Special Report

Taking the voice over IP plunge

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Taking the voice over IP plunge

Voice/data convergence has been talked about for 20 years, but this time it is really happening. One clue: none of the traditional PBX makers are developing next generation TDM machines.

Nonetheless, questions remain. Is VoIP up to corporate quality standards? What do you have to do to prep your network to support VoIP? And can you tackle this yourself or do you need to bring in experts?

This Network World special report is designed to answer pressing VoIP questions for technology decision makers.

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VOIP TIPS FROM THE TRENCHES

Based on our testing, here's how to prepare your network for VoIP.

BY KENNETH PERCY AND MICHAEL HOMMER
NETWORK WORLD GLOBAL TEST ALLIANCE
NETWORK WORLD, 01/27/03

In a voice-over-IP deployment, the hotspots aren't as obvious as you might think. The clear-cut decisions center on VoIP-specific products such as IP phones, IP PBXs and voice gateways, but weaknesses in your data network will become magnified when you introduce VoIP.

The first question to ask in order to avoid some postdeployment surprises is: In what kind of shape is my existing network? Real-time voice traffic will be affected by any bottleneck on the network. A delay of 1 second in retrieving a data file from a server because of congestion might be barely noticeable to the user, but add just 50 millisec of delay on a phone call and it's the difference between high-quality and very poor-quality voice communications.

Before deploying any VoIP gear, you must scrutinize your network with an audit that includes three primary considerations:

● **Utilization and network statistics.** Maximum, minimum and average metrics for bandwidth consumption, latency, jitter and packet loss should be included in your audit.

In the case of bandwidth utilization, the hotspot for potential bottlenecks lies in the interswitch links that make up your backbone. Maximum bandwidth utilization should be dictated by failover considerations, says Joe Tomasello, of Foundry Networks.

"Uplinks should always be deployed redundantly, at least," Tomasello says. "If one link fails, the other link should be able to handle the load for both links. Therefore, utilization on a trunked Ethernet uplink, for example, should never exceed 50%."

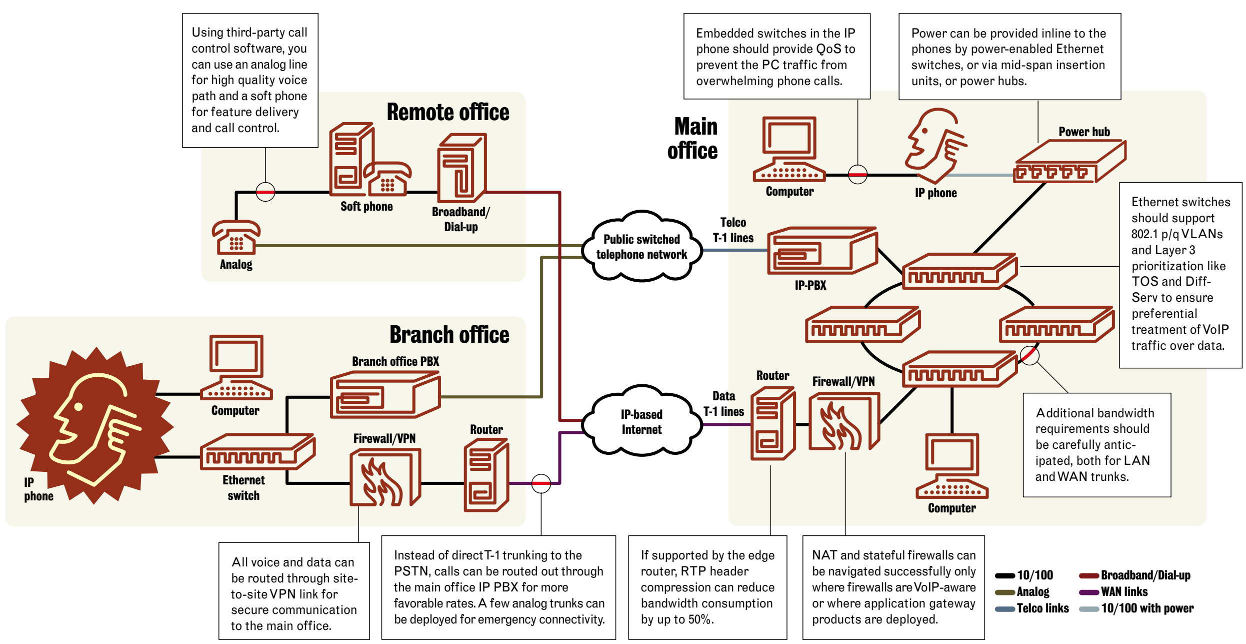
Latency, jitter and packet loss that would be detrimental to business-quality voice are rare occurrences on today's LANs. Where they do exist, they are usually the result of antiquated equipment (such as hubs, 10M bit/sec Ethernet switches or switches with low memory capacities) or silly mistakes. Examples would be a switch with its autonegotiation algorithm disabled, forcing all switch ports to default to 10M bit/sec half-duplex communications; or a swath of Ethernet cable that's a lot longer than 328 feet, the maximum supported Category 5 cable length for Ethernet.

