. The supervisor can add and remove groups from the graphic and review the status of individual agents (whether they are available, on a call, placing a call, etc.)

Statistics available on contact center activity can be ported to an Excel spreadsheet. Graphic reports are also available in pie charts.

With a score of 80, the Pingtel SIPxchange brings up the rear in this category, but not because it doesn't offer some good features. This score reflects a lack of breadth in this area relative to the other products reviewed.

One of Pingtel's most outstanding features is actually a performance enhancement—support for Enhanced G.711. This allowed SIPxchange to achieve an excellent 4.5 interactive voice quality score on our "impaired" Internet test. (See Performance for more details on this test.)

Enhanced G.711 (eG.711) is a proprietary codec supplied by Global IP Sounds (GIPS), based in Stockholm. eG.711 provides standard, transparent coding of PCM/G.711 when no packets are lost. However, when packet loss occurs, the eG.711 codec recodes the G.711 packets to reduce the packet loss and, thus, improve overall voice quality on a degraded link.

eG.711 works with another GIPS technology, called NetEQ, which provides additional packet-loss replacement functions to preserve toll-quality voice at packet loss rates up to 10 percent and allow only very small voice quality degradation at loss rates up to 30 percent. NetEQ also provides enhanced automatic adaptive jitter management to reduce latency.

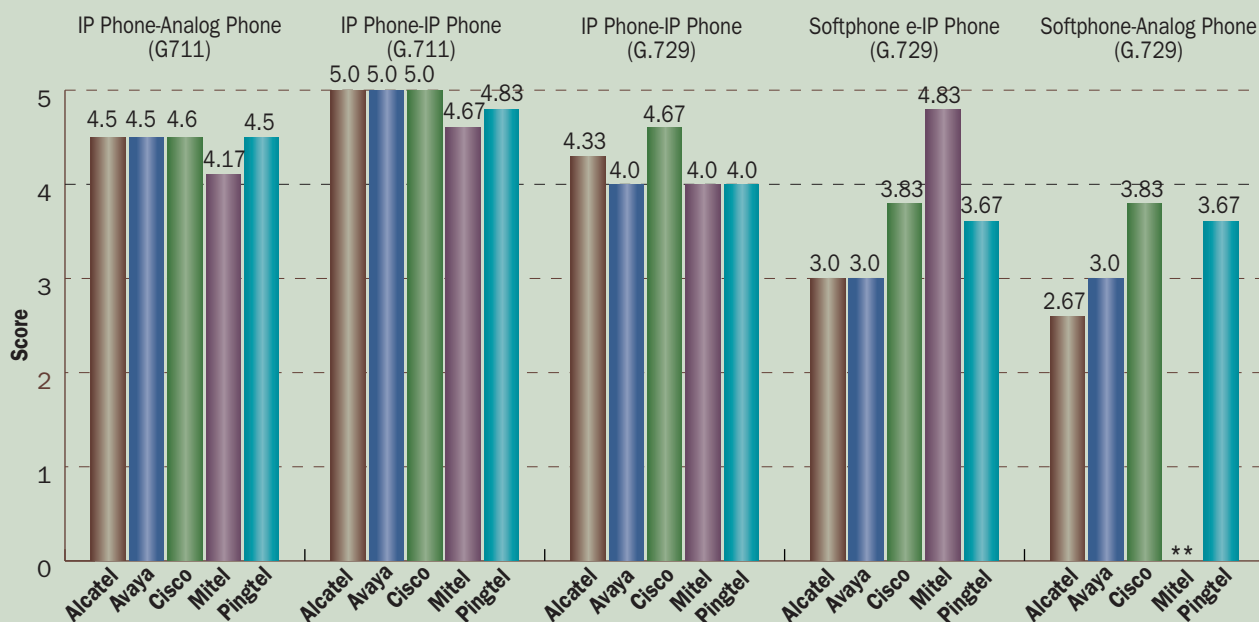
On another note, one of SIPxchange's unique features is the ability to allow users to configure their own distinctive ring tones, including various animal noises, the "007" theme song, a Big Ben ring, etc.

Pingtel also supports user dialing by SIP URL, as well as by telephone number. For example, a user at ext. 2400 may also be reachable at jsmith@remote.corp.com.

SIPxchange works with a centralized LDAP server to provide phone-book applications that allow users to look up and quickly connect to other employees in the directory.

Pingtel planned to release an enhancement to this feature last December that will start the directory application with a search screen. Once a contact is selected, the user can drill down into the LDAP information store to receive more information about that contact, including a photo, if available.

FIGURE 1 Interactive Voice-Quality Ratings*



* In the Interactive voice-quality tests three engineers, in rotation, conducted real-time, two-way conversations over separate connections. We ran five tests; all were conducted over a WAN. We performed countdown tests to measure the effects of latency and alphabet tests to examine bi-directionality. We noted any echoes, clipping, background noises or other impairments that affected clarity. Scores were given on a five-point scale with a score of 4.0 and above rated as "toll quality," suitable for all telephony applications.

