IP-PBXs: Ready And Waiting

Edwin E. Mier, Kenneth M. Percy and Kevin D. Brown

Next-gen enterprise voice systems are bigger, better and more numerous than they were a year ago.

We had a packed house for this year's *BCR* testing of IP telephony systems. There are lots of products out there, and the vendor community clearly expects 2002 to see sharply accelerating demand and deployment. Maybe.

We can't speak to the state of the economy, but we can attest to the readiness of the technology. Test slots were filled on a first-come, first-served basis, and the results of 10 tested IP-PBX systems from eight leading vendors are included here:

- Alcatel e-ND submitted the latest release, 4.2, of its OmniPCX 4400.
- Avaya submitted two systems, which share many of the same IP components and all of the same station equipment: the mid-range IP600 Communications Server, running release 9.5 software; and a Definity G3, model SI, running the new release 10 software.
- Cisco brought in the latest version, 3.1(1), of its AVVID/CallManager IP Telephony System.
- Mitel Networks delivered two systems: its low-end 3100 Integrated Communications Platform (ICP), version 3.0; and its mid-range 3300 ICP, also version 3.0. These two systems employ and support all the same station equipment.
- Shoreline Communications sent us version 3.1 of its Shoreline3 package.
- Siemens Enterprise Networks Division submitted version 1.0 of its Hicom 150 H IP Convergence Platform.
- 3Com delivered the latest rendition, version 4.0.1, of what it now calls the 3Com SuperStack 3 NBX.
- Vertical Networks brought in version 4.0 of its InstantOffice 6000, which, while still a lower-end offering, now supports up to 180 stations.

Trends

Besides the sheer number of vendors and products now on the scene with viable IP-telephony systems, notable developments were observed in a number of areas:

- **Architecture:** There's still a sharp schism between the "IP-oriented" systems and those consisting of "IP-enabled" upgrades to erstwhile TDM/digital PBXs. The IP-oriented camp includes Cisco, Shoreline and 3Com. The remainder—mostly vendors with large installed bases of legacy TDM/digital systems—includes Alcatel, Avaya, Mitel, Siemens and Vertical.

Other architectural elements of note: System capacities have nearly doubled for several vendors since a year ago, and "in-line power" is now generally provided—which is the way we believe it should be.

- **Standards:** Not a lot of good news here to report. Besides a standard, Layer-3 IP transport, almost all VOIP call control today, in all the systems tested, involves some proprietary protocol twists, despite the availability of several industry standards.
- **Management/administration:** Improvements in this area are seen across the board. Many of the latest developments in IP telephony management and administration—and especially their impact on moves, adds and changes—will be discussed in detail in a special feature appearing in next month's issue (*BCR*, February 2002).
- **Applications and features:** We asked vendors what IP telephony uniquely delivers that classical TDM/circuit-switched telephony does not. The answers we got highlight new and distributed applications and features, which are designed to exploit the geographic insensitivity and data orientation of the underlying IP infrastructure.
- **Performance:** Generally excellent and, based on this year's review, getting even better—with a few notable exceptions. Also, among the new twists added to our performance testing this year was our deliberate launching of several Denial-of-Service attacks against these systems' call controllers and IP phones. The results weren't as severe as we feared.

Bigger And Bigger

Table 1 (pp. 30–31) provides a thumbnail view of the 10 systems we tested. Although there are many points of comparison, the major distinguishing factor is size—the maximum station capacity per system. We define a "system" as one which handles all stations under a common call control. The high-end systems handle more than 1,000 stations, mid-range from about 200 to 1,000 stations, and low-end models handle fewer than 200 when fully expanded.

Cisco, Alcatel, Shoreline and Avaya compete at the high end

Cisco's 10,000-IP-phone fully expanded system capacity remains the same as a year ago, although the vendor hints at plans to increase that by an order of magnitude—to 100,000. No real details yet.

Alcatel's OmniPCX 4400 handles up to 5,000 total stations, or a max of 4,000 IP stations per system. The vendor touts its ability to network multiple systems up to a capacity of 50,000 stations, or up to 100 discrete call servers (systems). However, note that any call between IP phones on different distributed systems currently must be processed through Alcatel's TDM/digital switching fabric, which adds latency and affects end-to-end call quality.

Shoreline is also touting 5,000-station capacity, a sizable increase over its 3,000-station capacity of a year ago. Note, however, that each T1 trunk reduces the total stations supported by 24—i.e., minus 24 stations for each T1 trunk. Also, Shoreline remains one of the few "IP telephony" vendors that does not yet support IP phones.

Rounding out the high end is Avaya, with its ubiquitous Definity system. We tested a low-end model, a G3 SI, with all the latest IP appurtenances. It supported 2,400 analog or digital stations, but a maximum of just 1,000 IP stations. Higher-end Definity systems handle more stations, but IP-station support remains a fraction of the analog/digital-station capacity.

The mid-range offerings include 3Com's SuperStack 3 NBX, featuring a more powerful VxWorks-based call controller than the system boasted a year ago. As a result, 3Com now handles up to 750 IP stations. However, as in the case of Shoreline, each trunk channel reduces this number. So with a 4-to-1 station-to-trunk ratio, the maximum station capacity drops to 600. (Shoreline's station capacity would drop from 5,000 to 4,000 with a similar station-to-trunk ratio.)

The mid-range also includes: Mitel's 3300 ICP, capable of handling 700 IP stations; and Avaya's IP600 Communications Server, with a stated max IP-station capacity of 450.

TABLE 1 IP-PBX Configurations Compared

Vendor	Alcatel e-ND	Avaya, Inc.	Avaya, Inc.	Cisco Systems	Mitel Networks
HQ city, state	Calabasas, CA	Basking Ridge, NJ	Basking Ridge, NJ	San Jose, CA	Kanata, Ont., Canada
Web URL	www.alcatel com	www.avaya com	www.avaya com	www.cisco. com	www.mitel. com
System tested, software release/ version	OmniPCX 4400, Rel 4.2	Definity G3 SI, Rel 10	IP600 Communications Server, Rel 9.5	AVVID IP Telephony System, CallManager Ver 3.1(1)	3100 Integrated Communications Platform, Ver 1.0.21
Max stations per system	5,000; 4,000 if all IP	2,400; 1,000 if all IP	450 IP; 312 analog/digital	10,000	34; 24 IP and 10 analog
Call control	Unix module in chassis; or standalone Linux server	Proprietary card in chassis	Windows NT module in chassis	Windows 2000 standalone server(s)	VxWorks, integral to system controller
Analog phones	√	√	√	√	√
Fax, modem	√	√	√	√	√
IP phones; line-powered	√ √	√ √	√ √	√ √	√ √
Digital phones	√	√	√	No	No
Wireless	√,PWT	√	√	No	No
T1/E1 trunks	√, CAS/PRI	√, CAS/ PRI	√, CAS/PRI	√, CAS/PRI	No
Analog trunks	√	√	√	√	√
Redundancy support	Hot fail-over call controllers; load sharing, redundant IP controllers	Call-control and IP/ LAN-control cards can all be hot fail-over	IP/LAN controller cards can be hot fail-over	Load-shared, fail-over call controllers, remote-site fail-over	None
Price per station (US List), for 'typical' config	$791, based on a 200-stn configuration	$1,032, based on a 196-stn configuration	$683, based on a 196-stn configuration	$768, based on a 200-stn configuration	$410, based on a 34-stn configuration
What this varied configuration includes; total cost, based on U.S. List prices`	96 IP phones, varied models, $470 to $595 each; 104 analog phones ($50 apiece), voicemail and mgt:	100 IP phones, varied models; 96 analog; Audix voicemail and mgmt;$202,313	100 IP phones, varied models, 96 analog; integral voicemail and mgmt; $133,859	104 IP phones, varied models, 96 analog; all L2 switches; voicemail and mgmt: $153,620	24 IP phones (model 5010); 10 analog phones, L2 switch; voicemail and mgmt: $13,940