Check that your network latencies don't exceed 100 millisec, and maximum jitter should never be more than 40 millisec. Packet loss should be zero, but the rule of thumb for tolerable voice quality is less than 1%.

● **Review of infrastructure elements.** The gear that powers the network should be reviewed for necessary feature support and correct configurations. Ethernet switches that will be

Prepping your data net for voice

Every point on your enterprise network will need some attention in order to accommodate high-quality voice transmission.



touched by VoIP traffic should support virtual LANs. This will allow segmentation and isolation of your voice traffic across the data network.

IP-based quality of service (QoS) — such as type of service (TOS) or Differentiated Services (Diff-Serv) — should be supported. In a large VoIP deployment, this allows prioritization of packetized voice over more delay-tolerant traffic that must travel multiple subnets in a routed environment. A few IP PBX systems also require multicast support.

Next, these features should be reviewed to ensure that they're turned on, and to ascertain whether any configurations could pose problems. For example, if the Spanning Tree Protocol is enabled, changes in the Layer 2 topology could cause outages of up to 60 seconds while the updates are made to each switch's database.

• **Estimating bandwidth requirements.** Most value-added resellers and integrators have tools to help you ascertain how much voice traffic is currently carried by your voice network, both incoming and outgoing, on a per-station basis. If you prefer to arm yourself with your own calculations, there are two places to go for guidance. The first is to your existing PBX system. Most have reporting capabilities that yield utilization information. Some are easier to get at than others, but the utilization information you need — both station-to-station and station-to-trunk — should be there. The second is www.erlang.com, a Web site full of calculators and tutorials on voice traffic utilization.

When translating voice-utilization statistics into bandwidth requirements, we use the following rules of thumb for base, worst-case LAN bandwidth calculations. First, go with a G.711 coder/decoder (codec), because it consumes the most bandwidth and provides the best voice quality. For packetization rate — or the amount of voice payload per VoIP packet — assume 20 millisec, the default setting on most IP PBX systems. Using G.711 with a 20-millisec packetization rate, bandwidth utilization rounds up to 88K bit/sec per voice conversation. In calculating a worst-case, busy-hour scenario, assume that one out of every four users will be on the phone simultaneously.

In a 1,000-user IP voice system, multiply 250 (for the number of concurrent conversations) by 88K bit/sec per station for an additional bandwidth requirement of 22M bit/sec on your LAN.

The situation is far more complicated on the WAN.

There's no single rule of thumb for calculating bandwidth per call over a WAN because consumption varies by which voice coders (vocoder) and WAN protocols are used. Among low bit-rate vocoder options, G.729a is preferable. Repeated tests in our labs confirm this codec delivers the highest voice quality.

Voice Activity Detection (VAD) — also known as Silence Suppression — should be supported for each vocoders on the IP-PBX you select and, for purposes of calculating bandwidth, assume that it is enabled. With VoIP, speech and silence are packetized. To conserve bandwidth, VAD prevents "silence packets" from being transmitted. While the conventional rule of thumb is a 35% savings, our testing has seen bandwidth savings of 50% when VAD is used.

Assume the same 20-millisec packetization rate and, again, a 4-to-1 user-to-channel ratio.

For example, assume the use of frame relay as our WAN protocol, and G.729a vocoder. Plugging in the other variables outlined above, VoIP bandwidth over a frame link — with 35% bandwidth savings using VAD — rounds up to 18K bit/sec per VoIP conversation. So again, with 250 VoIP station users on the phone simultaneously as your worst case, assume you need an additional 4.5M bit/sec of bandwidth on your IP WAN.

One more tip concerning WAN bandwidth is to find out if your router supports RTP header compression. The standard IP/User Datagram Protocol (UDP)/RTP header consumes 40 bytes in a packet. If supported by your router, enabling RTP header compression can reduce the header information to just 2 bytes, yielding an overall bandwidth reduction of up to 50%.

Once you determine how much VoIP data will likely be running across your network, you can focus on the network it will be touching to accommodate it.

VLANs and QoS: Gotta have it

A successful VoIP deployment requires VLAN and QoS capabilities at every point on the network, necessitating more intelligence in closet switches than ever before. In a best-practices scenario, all voice traffic should be isolated on a Layer 2 VLAN dedicated to voice and signaling traffic.