** We were unable to obtain this result because we could not get the Mitel system to operate with our test tool.

Security

Avaya maintained its lead in this category with a score of 86, 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 also supports standard AES 128-bit encryption.

Tests showed that encryption had no effect on Avaya's voice quality. 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.

Avaya won points for temporarily locking accounts after too many failed attempts. Avaya also provides three levels of administrative access to the S8300/G700, including normal user and super users. Each person logging into admin services can be restricted from performing certain commands.

Mitel's 3300 ICP supports encryption currently only on its 5220 IP phones and 6010 Teleworker Gateway. This support is based on secure RTP using a CAST-128 algorithm. Key exchange takes place through a call control signaling channel, which is also encrypted via CAST-128.

Cisco is delivering AES 128-bit encryption in its next (v4.0) release of CallManager, which will apply to both the media stream and call control. The call controller will generate 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, and the CallManager will authenticate each call to avoid "man in the middle" attacks.

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

Alcatel's score was affected by the OmniPCX's lack of SSL and SSH support. In addition, the vendor had no plans to offer encryption. However, on the plus side, the OmniPCX supports automatic VLAN assignment of its phones.

Security on the SIPxchange communications server is provided through SIP-standard authentication. On a per-call basis, the communications server decides whether the call requires authentication; most authentications, according to Pingtel, are limited to calls through the PSTN gateway.

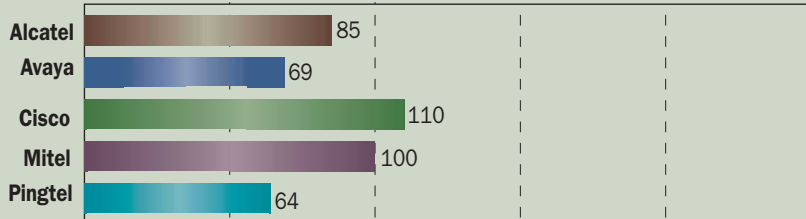
In this round of tests, as in the large-system testing, all the systems were susceptible to at least one type of denial of service (DoS) attack.

One attack caused the Avaya S8300 phones to lose dial tone, and the media gateway to re-register. On the Alcatel OmniPCX, one type of DoS attack caused the IP Media Gateway to reboot. On the Cisco CallManager one attack caused phones at the remote site to reboot, and voice quality was choppy at the far end. And on the Mitel 3300 ICP, one attack caused a loss of dial tone and phone re-registration.

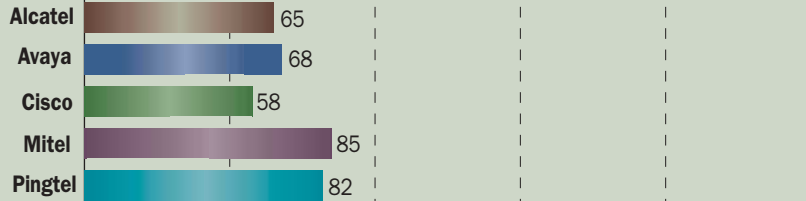
While DoS attacks did not cause Pingtel's SIPxchange or xpressa phones to reboot, the phones were very slow to respond during some

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

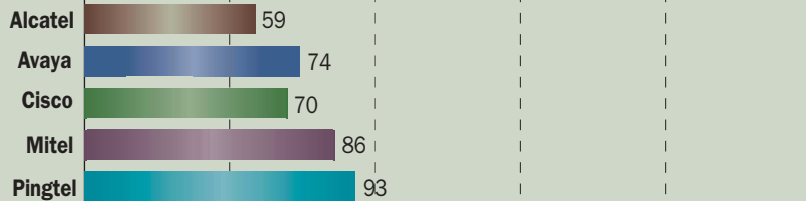
Baseline IP Phone-to-Analog Phone (G.711)



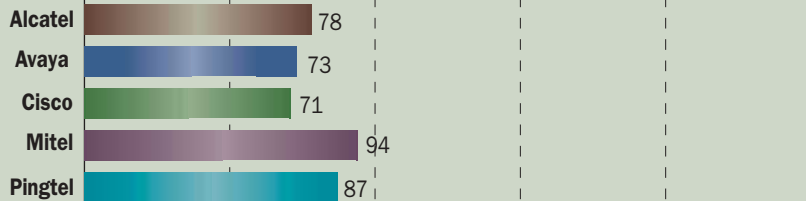
Baseline IP Phone-to-IP Phone (G.711)