The low-end category of the systems tested includes: Siemens Hicom 150 H IP Convergence Platform, handling up to 144 IP stations (though up to 250 if all are analog and/or digital); Vertical Networks' InstantOffice 6000, which also doesn't support IP phones but now handles up to 180 analog; and Mitel's super-low-end 3100 ICP, which does up to 24 IP stations, plus another 10 analog. In Vertical's case, the 180 analog stations, or up to 168 digital phones, is a doubling of last year's 84-station max capacity.

Two other configuration-related observations: First, the larger-capacity systems generally offer more redundancy and hot-standby fail-over options. That's understandable, since phone service to more people is affected if these systems' call control were to fail without a redundant backup. Secondly, the lower-end systems are more likely to include as integral certain key features like voice mail and auto-attendant capabilities. This is the case with both Mitel and the Avaya IP600. With the larger, high-end systems, it seems that everything is an add-on, extra-priced option.

Empowering IP Phones

Another architectural aspect of IP telephony involves delivering power to IP phones. And the news here is generally good.

As of a year ago, the most prevalent method for delivering power to IP phones was a local transformer/adapter, the type used to power laptop computers. But bulky local adapters take up precious space on the power strip, and also render phone communications subject to power outages.

Significant progress has been made in this arena with the products we tested this year. In fact, all the vendors whose IP-telephony system supports IP phones can provide "in-line" delivery of power over the Ethernet LAN cabling and into the phone via the ubiquitous RJ-45 connection. This puts IP phone sets on a par with digital and analog phones, both of which have long been powered in-line.

There's good news on line-powering of IP phones

Mitel Networks	Shoreline Communications, Inc.	Siemens Enterprise Networks Division	3Com Corp.	Vertical Networks Inc..
Kanata, Ont., Canada	Sunnyvale, CA	Boca Raton, FL	Santa Clara, CA	Sunnyvale, CA
www.mitel. com	www. goshoreline.com	www.siemens enterprise.com	www.3com.com	www.vertical. com
3300 Integrated Communications Platform, Ver 3.0	Shoreline3, Ver 3.1.7700, (late beta-release, Nov, 2001)	Hicom 150 H IP Convergence Platform,` Ver 1.0	3Com SuperStack 3 NBX, Ver 4.0.1	InstantOffice 6000, Ver 4.0
700	5,000 (less trunk channels)	250; 144 if all IP	600 (with 4-to-1 stns-to- trunks)	180 analog, 168 digital
VxWorks, integral to system controller	VxWorks, integral to system controller	Proprietary, integral to system controller	VxWorks, integral to system controller	Windows NT server, embedded in chassis
√	√	√	√	√
√	√	√	Fax only, no modem	√
√ √	No	√ √	√ √	No
No	No	√	No	√
√ Symbol IP	No	No	No	No
√, CAS/PRI	√, CAS/PRI	√, CAS/PRI	√, CAS/PRI	√, CAS/PRI
√	√	√	√	√
None	Call control replicated across all modules; synchronized	Can auto route around failed trunks, if provisioned redundantly	Redundant auto-fail-over IP links on all major nodes	Redundant power; disk mirroring for NT call controller
$450, based on a 96-stn configuration	$846, based on a 192-stn configuration	$540, based on a 100-stn configuration	$687, based on a 200-stn configuration	$685, based on a 96-stn configuration
48 IP phones, varied models, 48 analog; voicemail and mgmt: $43,175 $158,214	192 high-end analog phones ($150 each list); voicemail, and mgmt: $162,393	48 IP phones (Omnipoint 400); 52 analog phones, voicemail, and mgmt: $53,975	100 IP phones; 100 analog phones with adapters; voicemail, and mgmt: $137,382	96 digital phones, models, voicemail and mgmt: $65,740

The standards picture is complicated

Power-over-Ethernet is delivered in one of two ways: There's Cisco's way, and then there's the way everybody else does it. Cisco delivers low-voltage DC power over active data pairs, a technique referred to as "phantom" power. In the configuration we tested, Cisco delivered phantom power out to its IP phones directly from Catalyst 3524 switches.

The method everybody else uses is to carry power over unused wire pairs—that's conductors 4 and 5, and 7 and 8, out of an 8-conductor run of Cat 5 cabling. This approach is specified in the IEEE 802.3af draft standard, which most expect will be finalized soon. The vendors we tested who powered their IP phones this way—Alcatel, Mitel, Siemens and 3Com—employed either a unit they manufactured or, more commonly, a "power hub" unit OEMed from PowerDsine, which now dominates this power-supply marketplace niche.

Costs for the in-line power sources were included in the typical system prices shown in Table 1. The only exception was Avaya, whose power devices were not yet shipping at the time of testing. Individual plug-in adapters powered Avaya's IP phones in the configuration tested, and the Avaya systems were priced accordingly (adding $20 per power adapter per IP phone).

Besides Cisco, Mitel Networks, with its low-end 3100 system, also provided in-line power directly from an Ethernet switch device. For its higher-end 3300 system, though, Mitel used the power hubs, which sit between the patch panel and the data switch and "inject" power onto the Cat 5 cabling and out to the IP phone. Alcatel, Siemens and 3Com also tested with power hubs. Alcatel's power hub is proprietary; the other three vendors OEM their devices.

Standards...What Standards?

Some of the systems tested embrace VOIP industry standards—most notably the ITU's bulbous H.323 umbrella standard—more than others. But that is not to say that the better systems are fully standards-based; it's not that simple or straightforward.