Switches also should support Layer 3-based QoS features, such as TOS or Diff-Serv, for prioritizing traffic through multiple subnets. This is particularly important at aggregation points within the network. QoS will ensure that the VoIP traffic gets top billing at congestion points and help to maintain low latency and jitter. Setting the exact QoS value for TOS or Diff-Serv would be handled on a case-by-case basis and would depend on what the network looks like and what other traffic might need to be prioritized.

But prepping for voice-specific VLANs and QoS capabilities doesn't stop at your network switches. IP phone support for 802.1p/Q VLAN tagging, or the ability to "tag" packets coming out of the phone that defines membership in a particular VLAN, also is necessary. Phones should come with the ability to set TOS or Diff-Serv priority bits within the packets' IP headers.

The new breeds of IP phones enable "one-wire-to-the-cube" installations, supporting two-port mini-switches on the backs of the phones. The Ethernet connection from the LAN attaches to one of these switch ports. The other connects to the PC, positioning the phone directly inline between the PC and the surrounding LAN. Both data and voice, then, can access the network over the same 100M bit/sec pipe.

To prevent PC traffic from overwhelming voice conversations, most phones include some kind of traffic-shaping mechanism in the switch. For this reason, phones with hubs in the back — rather than switches — are less desirable. Hubs operate at only half-duplex, minimizing bandwidth to the cube, and they cannot implement QoS.

Inline power over Ethernet: Gotta get it

In a VoIP environment, you can look to good old Ethernet to provide an important new functionality over and above a data link — delivering power inline to IP phones. There are essentially two options for delivering power over Ethernet.

First is to purchase Ethernet switches that deliver data and power. Avaya and Cisco are examples of vendors that sell Ethernet switches with inline power.

The second is to provide midspan power insertion devices called "power hubs." Most power hubs are OEMed from the Israeli firm PowerDsine. Ethernet cable runs from the cubicles are connected to the power hubs, which are patched to the Ethernet switch. So a 48-port power hub, for instance, will power 24 IP phones.

Most IP phone interfaces are "power aware" to receive power via the Ethernet connection. Power is sent over the unused pairs of a Cat 5 cable or over the same pairs (phantom power) used for data, in which case the Ethernet power sources employ a detection algorithm to determine whether to send power to a connected interface.

Those few IP phones today that do not have power-aware Ethernet interfaces can use a power hub and maintain the one wire to the desktop. They usually ship with special line splitters, or "pig tails." These line splitters have a female RJ-45 connector on one end to receive the powered Ethernet, branching out into two prongs. One is a DC power connector for the phone, the other is a male RJ-45 Ethernet connection to the phone. Power is effectively "peeled off" the incoming Ethernet connection and sent to the phone's DC power jack. The data goes to the Ethernet port.

Determining whether powered switches or mid-span insertion is better is a matter of cost. If you need more switches, or those you have need replacing, you might as well go with powered switches.

Because power failure to your Ethernet switches will leave your company unable to communicate via voice or data, back-up power for your Ethernet switches should be closely examined.

Legacy digital phones sets are terminated at the PBXs, so UPSs for the PBX protect the phones and the PBX. IP station connections terminate at the Ethernet switches. If power to the switch is lost, loss of both voice and data could occur. Upgrading UPSs to prevent longer blackouts also might become necessary. The ratio of necessary switches to UPSs depends on the size of the UPS.

Most major UPS vendors provide resources to determine which model UPS a user should get based on the load and the duration of coverage.

Security: Gotta foil eavesdroppers

Security issues have always been a standard refrain for VoIP detractors, and with good cause. Lest we forget, VoIP is data, with all its vulnerabilities. But, while security at the network's edge is always the prevalent concern, we also warn that you have to look inside for security concerns with a VoIP deployment.

Treat your IP-based PBX with at least the same diligence as you would any mission-critical server. Defining VLANs and enabling security features on the switch that map specific media access control addresses to specific ports is a good start. In large implementations, install a dedicated internal firewall to protect the PBX.

Perhaps the most dreaded form of deviant behavior facilitated by a VoIP platform is eavesdropping. TDM-based PBXs sit on physical isolated networks, and they use highly proprietary digital signaling. While conversations can be tapped and recorded, both require physical access to the PBX or physically splicing phone lines.

With VoIP, off-the-shelf packet sniffers can capture conversations and not only replay them, but also store and distribute them as electronic files. VoIP equipment vendors are beginning to add security features to encrypt media streams. For example, on its S8700 Media Server and G600 Media Gateway, Avaya has added Media Encryption on an active IP call. When nonauthenticated users attempt to intercept the packets, they hear white noise when replayed.