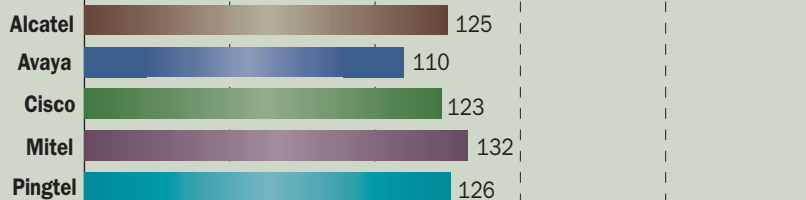
Baseline IP Phone-to-IP Phone (G.729)



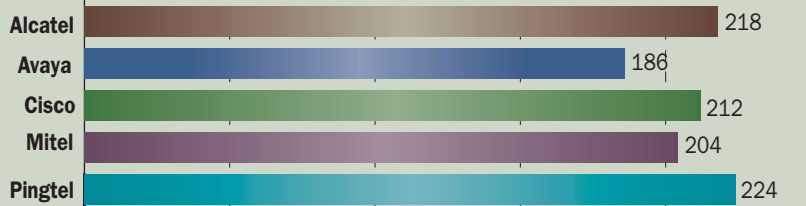
Campus IP Phone-to-IP Phone (G.711)



Intranet IP Phone-to-IP Phone (G.729)



Intranet IP Phone-to-IP Phone (G.729)



Average Latency in Milliseconds

- 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.

attacks. Voice quality was reduced significantly, and calls in progress were delayed or impaired.

Performance

The Cisco CallManager was the top-ranking performer in this review with a score of 90. CallManager had the best interactive voice quality scores, beating its competitors in all but one test—in which it had the second-best rating. (Figure 1)

The CallManager also delivered a 100-percent call completion rate and relatively low latency metrics in all but the “impaired network” test, in which none of the products performed well.

What pushed Avaya up in this category—helping the S8300 to tie with CallManager for first place—was its 100-percent call completion rate, which the vendor did not achieve in our tests of large systems. Furthermore, the S8300 also impressed us with a 100-percent call completion on our “bonus” round in which we doubled the number of calls the product had to complete within our nine-hour test.

(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.)

The Pingtel SIPxchange won points in Performance for its 100-percent call completion. The Alcatel OmniPCX had a 99.9984 percent rate, and the Mitel 3300 ICP posted a 99.9503 percent completion rate.

Mitel’s showing in call completion was negatively affected by the generation of system status reports that occurs nightly at 11 p.m. These HTML pages run at an elevated priority that interfered with call-processing. Mitel has since lowered the priority on the task, so the call-completion rate should be higher in the next release.

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

We’ve found that voice quality on a softphone is often affected by the underlying hardware—laptop, type of headset used, etc.—something to take into consideration when deploying these devices.

On one-way measurements conducted over the WAN between IP phones, latency on all the products remained below 100 milliseconds (ms) on baseline tests (Figure 2).

On our impairment tests, in which we added latency, jitter and packet loss, latency increased gradually as the impairment level increased. In the first “campus” test between IP phones, we introduced 10 ms of added latency, but latency metrics remained below 100 ms.

When we added 30 ms of latency and 10 ms of jitter, all systems posted metrics above 100 ms, but a spot check of voice quality showed that scores were still around 3.0, which is acceptable.

On our third test, we 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 increased substantially, with only the Avaya S8300 coming in under 200 ms. At latencies of 150 ms and higher, voice quality begins to degrade substantially.

We also conducted call control failover tests to determine how long it took the system to switch from a primary to a backup controller. The SIPxchange was the clear winner here with an instantaneous failover. Alcatel had the second-best time—failover in 21 seconds. Cisco’s controller failed over in 57 seconds and the Mitel 3300 ICP in one minute and three seconds—both acceptable times for this type of equipment.

However, Avaya’s failover time of 2 minutes and 40 seconds was on the high side. The time was extended because the S8300 had to place the failed IP trunk into maintenance mode before failover to the backup unit occurred. Avaya is addressing this issue in a future release.

Conclusion

The IP-PBX market is undergoing a shift in which products targeted to both medium and large enterprise networks are offering most of the same features, endpoints, equivalent performance and management applications.

The only major difference between these two groups is in the capacity of the underlying call server. Therefore, moving to a higher-end server is often all that’s necessary to scale the system.

The split in this market is now between enterprise IP-PBXs and those designed for the 99 stations-and-under market. We’ll examine the latter in our next review.

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, Compaq ML-350 Servers, HP Compaq laptops, the 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 Systems (www.cisco.com)
Compaq (www.compaq.com)
Dell (www.dell.com)
Global IP Sound (www.globalipsound.com)
Hewlett-Packard (www.hp.com)
Microsoft (www.microsoft.com)
Net6 (www.net6.com)
Pingtel (www.pingtel.com)
Polycom (www.polycom.com)
Symbol (www.symbol.com)
Toshiba (www.toshiba.com)

To scale from medium to large systems, all you may need is a bigger server