TABLE 2 VOIP Support By IP-PBX Systems

Vendor	Alcatel OmniPCX 4400, Rel 4.2	Avaya Definity G3 SI, Rel 10	Avaya IP600 Comms Server, Rel 9.5	Cisco AVVID IP Telephony System, CallManager Ver 3.1(1)	Mitel 3100 Integrated Comms Platform, Ver 1.0.21
Max IP stns per system	4,000 IP	1,000 IP	450 IP	10,000 IP	24 IP
IP phones	Yes, module snaps onto digital sets	Yes	Yes	Yes	Yes
How IP-to-PSTN calls processed	Via TDM switching fabric	Via TDM switching fabric	Via TDM switching fabric	Via VOIP-T1 or VOIP analog gateways	Via TDM switching fabric
G.711 for local VOIP	Yes	Yes	Yes	Yes	Yes
Other, low bit-rate vocoder(s)	G.729a, G.723.1	G.729a, G.723.1	G.729a, G.723.1	G.729	None
VAD compression supported	Yes, for all supported vocoders	Yes, for G.729a only	Yes, for G.729a only	Yes, for all supported vocoders	No
Native VOIP call control protocol(s)	Proprietary (INT/IP)	H.323v2 with H.225 security	H.323v2 with H.225 security	Proprietary (SCCP 'Skinny')	Proprietary
IP trunking between distributed systems (GK= Gatekeeper)	Yes, proprietary (ABC/Q.SIG over IP); all calls via digital TDM fabric	Yes, H.323 GK-GK; proprietary signaling extensions (DCS/Q.SIG)	Yes, H.323 GK-GK; proprietary signaling extensions (DCS/Q.SIG)	Yes, via H.323 GK-GK connection	No
H.323 gatekeeper (GK) support	Yes, integral; for 3rd party-H.323 support, also supports external GK's (incl Cisco)	Yes, integral; used for all VOIP communications	Yes, integral; used for all VOIP communications	Yes, integral; for 3rd-party H.323 connect and between CallManager clusters	No
Other VOIP protocols supported	MGCP planned (2003)	None	None	MGCP, to Cisco IOS gateways; IP phones can support SIP	None
VOIP prioritization	TOS, DiffServ, 802.1p/q	TOS, DiffServ, 802.1p/q, UDP port range	TOS, DiffServ, 802.1p/q, UDP port range	TOS, DiffServ, MPLS	TOS, DiffServ

As indicated in Table 2, Avaya has probably embraced and adopted H.323 more fully and formally than any other vendor in this test. Internal call control is fully via H.323 gatekeepers, as are IP calls between systems over IP trunks. But, as of the time Avaya was tested, the vendor acknowledged that there still were no third-party H.323-based products that could interoperate with Avaya's. This is because Avaya also adopted another related standard, H.225, in its implementation, which provides impressive added security—authentication, encryption, etc. But few other H.323-supporting vendors now support H.225-based security as part of their H.323 implementation. Hence, very limited third-party interoperability prospects.

Most enterprises have a lot to learn about VOIP standards, and especially what "support for standards" really means. Call control is one major role of VOIP standards, but another discrete aspect is the delivery of features to IP phones. And it is in this second realm that virtually all IP-telephony vendors still march to the beat of their own drum.

Even Avaya employs proprietary protocol twists—notably running over and above H.323—for the implementation of certain phone features over IP. This is because the VOIP standards define how to implement some—a dozen or so—fairly basic phone features. But legacy vendors like Avaya, Alcatel, Mitel and Siemens have hundreds of phone features that they had to extend out to their IP phones, so they could operate on a par with their digital phone set predecessors.

As Table 2 shows, vendors have implemented H.323 to various degrees and in various ways. In general, there is still no interoperability today between the IP phones of one IP-telephony vendor and the central processor or call controller of any other.

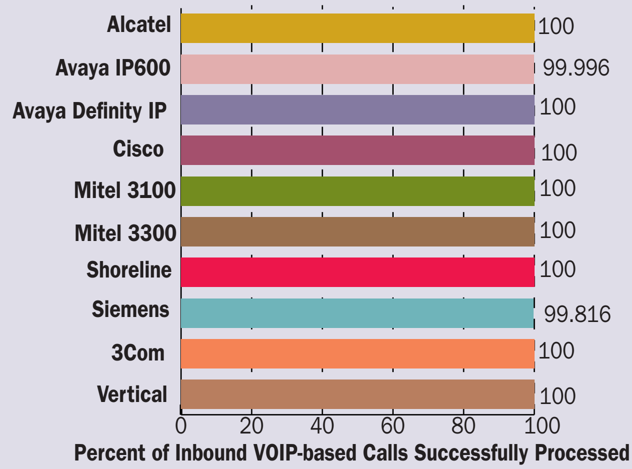
There is no multivendor interoperability between IP phones and call controllers

IP-Enabled Features And Applications

An often-cited justification for moving to IP telephony and VOIP is that new, value-added features

Mitel 3300 Integrated Comms Platform, Ver 3.0	Shoreline Shoreline3, Ver 3.1.77 (late beta-release)	Siemens Hicom 150 H IP Convergence Platform Ver. 1.0	3Com SuperStack 3 NBX, Ver 4.0.1	Vertical InstantOffice 6000, Ver 4.0
700 IP	No IP; up to 5,000 analog	144 IP	750 IP, less trunk channels	No IP; up to 180 analog
Yes	No, analog stations only	Yes	Yes	No; analog and digital stations
Via TDM switching fabric	Via VoIP-T1 or VOIP-analog gateways	Via TDM switching fabric	Via VOIP-T1 or VOIP-analog gateways	IP trunking only; no VOIP to/from stations
Yes	Yes	Yes	Yes	VOIP only over IP trunk
G.729a, G.723, G.726	G.729a, G.726	G.723.1	G.723.1, G.726	G.729a, G.723.1
Yes, for low bit-rate vocoders	No	No	Yes, for all supported vocoders	Yes, for all supported vocoders
Proprietary	Proprietary (DCCP)	Proprietary (CorNet-over-IP)	Proprietary (H3; runs at Layer 2)	H.323v2 or MGCP
Yes, proprietary (MSDN IP); all calls via digital TDM fabric	Modules within system all connect via IP (up to max 5,000 stns, less trunks)	Yes, proprietary (CorNet/IP)	Proprietary (H3 over IP); max 24 concurrent IP-trunk channels	Yes, via H.323 or MGCP; max 8 concurrent IP-trunk channels
No	No	No	Optional standalone H.323 gateway; works with external gatekeeper	Supports external GK; works with several (including Cisco)
Opt H.323 gateway, for Symbol IP-wireless phones	None	Low-level H.323, for MS NetMeeting connectivity	H.323, on IP trunks, for 3rd-party connectivity	MGCP, for connectivity to external MGCP controller
TOS, DiffServ	None	TOS, DiffServ, 802.1p/q	TOS, DiffServ, 802.1p/q	TOS, Diff-Serv; other, router-based prioritization mechanisms

FIGURE 1 Reliability: Inbound IP Call-Completion Rate (1)



(1) Tests were all run continuously for at least 12 hours.

A Hammer/Empirix call generator (a Load Blaster 500 system) delivers 12 concurrent calls into each IP-PBX system via a T1 trunk, at a rate of two calls per channel, per minute (1,440 call attempts per hour).

After ringing twice at an IP phone (and recording that as a successful result), the call generator would drop the call, wait 15 seconds, and then place another call to the same station.

Exceptions were: the Mitel 3100, which supports just eight analog trunk channels (T1 is not supported); Shoreline, which supports only analog phone sets (although all calls are VOIP-processed); and Vertical, which supports just eight concurrent VOIP channels (calls were delivered to digital phone sets; all calls, however, were VOIP-processed).

and applications are made possible. To quantify this claim, we asked vendors to detail the unique and special features enabled by their IP-telephony platform. Their responses highlighted features and capabilities in three general categories:

- New PC-based utilities, such as softphones.
- Enhanced user mobility.
- Network- and server-based applications, ranging from unified email/voice mail messaging, to new voice-enabled systems, such as a directory database.