While it obviates the economies converged networks can produce, some IT administrators take security concerns so far as to run parallel physical networks for voice and data rather than run both across the same links. Customers with the budget for it can achieve the best of all worlds from a security point of view. But this option is expensive, and tight security can be achieved with a well-conceived deployment.

The edge: Gotta get around NAT

An important fundamental aspect of VoIP is that there are two different data paths: the signaling path and the voice path. When a PBX-attached IP phone goes off-hook, it signals the start of a call-setup process. In most IP PBX systems, the back-end call server will set up, tear down and peripherally monitor call states. But the packetized voice conversation occurs directly between the endpoints (peer-to-peer) without further back-end intervention.

This characteristic makes network address translation (NAT) a troublesome proposition because signaling comes from one network node (the call server), and the media stream comes from another (IP phone). This problem is compounded because NAT functions at Layer 3. Peer-to-peer voice communications occur via the Real-Time Protocol (RTP), which embeds the source and destination IP addressing in the Layer 7 headers, rendering the return data address inaccessible to any NAT engine.

Stateful firewalls are equally problematic. Outbound VoIP communications create "pinholes" through the firewall to allow outbound voice communications. However, inbound voice data will attempt to enter the network using different socket information than the signaling data used, and the firewall will consequently block the RTP stream. Furthermore, creating pinholes for all the possible port ranges negotiated by the endpoints defeats the firewall's purpose.

One possible solution is a VoIP-aware firewall, which adds application-proxy functionality to base firewall products that enable dynamic opening and closing of firewall ports on a connection-by-connection basis. This functionality can be added via upgrades to an existing firewall product, or via third-party hardware that resides logically alongside the firewall, such as those offered by Jasomi Networks and Kagoor Networks.

You also can take a VPN route to support VoIP between sites or for remote access because VPNs circumvent NAT and firewall issues by tunneling. Site-to-site VPN links are becoming more common. If you're considering one, using it to link your PBX network should be added to the plus column. However, the gotchas with VPNs that you should beware of include bandwidth and latency. The overhead and delay encryption adds should be taken into account for optimal planning.

Planning makes perfect

While VoIP can ride over the highways as our data currently does, it is a new application with new rules. Deciding on the right VoIP solution is just the beginning; deploying it on the network properly is the real task.

Knowing your network, ensuring the quality of your voice traffic, making sure your network and personnel infrastructures are up to the task and properly protecting your IP PBX will help your deployment be successful.

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QUALITY IS A TOP VOIP QUESTION

BY PHIL HOCHMUTH
NETWORK WORLD, 10/07/02

Is the voice in voice over IP good enough to bet your business on?

That's the make-or-break question for many network executives as they consider the promise of integrated voice/data.

Most users and industry watchers say it's relatively easy to achieve toll quality in the lab, but it can be more difficult in a production network. That helps explain why some companies see a future with VoIP everywhere, while others are hesitant to use it on critical links, or for customer-facing applications, such as in call centers or help desks.

"Traditional businesses like ours are . . . on the conservative side. We're not willing to jeopardize our brand name," says Jeff Fountaine, a network analyst with Armstrong World Industries, a Lancaster, Pa., maker of industrial and home flooring and ceiling products. Among his concerns would be a phone order going awry because of poor sound quality over an IP link.

Still, the potential cost savings are strong enough that Armstrong is willing to give VoIP a shot for certain applications. The company is planning to test an internal campus-to-campus IP telephony deployment before delving deeper into the technology.

Quality measurement

In a recent Network World survey of 250 IT executives, the top perceived drawback for network convergence was the lack of quality-of-service (QoS) assurance on corporate networks. Almost half of those surveyed said that the quality of IP voice was a drawback.

With voice quality being such a sticking point, the trick is to come up with a good way to ascertain whether your network can support toll-quality VoIP. Some experts say there are hard tests and metrics for proving an IP telephony system, while others say the process is more art than science.