Accentuating the emergence of the PC as a PBX station or client, every vendor exhibited some form of PC-based application. These ranged from softphones or software-based attendant consoles to browser-based user-configuration utilities offered by Cisco and Mitel. All of the IP-PBX vendors offered software supporting SMTP-based voice mail and/or "unified messaging," which we define as the consolidation of email and voice mail behind a common interface.

Some PC applications were especially noteworthy. Mitel's 5550 IP Console, accompanied by a special keypad and headset, effectively morphs a PC into an attendant console. Siemens' very slick Attendant P Office displays "busy lamps" on the GUI for off-hook extensions.

PC-based contact-center utilities include Avaya's IP Agent, which lets call-center agents work remotely from any PC with high-speed Internet access. Vertical's InstantOffice lets you browse into the InstantOffice Contact Center, a

Review Of Systems Tested

Alcatel OmniPCX 4400: The system is now undergoing an architectural transition—away from its TDM-based past and towards a fully distributed, pure-IP future. Based on our lab testing of the latest 4.2 release, the OmniPCX retains some of the best aspects of its prior life—broad feature support, excellent reliability and top voice quality, to name a few—even as it moves to become a full IP telephony system.

Currently, the OmniPCX employs proprietary protocols to make everything work and to retain full backward feature and signaling support. But the grand plan, the vendor says, is to evolve call control and signaling between its distributed IP-PBX modules to be purely Media Gateway Control Protocol (MGCP) standards-based—around 2003.

As part of its evolution to IP, call control, which is at the heart of the system, is being extracted out of the current OmniPCX multi-slot chassis and run in a standalone Linux-based server. We tested an early model, called the Enterprise Call Server. With call control moved out of the chassis, Alcatel is renaming what remains the Alcatel Media Gateway, since its main role in an IP environment is to convert voice streams between VOIP and non-VOIP legacy connections (analog, T1 trunks, digital phone sets, etc.)

The most laudable aspects of the OmniPCX 4400 include scalability to 4,000 IP stations per system (up to 50,000 in a network of systems), low latency and excellent intra-system VOIP voice quality. Various other options are supported for third-party IP connectivity. For example, the system can assume the role of a full H.323 gatekeeper, or connect as a client to a third-party's gatekeeper. And there are several other notable IP capabilities, like a very slick softphone and software-based attendant console, plus the ability to automatically invoke the right VOIP vocoder, which determines how much bandwidth is needed, depending on whether the IP call is local or remote.

Drawbacks? Alcatel's IP phones currently consist of modules that snap onto existing digital phone sets. Several pure-IP phone models are in the works, but they're not here yet. Also, while Alcatel is continually enhancing its management interface, it remains fairly arcane and a lot less than intuitive to use.

Avaya's Definity IP: The IP-enabling of the ubiquitous Definity is fairly well thought-out and effective. The low-end Definity G3 SI—which supports up to 1,000 IP phones and is Avaya's VOIP retrofit of the Definity—works well. Voice-quality ratings, based on Avaya's 4600-series IP phone sets, were among the best of all those tested, with very acceptable latencies. In addition, the benefit of Definity's decades of software development has been extended to the IP environment; all the features we asked about were fully supported.

The Definity is sturdy: Not a single call attempt failed during our reliability tests. And both redundant chassis and individual board-level components failed over successfully small-office contact center with a big-office look. Then there's Mitel's 6110, a Web-based contact-center system, which allows call-center supervisors to work from anywhere.

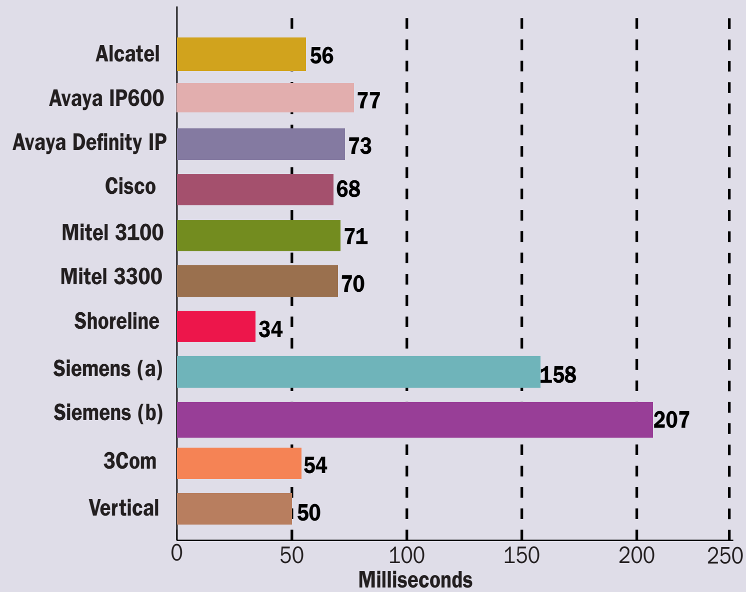
Alcatel, Avaya, Mitel and 3Com all offer sophisticated softphone applications. The Alcatel Soft Phone 4980 serves as both a softphone and as a third-party call controller. In the case of road warriors, for example, the application allows full access to IP-PBX features from a remote PC, while using a POTS phone for voice connectivity. Other mobility-minded features included Avaya's EC500, a "follow-me" feature, and Cisco's Extension Mobility, the AVVID system's enhanced free-seating capability.

Some vendors showed new server-based applications designed to exploit the IP convergence and distance-insensitivity aspects of IP telephony. Cisco and Mitel, for example, separately brought impressive, server-based voice-enabled applications. Mitel's voice-enabled directory, for example, lets callers navigate an auto-attendant-type directory system by speaking naturally—an improvement over listening to "press 2 for sales, press 3 for tech support," etc.

Solid Performers

A number of new and modified performance metrics were applied in this year's IP-PBX testing, as shown in Figure 1. Due to the varied architectures of IP-telephony systems, we wanted to develop a test which we would run for at least 12 hours, and

FIGURE 2 One-way Latency: Local Connections, G.711 (1)



(1) Tests were all run between IP phones over the 10/100 LAN using G.711 vocoding.

Exceptions were: Shoreline, which supports only analog phone sets (although all calls are VOIP-processed); and Vertical, which supports analog and digital phone sets, but not IP (Vertical's latency was measured between its digital phones, across an IP/VoIP trunk)

Siemens was tested with two separate jitter buffer settings: short (a), which requires a special manual adjustment; and normal (b), which is the default setting on the system.

—for the most part. Fail-over of one key IP module—the C-LAN card, which processes all IP calls—was less than elegant. IP phones registered to the active C-LAN continued ringing after the card was intentionally failed, until the redundant card took over—more than five minutes later.

How Avaya has architected its IP on the Definity is worth noting. Laudably, the VOIP implementation is perhaps the most comprehensively standards-based of any IP-enabled product we reviewed. Call control on Avaya IP-enabled systems occurs via H.323 v2 for both local and remote IP connections—and between systems on a full, formal H.323 gatekeeper basis.

Among the features Definity supports, several are especially noteworthy. The IP Agent is a PC-based application that enables mobility for call-center agents. Any PC running IP Agent can connect back to the call-center PBX and receive calls at any location. Then there's the EC500, a software option that is essentially a "follow-me" bridge that allows simultaneous ringing of both a station phone and any user-defined end-device—cell phone, pager, home phone, etc. Avaya's IP Softphone R3 is a PC-based, GUI version of an Avaya digital phone. It fully emulates the genuine article, with full feature support and directory access.

The venerable Definity has its blemishes, though, too. You'll learn how to use Avaya's management interface eventually—but it'll take time. Another small issue: some beta code caused phones to fail when G.729 vocoding was enabled with VAD compression, requiring reset of the phone. And speaking of VAD, it is supported on the Definity only for G.729 encoding. The last lament is price. At better than $1,000 per station, it was the most expensive system tested.

Avaya IP600: The Avaya IP600 Communications Server is designed to be a lower-priced, smaller-capacity "Definity" for branch offices or small businesses, but delivering all the same features and functionalities. The exact same IP modules can be used either with the Definity or the IP600; they're architected identically, and support all the same station devices and features. They are even managed via the same management interface. Like the Definity, the IP600 supported 100 percent of the features we looked for, and also registered solid voice-quality ratings. Not surprisingly, latencies were nearly identical.

However, the system was not quite as reliable as the Definity. When interrupted in mid-call setup, one module that should have been fully redundant wound up resetting the whole chassis and the IP phones. We also experienced problems in getting a T1 with robbed-bit signaling (RBS) to work cleanly on an IP600 trunk. PRI signaling on the T1 worked like a charm, however.

In understanding how the two systems could behave slightly differently under the same circumstances, consider this: The IP600 runs on Windows NT, while Definity runs on a Unix-based call-control platform.

As for special features, the IP600 supports virtually all the same features as the Definity. However, some which would gauge relative IP-based call completion rates among the different systems, given a moderate call load. We collaborated with Empirix and jointly developed such a custom test, which employs and runs on Empirix's Hammer test system.

The reliability test involves calls delivered to the IP-telephony system over 12 channels of a T1 trunk, at a rate of 24 call attempts per minute, or 1,440 calls per hour. The system-under-test had to promptly and correctly route each call to an IP phone (except for Shoreline and Vertical, which don't support IP phones, but which sent VOIP-processed calls to analog and digital stations, respectively). The target phone would ring twice, and then the Hammer call generator would drop the call, wait 15 seconds and re-dial the same IP phone.

As Figure 1 shows, the call-completion rates were perfect or near perfect, in all cases except for Siemens' Hicom 150 H. In Siemens' case roughly two calls out of each 1,000 were not successfully completed.

Our performance tests also measured one-way, end-to-end latency for each system, for various "environments," involving different vocoders, voice-activity-detection (VAD) compression, connections and station equipment. Figure 2 shows the one-way latency measured between IP phones connected over a 10/100 LAN and using high-bandwidth G.711 vocoding. Note that the test con-

capabilities integral to the IP600 are extra-priced options with Definity. Natively resident within the IP600, for example, is all the contact center functionality. This allows the IP600 to be either a freestanding contact center or a remote extension of a larger one.

While not priced as steeply as the Definity ($683 per station with the IP600) and not quite as rock stable, the IP600 shares a few of the limitations of its big brother. Voice activity detection (VAD) is supported only for the G.729a vocoding, for example. The management interface, as noted with Definity, takes some getting used to.

Cisco AVVID: Cisco's crusade to convince enterprise decision-makers to buy into IP telephony carries with it the onus of delivering a viable, large-scale IP-based alternative to the traditional TDM-based PBX. Cisco also must now contend with competitors' VOIP-based products, which are proliferating. This year's edition of the AVVID IP Telephony System suggests that Cisco is holding up quite well in both arenas, with the big stories being steady performance, improved feature support and new applications.

Performance tests on the AVVID/CallManager system yielded solid results. Voice quality was notably above toll quality in all scenarios tested. Latencies were well within acceptable limits. No hitches occurred in any of the reliability tests, and all redundant components failed over as expected, without incident.

Cisco also has shored up AVVID's feature set, registering support for 91 percent of the 32 telephony features that we looked for in this year's testing. While still shy of some competitors, Cisco is narrowing the gap.

Cisco's modular architecture allows new features to be added in a number of ways. One example is the new Survivable Remote Site Telephony (SRST), an IOS software-based feature that Cisco demonstrated on a Catalyst 4224 switch/router. Designed for branch offices, SRST functions as a local back-up call processor for phones located at a remote office, in the event that the IP connection back to the main Windows 2000-based CallManager should go on the fritz.

Another example of add-on functionality is the Personal Assistant (PA), a new software feature that requires a dedicated server. PA provides both rules-based call routing and impressive speech-recognition capability, allowing users to verbally issue commands to the phone. Extension Mobility is Cisco's new moniker for its enhanced "free-seating" capabilities. It lets clients log onto any phone in an AVVID system and download their CallManager profiles to that phone, including extension, button mappings and voicemail accounts.

There are, however, opportunities for improvement. First, while 91 percent of our feature list is now already fully supported, 100 percent would be better. Other laments relate to Cisco's management interface. The many gateway modules that comprise the AVVID system must be configured via each gateway's IOS command line; they are not configurable via the same browser interface that's used for the CallManager call controller. While this probably makes IOS-philes happy, the multiplicity of required management interfaces is confounding. Secondly, limitations in CallManager's real-time monitoring capabilities persist, though Cisco's new Admin Serviceability Tool is a step in the right direction.

Mitel's 3100: The 3100 Integrated Communications Platform (ICP) is likely the lowest-end IP-PBX on the market today, with a maximum station capacity of 34–24 of which can be IP phones, with in-line power. This truly IP-oriented system represents an aggressive foray by Mitel into the small-office market, where 3Com and Vertical Networks have been successful.

The 3100 ICP's quick and simple installation, built-in voice mail and automated attendant, and support for the full complement of Mitel's IP phone sets and telephony features, all contribute to our recommending it to the forward-thinking, small-office buyer.

To complement the 3100 ICP, Mitel offers the optional SME Server, a data-network-in-a-box. The Linux-based server includes all the necessary underlying network infrastructure (IP routing, DNS, firewall, etc.) and services (file and print email, Web server, etc.). As an added option with SME Server, a ServiceLink agreement establishes an SSL connection from the unit back to a Mitel NOC, to which the database is mirrored and email is automatically backed up. All of this is a package offered to small office managers who are willing to write one check and have all their voice and data needs addressed.

The 3100 ICP's performance is steady. No reliability issues came up, and voice quality on the IP phone sets rated as business-quality or better, with latencies well within acceptable limits.