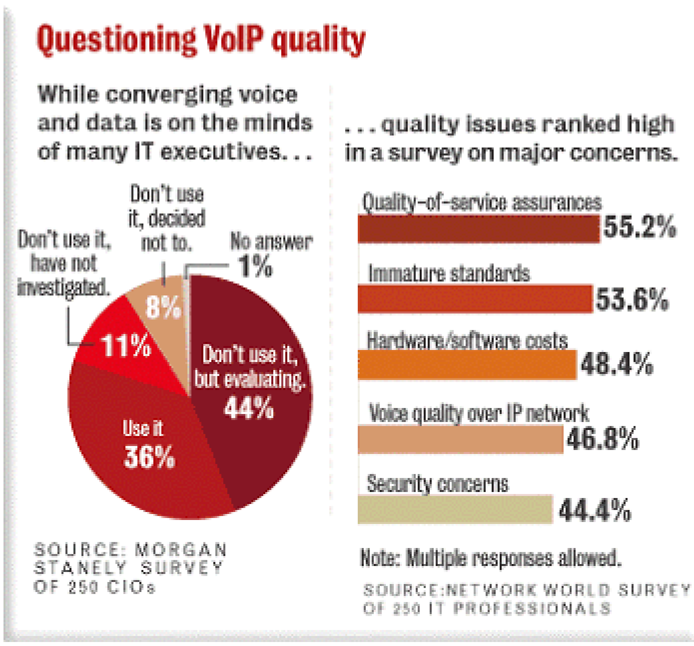
Factors that might diminish the quality of an IP phone conversation rarely lie in the actual VoIP gear anymore, says Mike Hommer, manager of lab testing at Miercom, an independent IT testing and consulting firm and a member of the Network World Global Testing Alliance.

"When we started testing VoIP products in 1997, 80% of the metrics we looked at were related to the performance and voice quality. Now that's down to about 10%," he says. "The quality issues - as far as IP voice equipment being able to efficiently encode and decode voice - have become less of a concern."

The issue now is on the network, Hommer says.

"Some people may have no idea how good or bad their network is for supporting real-time protocols like voice," he says.

Network latency is the No. 1 killer of real-time packetized voice traffic, Hommer says. The result of latency is jitter, which can cause an IP voice conversation to break up. Most IP voice products have jitter buffering technology or other technology that smoothes out and reorders voice packets before turning them into audio, but sometimes excessive network latency cannot be overcome.



Packet analyzers and port mirroring applications can be used to measure IP traffic volume and patterns to determine the sources of latency on a LAN or WAN. There are also voice-specific tests and methods that can be used for a finer level of testing.

One time-tested method for determining voice quality is the Mean Opinion Score (MOS), a test accepted by the International Telecommunications Union whereby a sample of 40 or more people from different ethnic or language backgrounds are given an audio sample that's several seconds long to listen to, and each person rates the quality of the audio on a scale from 1 to 5. By ITU standards, 5 is a "perfect" MOS score, while 4 is considered "toll-quality" or a high enough standard for delivering land-line service by public switched telephone network carriers.

"The good side of [MOS] is that you're using a human ear to determine voice quality," Hommer says. "The bad thing is that it doesn't take into account things like bidirectional speech quality," where a segment of speech might sound clear, but network congestion or inefficient routing might create pauses in a conversation, he adds.

Another metric for gauging VoIP quality is the Perceptual Speech Quality Measurement (PSQM), which is a computer algorithm used in testing tools from vendors such as Agilent, Empirix, Finisar and NetIQ for checking out IP voice equipment and network performance. Hommer says these tools are useful but don't always produce accurate results. Some of the computerized metrics, he says, are so stringent, "that the algorithm gives it a bad score, but from a human ear standpoint, you can't tell the difference."

Hommer says a mix of MOS scoring and testing with tools based on standards such as PSQM can provide the most-accurate assessment of how a VoIP network might sound.

"If you are a 50-site enterprise rolling out VoIP, and you have your own internal test lab," you might want to consider buying testing equipment before deploying VoIP, Hommer says. However, with the average VoIP deployment being in businesses with fewer than 100 lines, high-powered testing tools that can cost from $25,000 to $50,000 might be out of reach.

"The best way to test quality is to just put phones out there and see what people say," says Matthew Liste, practice manager for ThruPoint, a network consultancy that specializes in IP telephony technology. He recently completed an IP telephony project for Merrill Lynch where more than 1,000 Cisco IP phones were deployed at a site in New Jersey. He says enlisting the help of employees can be useful in determining whether an IP PBX system or LAN needs fine-tuning.

"If you put [IP] phones out and ask people to occasionally write down the quality of their calls [on a scale] from 1 to 5, you basically have a MOS score right there," Liste says. The advantage in that is minimal disruption to employees, he adds.

Another IP telephony consultant takes a more general approach.

"You can buy a lot of expensive equipment to test voice quality," says Susan Knott, global network architect for PricewaterhouseCoopers.

"But I've found that if my vice president of finance can talk to my CIO [over a VoIP connection], and they both say the quality of the connection is OK, then I say that's good enough," she says.

In an interview earlier this year, members of The Seattle Times' communications and IT staffs said they initially experimented with IP phones on an existing infrastructure consisting of shared 10/100M bit/sec Bay Networks hubs and Category 3 wiring.