Noteworthy features besides the optional SME Server include the 5822 IP Softphone, which can provide point-and-click access to MS Outlook contacts; and the figuration was slightly different for Shoreline and Vertical. All latencies except for Siemens' are excellent, and well under 100 milliseconds (ms). Siemens' latencies, under different settings, were 158 and 207 ms. These latencies are high enough to be discernible and, in some cases, even annoying to users.

Figure 3 shows the one-way latency in the case where an IP phone (again, except for Shoreline and Vertical) is connected to a "remote" analog phone off a T1 trunk. A low-bandwidth vocoder is used in this scenario—G.729, G.729a or G.723.1, depending on which the system supports. While the latency of most systems increased slightly in this case, compared to local IP phone-to-IP phone connections, it dropped somewhat for Siemens. This is because Siemens processes all local IP phone-to-IP phone calls through its TDM/digital circuit-switching fabric, which adds appreciably to end-to-end latency.

We did not perform Mean Opinion Score (MOS) ratings of IP-PBX voice quality this year, for several reasons. MOS scores have tended to rise in recent years, due to vendors refining and fine-tuning their vocoders, to the point where all MOS scores are now generally excellent—in the range of 4.5 to 5.0. Also, MOS is a one-way playout of a recorded voice sample, and so doesn't

Desktop Tool, which allows users to customize their IP phone sets via a browser connection to the 3100 ICP.

The main shortcomings of the 3100 are in its management interface: There are no canned reports and no monitoring capabilities to identify hot spots. Additionally, the 3100 ICP could use T1 support—it currently supports only up to eight analog, loop-start trunks. Also notably absent is a low bit-rate vocoder and/or VAD-compression support for IP bandwidth optimization.

Mitel 3300: The 3300 Integrated Communications Platform (ICP) is a robust, feature-rich, end-to-end IP-based telephony platform, with IP trunking support to network up to 80 discrete systems. The latest edition of the 3300, which we tested, also features enhanced capacity and browser-based management.

The system's feature support is especially rich. All 32 of the list of features we looked for were fully supported. As far as other IP-enabled capabilities, Mitel's 6110 software package is a Web-based contact-center management application, which allows highly granular management of distributed contact centers through an intuitive and well-organized graphical front end. The 5550 IP Console is a nifty PC-based "attendant console" application, which supports an optional telephony keypad and headset. The 5310 IP Boardroom Conference Unit, an attachment to Mitel's high-end 5020 IP phone set, is a high-quality, IP-based, full-duplex conference phone. We confirmed in the lab that the unit delivers very good voice quality.

Additionally, the 3300 ICP registered very good voice-quality performance in all interactive tests, and no stability issues were observed. The system achieved a 100-percent call completion rate on our reliability test.

While all this suggests a sound, solid IP-telephony system, there are gaps in the 3300 ICP's management interface, and redundancy options are conspicuously absent. Some of this will change with Mitel's next release. Also, IP trunking between distributed 3300 systems is currently accomplished through the use of an optional, standalone 3800 IP Gateway server, which links to the 3300 ICP via T1. This call routing is a bit convoluted, and also adds to the latency of IP calls between geographically dispersed systems. This connectivity needs to be made cleaner and integral.

Shoreline's Distributed Call Control: The Shoreline3 system is straightforward to deploy and consists essentially of three pieces: a "PSTN interface module" (ShoreGear-T1 or E1); a "phone station interface module" (ShoreGear 12, 24, or Teleworker); and a Microsoft IIS-based "ShoreWare Director" server node. The pieces are all connected via IP links.

The ShoreWare Director server—accessible from any browser—is where system configurations are generated and pushed out automatically to all deployed components. Call control occurs on all ShoreGear devices, each of which contain replicated call-routing intelligence. We agree with the vendor's contention that this approach eliminates any single point of failure.

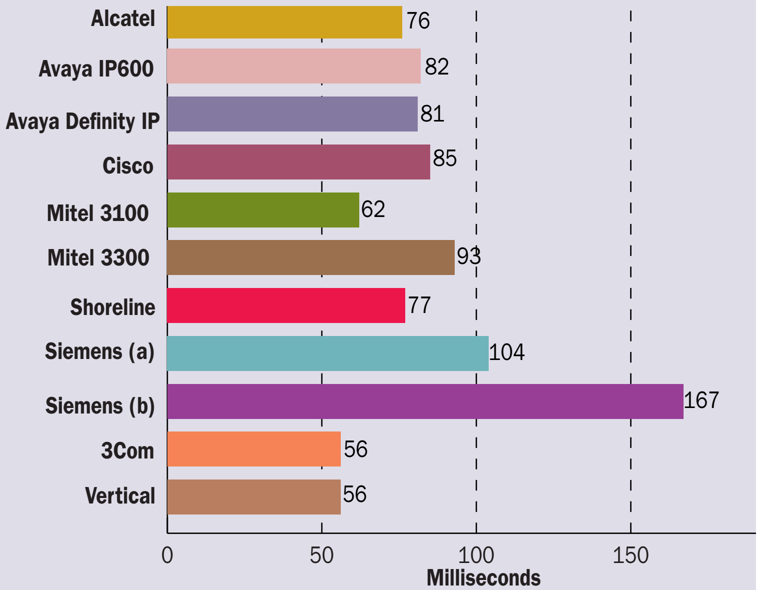
Scalability—up to 5,000 users, although this number is reduced by each trunk channel—is achieved by simply dropping ShoreGear modules where needed and configuring them with what our testers concur is the most intuitive management interface of any IP-PBX we've reviewed to date.

The Shoreline3 currently supports only analog phones, however. Interactive voice quality over local analog phone-to-remote analog-phone connections is excellent—ranging from 4.2 to 5.0 out of 5.0 across all codec environments. Perhaps most impressively, one-way latencies for all G.711 and G.726 vocoding conditions were lightning fast, only about 36 milliseconds. Add solid reliability, easy setup and integrated voicemail, and this all adds up to a top-shelf product.

Shoreline's support for only analog phones does have an effect, however. Two of the list of 32 features we looked for were not supported with the analog phones Shoreline provided for the testing. In addition, four of the features were supported, but only if the user also employed a PC running Shoreline's Personal Call Manager software. Shoreline says that all of the unsupported features could have been handled using some manner of analog phone other than the models they provided. Also, while even high-end analog phones cost just a fraction of what proprietary digital or IP phone sets cost, the Shoreline3 still doesn't come cheap at $846 per line. The system could also use support for VAD silence suppression, although support for a very capable low-bit-rate vocoder, G.729a, was added in the latest system version.

Summary highlights of the Shoreline3 would underscore the system's distributed switching and call control, and Shoreline's browser-based management interface, ShoreWare Director. System configuration modifications are quickly and dynamically propagated system-wide. Tangential to its architecture is Shoreline's Distributed Voicemail and Auto-Attendant Application, which allows voice mail and auto-attendant consoles to be distributed

FIGURE 3 One-way Latency: IP To Remote Analog, G.729 Or Other, Low-Bit-Rate Vocoder (1) **And VAD**



(1) Tests were all run between an IP phone on the 10/100 LAN out to a remote analog phone, connected off a T1 trunk, using a low-bit-rate vocoder, such as G.729.

Exceptions were: Shoreline, which supports only analog phones (although all calls are VoIP-processed); Vertical, which supports only analog and digital phone sets (although all calls were VOIP-processed); Siemens, which supports only G.723 for its low-bit-rate vocoder; and 3Com, which supports G.726 as its low-bit-rate vocoder. Siemens was tested with two separate jitter buffers, short (a) and normal (b), which is the default setting on the system.

measure or address crucial metrics such as latency or bidirectionality.

Those metrics are exactly what Miercom's "Interactive Connection/Voice Quality" tests are designed to assess. Figure 4 shows the comparative quality of the different IP-PBXs for varied connectivity environments. A 1-to-5 scale, similar to MOS ratings, is used.

Hack-Attack Vulnerability

Something new in this year's performance testing was to see how the IP-PBXs reacted to conventional Denial-of-Service (DoS) attacks. All the products were subjected to three different off-the-shelf DoS attacks, to observe the effects on the call controllers and IP phones separately, both during and after the attacks. The three DoS attacks we applied were: Ping Flood, a perpetual "ping" from any Unix OS; UDP Flood, which congests the target node's network connection with UDP packets; and Jolt2, which actively attacks the IP stack of the target node.

We've decided not to release the details of how individual systems tested, due to the limited scope of our DoS testing and in order to not provide hackers with useful information. However, there is cause for concern:

- In half the systems, the call controllers were vulnerable to Jolt2 attacks; phone calls could not be placed while the attack was in progress.
- Some of the systems that failed Jolt2 also were

across a wide IP network, and yet still behave like a single system.

Siemens' Hicom 150 H: The Hicom 150 H IP Convergence Platform is a classic TDM-based digital PBX which is transitioning to the world of IP telephony. Accordingly, it has its best days ahead of it. We encountered several issues with IP-enabled aspects of the system, most of which, Siemens promises, will be addressed with version 3.0 of the system software, slated for March 2002 general release.

Some architectural aspects, however, give cause for concern. For example, all calls, even local IP-to-local IP calls, are processed by the TDM switching fabric. As a result, one-way latencies were on the high side—over 100 milliseconds in almost every case—and interactive voice-quality ratings on IP-to-IP calls, predictably, were on the low side. Voice-activity detection (compression) is not now supported by the Hicom 150 H. The only supported low bit-rate vocoder is G.723.1, which exhibited marginal voice quality: Connections were rated 3.0 out of 5.0, for IP phone-to-IP phone connections. Much of this was due to irritating high latencies.

Reliability "call-completion" testing achieved not quite "3-nines" performance. And the beta code we tested on the IP phones themselves exhibited some instability. Also, we found that a cold reboot of the system requires a manual reset of each phone.

The Hicom 150 H supports the entire list of features we used for comparison purposes, as well as some notable advanced features. One is the Attendant P Office V4.0, a software-based attendant console that supports slick network-wide monitoring of stations, and is one of the best PC-based applications we reviewed in this test project. The Hicom 150 H also offers broad station device support for a relatively low-end system, including wireless phone support. Reflecting its heritage and enormous installed base in the TDM/digital marketplace, the system also supports a broad range of trunking protocols. The management interface is organized and reasonably simple to drive. What's more, this is among the lowest-priced systems of the ones we tested.

Siemens' on-site technicians demonstrated transparent feature support and management across multiple distributed systems. Also noteworthy is Siemens' Multimedia Messaging product, a unified messaging application that works with either Lotus Notes or Microsoft Exchange, and is designed to work with Siemens' HiPath ProCenter Office call-center application.

3Com's Superstack 3 NBX: This was among the first chassis-based, all-IP, telephone systems on the market, and carved out a niche for 3Com in the small office market sector. Testing of its most recent incarnation, the Superstack 3 NBX, suggests that the system has justified staying power and that it's not just for small offices any more.

The Superstack 3 NBX now supports up to 600 IP stations—given a four-to-one station-to-trunk ratio (750 vulnerable to other attacks (one failed a Ping Flood, the other a UDP Flood). However, in every case but one, the call controllers resumed normal activity after cessation of the attack; the exception required a manual reset.

■ When IP phones were attacked directly, they were incapacitated—incapable of receiving or placing calls—during the course of the DoS attack. All the systems with IP phones were incapacitated while under a Jolt2 attack, except for 3Com, whose phones operate on the LAN exclusively at Layer 2 and therefore cannot be individually DoS-attacked. Five of the systems' IP phones also failed during Ping Flood attacks, and three systems' phones were unable to recover immediately after cessation of the DoS attack, requiring a reboot in order to resume normal operation.

Clearly, this is an area that warrants careful evaluation by prospective users. If DoS attacks such as these are launched by a local perpetrator—someone with a station on the same IP subnet as the target call controller or IP phone—common security measures such as firewalls and VLANs are ineffective (these protections are mainly for IP traffic traversing multiple subnets).

There are steps to take to shore up system vulnerabilities to such attacks. Security enhancements like intrusion detection sub-systems, for example, can be added to the Layer-2/Layer-3 infrastructure underlying the IP-telephony system. Another effective defense was to statically map the MAC addresses of LAN devices to specific switch ports, which precludes unauthorized devices from accessing the network.

Denial of service attacks incapacitated IP phones

Conclusion

Reviewing these 10 IP-PBX systems reveals broad product differences in almost all respects. While the capacities of several of the systems have increased substantially over the last year, users will generally categorize these as we have, with high-end systems (over 1,000 stations) from Alcatel, Avaya, Cisco and Shoreline, mid-range systems (200 to 1,000 stations) from Avaya, Mitel and 3Com, and low-end systems (under 200 stations) from Mitel, Siemens and Vertical.

total devices are supported, but this number is diminished by each trunk channel). The system is a clean, simple, modular IP-PBX, which is straightforward to deploy and administer from a standing start. It is a stable system and performs well, with superb voice quality using G.711 vocoding and average latencies among the lowest of all systems reviewed. The Superstack 3 NBX passed all reliability tests without issue. In addition, the management interface is well organized and user friendly.

On the LAN, the 3Com system is more "voice-over-Ethernet" than it is voice-over-IP. This very proprietary approach boasts some processing, bandwidth and speed efficiencies by keeping voice communications a strictly Layer 2 transaction. For remote IP communications, the Superstack 3 NBX invokes "IP-on-the-fly," where outbound voice streams are wrapped in IP packets for transport over an IP link.

A notable feature is the 3Com pcXset, a standalone soft phone application that extends all NBX phone features to the PC. The pcXset can be upgraded to a Complements Attendant Software (CAS), a rich, software-based attendant console. Multi-Site Messaging is 3Com's scheme for meshing multiple, distributed voice mail systems without consuming VOIP trunking resources. 3Com also boasts of its system's application programming interfaces and its partner program, through which third-party vendors deliver added features and applications for the Superstack 3 NBX platform.