"We had a phone on one hub that was averaging 60% to 70% utilization, and you would get some pops and snaps, but nothing the person on the other end noticed," said Paul DeWees, a network systems analyst.

At the Paulding School District in Georgia, Alcatel IP phone equipment was tested in a lab before phones and switches were deployed to 23 schools.

Paulding last year replaced several separate key systems throughout the district with a single Alcatel OmniPCX system running over LANs at the schools and connected by a fiber-optic, Gigabit Ethernet IP backbone.

"To test the voice quality, we pretty much just played it by ear," says Chris Ragsdale, director of technology at the school district. "We did some live calls in a lab setup, with the new IP phones right next to the phones we had in place."

Ragsdale says the calls sounded comparable between the circuit-switched and IP gear, and the deployment went fairly smoothly except at one school where some older LAN hubs were being used. The hubs were soon replaced with 10/100 Alcatel switches with QoS support.

"The hubs were creating too many collisions, and that was causing unacceptable clarity," Ragsdale says. "It was just something unexpected we ran into - something we didn't figure on when we tested the stuff out in the lab."

IS THIS A DO-IT-YOURSELF PROJECT?

Net execs weigh pros, cons of outsourcing a converged network.

BY TIM GREENE AND PHIL HOCHMUTH
NETWORK WORLD, 10/14/02

When Concordia College was upgrading its LAN from shared 10M bit/sec Ethernet to switched Gigabit Ethernet, Verizon approached the school with this proposition: How about running more than just data over the revamped network?

"I told them if I didn't have to pay any more for voice over IP, I'd be willing to [try] it," says Brian Heinemann, dean of IT at the Ann Arbor, Mich., school, which was using Verizon to manage its aging NEC PBX. "I didn't think they could do it, but they kept coming back with bids."

Verizon got the contract, and earlier this month the school switched 100 of its 400 phones to a Cisco equipment-based IP Centrex service, with plans to convert the other phones over time based on student demand.

Once sold on VoIP, Heinemann quickly realized that outsourcing was his only option. His data staff is small and he lacks a convergence expert. What's more, he couldn't duplicate in-house the round-the-clock network monitoring Verizon offered.

Concordia faced the same decision every other new VoIP user confronts: Whether to do it yourself, outsource it or do a little of each. The usual arguments about cost, staffing and network ownership come into play with VoIP as they do with any other network technology that might be outsourced, but there also are issues unique to VoIP that can affect the decision. Such factors include the lack of staff trained on voice and data technologies, and the need to keep a particularly sharp eye on bandwidth usage given the sensitive nature of voice traffic.

Among those opting for an in-house approach is the El Monte Union High School District near Los Angeles. The district installed a Mitel IP PBX to handle 500-plus phones in 10 facilities connected by a Gigabit Ethernet metropolitan-area network (MAN). The phones use a single dialing plan and are managed from one site, with voice traffic confined to virtual LANs.

The school district decided against outsourcing after it had bad experiences with Pacific Bell Centrex service, says Garett McKay, director of technology for the city of El Monte.

"We were dealing with a lot of problems in getting our carrier to respond in a timely fashion," McKay says of the Centrex service. The district already had dumped Pacific Bell's Centrex service in favor of Toshiba telephone key systems in each school, but they weren't ideal in that they required site visits to maintain.

McKay hasn't ruled out using a managed VoIP service at some point, noting that he hasn't had any complaints about PacBell's handling of the Gigabit MAN.

"That might make sense," McKay says, "since our routers [JDS Uniphase boxes, owned by Pacific Bell] are located at PacBell's [central office] anyway. I could see some efficiencies if they just managed our phone switches from those locations as well."

Another VoIP user, Rogers Group, cites a need for direct control over bandwidth as a top reason for not outsourcing. The Nashville, Tenn., crushed stone maker wants to ensure it can adjust bandwidth allotments for voice as it crosses the company's frame relay network among sites in five states. The company's WAN service is already shaky enough when it comes to handling data, says Mark Eckstein, a network administrator.

"We deal with many carriers that say they can offer service guarantees . . . but often they're not willing to put teeth into their contracts," he says. "Our phone service is vital, and I don't think we want to put that in someone else's hands."

Rogers, which is making the shift to IP to save on toll calls among sites, hasn't determined what gear it will use, although it is considering installing VoIP gateways to its Nortel PBXs or going to another vendor for an IP PBX. "We're still debating whether to go with a pure-IP system," Eckstein says.

Why VoIP?

25% of enterprise IT executives Gartner surveyed said that long-distance toll bypass and reduced service provider connections were the biggest benefits of IP telephony.