Our testers had a few complaints about this IP-PBX, however. Voice quality ratings when using the G.726 (ADPCM) vocoder were marginal. Another issue is site-to-site IP scalability. A maximum of just 24 concurrent VOIP channels between distributed Superstack 3 NBX systems is supported. And furthermore, users are charged extra for this capability, which 3Com calls "Virtual Tie Lines." It is licensed for 2, 4, 8, 16 or 24 channels.

Vertical's InstantOffice 6000: This latest offering from Vertical Networks supports up to 180 analog and 168 digital stations, essentially doubling its capacity of a year ago. It has a modular, chassis-based design, which allows users to add blades when needed. It offers an intuitive management interface, easy setup and powerful diagnostic tools, which many higher-end systems lack.

The InstantOffice 6000's performance is impressive, although the system supports VOIP only on the trunk side, between distributed systems. Interactive voice-quality ratings using analog phones earned a perfect 5.0 in all analog-to-remote analog calls, in all conditions tested; and 4.5 for digital-to-remote digital calls. Latencies were very low, under 56 milliseconds in all conditions tested. The InstantOffice 6000 supported 31 of the list of 32 telephone features we looked for.

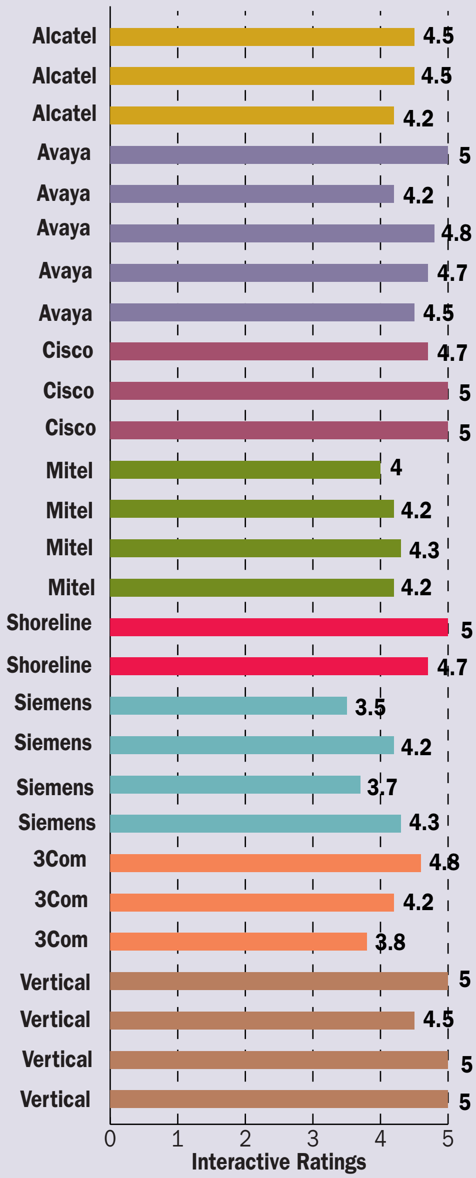
The InstantOffice proved reliable, completing 100 percent of call attempts in our reliability testing. Redundant power supplies and in-the-box disk mirroring all enhance the system's reliability.

Among noteworthy features is the InstantOffice Contact Center, by Interactive Intelligence, which is OEMed and co-developed by Vertical. The package, while a scaled-down version, is robust and easy-to-use, and it integrates call-center capabilities such as skills-based routing, call monitoring and data-based call prioritization.

InstantOffice's management interface, the Remote Management Console, is Web-based and designed for use at small and branch offices, which typically lack technical expertise. The interface features a slick graphical chassis representation, where double-clicking on any module allows configuration of that particular module. The InstantOffice platform connects to both circuit- and packet-switched worlds via a variety of protocols and interfaces. The system integrates a PBX, a multiprotocol router, VOIP gateway and enhanced applications all in one chassis.

Only one complaint with the InstantOffice 6000—there is no IP phone support□

FIGURE 4 Interactive Connection/Voice-Quality Ratings



Vendor	Connectivity	Vocoder	VAD	Rating
Alcatel	IP-IP phones via LAN	G.711	No	4.5
Alcatel	IP-remote analog via T1	G.729a	No	4.5
Alcatel	IP-remote analog via T1	G.729a	Yes	4.2
Avaya IP600	IP-IP phones via LAN	G.711	No	5.0
Avaya IP600	IP-remote analog via T1	G.729a	No	4.2
Avaya Definity	IP-IP phones via LAN	G.711	No	4.8
Avaya Definity	IP-remote analog via T1	G.729a	No	4.7
Avaya Definity	IP-remote analog via T1`	G.729a	Yes	4.5
Cisco	IP-IP phones via LAN	G.711	No	4.7
Cisco	IP-remote analog via T1	G.729	No	5.0
Cisco	IP-remote analog via T1	G.729	Yes	5.0
Mitel 3100	IP-IP phones via LAN	G.711	No	4.0
Mitel 3100	IP-remote analog via T1	G.711	No	4.2
Mitel 3300	IP-IP phones via LAN	G.711	No	4.3
Mitel 3300	IP-remote analog via T1	G.711	No	4.2
Shoreline	Analog-analog via LAN	G.711	No	5.0
Shoreline	Analog-rem. analog via T1	G.729a	No	4.7
Siemens (a)	IP-IP phones via LAN	G.711	No	3.5
Siemens (a)	IP-remote analog via T1	G.723	No	4.2
Siemens (b)	IP-IP phones via LAN	G.711	No	3.7
Siemens (b)	IP-remote analog via T1	G.723	No	4.3
3Com	IP-IP phones via LAN	G.711	No	4.8
3Com	IP-remote analog via T1	G.726	No	4.2
3Com	IP-remote analog via T1	G.726	Yes	3.8
Vertical	Analog-analog via IP trunk	G.711	No	5.0
Vertical	Digital-digital via IP trunk	G.711	No	4.5
Vertical	Analog-analog via IP trunk	G.729a	No	5.0
Vertical	Analog-analog via IP trunk	G.729a	Yes	5.0

Shoreline and Vertical Networks are notably different from the others because their systems do not now support IP phones.

All the vendors offer new and unique applications and features that exploit the characteristics of the underlying IP infrastructure. The most common ones include softphones, unified messaging systems and distance-independent call-center systems.

There is widespread support for VOIP standards in today's IP-PBX products, notably H.323, but variations in implementations continue to frustrate interoperability. Also, the standards do not now address all the telephone features that IP-PBX vendors need to implement, leading to some proprietary protocol "extensions" with virtually all the systems.

With only isolated exceptions, though, the systems all proved they deliver excellent performance. In our estimation, users of these systems' IP phones and/or VOIP processing would not be able to distinguish call quality from predecessor TDM/circuit-switching systems□

Companies Mentioned In This Article

3Com (www.3com.com)
Avaya (www.avaya.com)
Alcatel (www.usa.alcatel.com)
Cisco (www.cisco.com)
Empirix (www.empirix.com)
Mitel Networks (www.mitel.com
Nortel Networks www.nortelnetworks.com)
PowerDsine Inc. (www.powerdsine.com)
Shoreline www.goshoreline.com
Siemens www.siemensenterprise.co
Vertical Networks (www.vertical.com