Just a little help

Some companies want to do most of the work themselves, though not the initial effort. The Royal Society for the Prevention of Cruelty to Animals (RSPCA) in England relied heavily on Cable & Wireless when it shifted from traditional voice to IP voice when it moved to new headquarters last year.

The RSPCA has installed Cisco VoIP gear at its central headquarters and a regional headquarters totaling more than 450 phones, and plans to extend the converged network to 47 other sites. In setting up the first two VoIP sites, the agency saved more than $100,000 by eliminating separate wiring for phones and another $400,000-plus because the IP PBX cost less than a traditional one, says Matt Winckless, technical communications manager. "We [just] needed help with the design and installation," he says.

Winckless says the RSPCA's IT staff of 30 can handle the ongoing management and likes the feature control he gets with the new system. When response to an RSPCA animal rights campaign flooded the agency's call agents, his staff was able to quickly create an interactive voice response option to go to recorded campaign information rather than to a live agent. It was done faster in-house than it would have been by a service provider, he says.

"We can engineer our own solutions as users come to us with new requirements," Winckless says.

Having a more hands-on approach also can smooth VoIP acceptance, Winckless says. Some users are reluctant to use new features and need to be eased into the new phone system. "The technology was the easy part. It's getting the users used to the applications that's hard."

Encorp, which makes switches for generators, also is handling its VoIP system internally. A prime reason for doing so is that it just doesn't appear to be all that tricky for IT staffers already familiar with network technologies, says Stan Seago, the Windsork, Colo., company's IT director.

"If they know networking, they're already up to speed. You just program the switch and assign IP addresses," he says. "If they're all on the same [virtual LAN], you don't even do the IP addresses; [Dynamic Host Configuration Protocol] does it for you."

Encorp initially ran VoIP over an 802.11b wireless LAN across four buildings, but upon relocating to a single building moved to a 150-user IP voice network anchored by an Alcatel 4400 IP PBX.

Because most VoIP installations are still relatively new and there are relatively few experienced colleagues with whom to consult, users acknowledge that making the decision to outsource or not is difficult and that it still might be a while until they know whether they did the right thing. "[As for the] potential benefits, we're not really going to know until a year from now," Concordia's Heinemann says.

In or out?

Factors to weigh about whether to outsource or build your own IP voice network:

	Pros	Cons
In-house	• Gain or retain control of the network. • Telecom and data staff get cross-training.	• Requires hiring experts or training. • Can cause data/telecom rift in staff. • Capital expenditure required.
Out-source	• No hiring or training needed to gain expertise. • Predictable costs. • May blend with existing managed data services.	• Loss of control. • Upgrades to new features on provider's timetable.

USERS HOPING SIP'S THE ANSWER

But concerns remain over whether VoIP products will work together.

BY PHIL HOCHMUTH AND TIM GREENE
NETWORK WORLD, 10/21/02

John Ridley is stuck between the old world of circuit-switched telephony and the new world of voice over IP.

How soon the Coca-Cola network executive can move forward depends largely on which IP telephony standards key vendors support and how true they stay to those standards.

Ridley, who is looking to replace a loosely connected collection of old PBXs, is among a growing legion of network executives who say products based on Session Initiation Protocol (SIP) are the best bet for delivering the true benefits of VoIP. Such gear could help simplify network management and support new applications, they say, although only if the products boast Ethernet-like interoperability.

"The problem with IP telephony equipment today is that there is no [interoperability] among vendors," says Ridley, whose converged network would serve 70,000 employees.

Network executives are wary that vendors will repeat the mistakes they made with the older, less-functional H.323 technology. While H.323 has been implemented widely, vendors took so many liberties with it that getting their products to work together can been difficult.

"There are a lot of slacker customers out there like us who are just sitting on our old legacy stuff, waiting for the market to evolve," Ridley says.

Making the case for SIP

Work on SIP, now an IETF standard, started in 1995. It was designed to run on IP and supports a plethora of communications technologies from voice to instant messaging to video. SIP also lets users establish presence at different locations on a network, saying "I am here" and letting everyone or just a select group know it. SIP promises to support new services such as click-to-dial phone calling, interactive voice response navigation of Web sites and conferences that are set up when all participants are ready.

SIP is considered more efficient than H.323, which is commonly criticized as being too chatty, sending lots of messages over the network and creating potential congestion if VoIP is heavily used. Critics of H.323 say the overarching standard for interaction among a set of other standards is too unwieldy to customize.

Enthusiasm for SIP has been on the rise in recent years because of work done by organizations such as the SIP Forum, which now has 27 member companies including Cisco, Lucent and Nortel. Microsoft last year gave SIP a boost when it replaced H.323 with SIP in its Windows Messenger application, which supercedes Windows NetMeeting from the days of Windows 95/98.

SIP "bake-off" tests performed by Network World and by industry groups such as the SIP Forum also have helped build hope among network executives that SIP products would work together once released.

A survey of 96 vendors last year by Network World and Miercom showed 73% had H.323 products, while only 40% had SIP gear. However, 51% said they planned to implement SIP on their products over the next year.

Nortel and Siemens are among the vendors pushing SIP. Nortel says it shipped its Succession Communication Server for Enterprise Multimedia Xchange (CSE MX) in December. Siemens, which already has SIP-capable phones, says its HiPath IP PBX software will support a SIP stack along with its core H.323 code in the next major revision, which is expected in the first quarter of next year. Alcatel has said its OmniPCX IP PBX gear will be "SIP-capable" this year.

Others, such as 3Com, which use a proprietary version of H.323 for call control on its NBX IP telephony server, and Avaya are less enthusiastic. Avaya, third in IP telephony sales, says its line of ECLIPS IP telephony equipment can be enabled with SIP or H.323, but the company's proprietary H.323-based protocol is still the default.

And then there's VoIP market leader Cisco. The company supports SIP across its gateways, routers and some IP phones, and it says the protocol will be added into its CallManager enterprise IP PBX. But observers have questioned the company's reliance on proprietary protocols and whether that would interfere with the interoperability of SIP-enabled gear from Cisco and others.

"Cisco's voice solutions contain a number of proprietary and prestandard aspects," according to a recent report by Gartner Vice President Mark Fabbi. "Although Cisco supports SIP . . . in a number of products, its integrated solution requires that users implement its proprietary 'Skinny' protocol," or Skinny Call Control Protocol.

The vendor's primary call-control technology for its corporate IP telephony products remains proprietary, Fabbi writes. (Cisco also licenses its Skinny protocol to other vendors, such as Polycom). He adds that "nearly all" vendors use proprietary hooks in their VoIP gear that prevents companies' products from working well together.

"Most vendors don't want to be interoperable," says Brian Strachman, a senior analyst with Cahners In-stat/MDR. "No one wants to say, 'Go ahead and buy our IP PBX phone system, and oh, you can use Cisco or 3Com phones with it too.'"

The result, Strachman says, would be increased competition, which would force vendors to reduce prices and suffer lower profit margins.

"The [traditional] telecom mindset," has crept into the IP telephony world, he says, but he adds "it's been changing. Eventually it will be more open."

H.323, here and now

While many observers consider SIP to be the future of VoIP, we live in an H.323 world.

H.323 is still used widely on ViDENet, a multivendor IP voice and video network started in 1995, which connects more than 70 universities, research institutions and corporate networks via the Internet and Internet 2.

More than 500 gatekeepers and gateways, IP phones and video stations based on H.323, from vendors such as Cisco, Polycom and RadVision, are deployed. Each member institution is registered in a central directory, which lets H.323 voice or video sessions be set up easily.

"The reason for our widespread adoption of H.323 was that at the time, it was the protocol that proved to work with off-the-shelf

H.323 vs. SIP

While both H.323 and Session Initiation Protocol (SIP) can support VoIP, the protocols have plenty of differences.

H.323	SIP
ITU standard	IETF standard
Designed on models of ISDN and ATM signaling	Designed for use on the Internet
Older and more established, particularly in LANs	Newer standard
Complex, using both binary encoding and abstract syntax notation	Relatively simple, text based; similar to HTTP
Difficult to customize	Accessible to customization

components," says Jill Gemmill, who was ViDENet chair until September, and is assistant director of academic computing at the University of Alabama Birmingham.

Petroleum company Schlumberger chose H.323 over SIP for the same reason.

"SIP was pretty new when we first started looking into [our VoIP] project," says Brian Spolnicki, an information solutions technical lead at the company's Houston office. The company recently installed an IP-enabled PBX with H.323 VoIP in a call center to consolidate technical assistance into three call centers in Houston; Calgary, Alberta; and Caracas, Venezuela.

The centers, which support about 60 agents, are connected via T-1 lines and Schlumberger's DeXa.NET, a private OC-48 WAN. An Ericsson PBX with an IP card sits in Houston and supports call agents using Ericsson H.323-based IP phones in the Canadian and Venezuelan offices.

Spolnicki says he is comfortable with swapping out phones for any other commodity H.323 phone, or a Windows PC with the H.323-based NetMeeting program, if necessary.

"SIP may be involved in other segments of the company in the future," Spolnicki says, but "it didn't fit for this particular project at the time."

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