NETWORKWORLD®

EXECUTIVE GUIDE

VoIPs role in helping New Orleans recover; the future of VoIP peering; our testing of analysis tools; and much more.

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Introduction

VoIP: No turning back

The costs and benefits are right, but you still have to watch how you integrate real time traffic on your data infrastructure.

"It's actually a symbol not of where we've been, but where we're going. People can look back and say New Orleans was leading and not just in po' boys and beer. It was something pretty important."

That's New Orleans CIO Greg Meffert discussing an innovative disaster-recovery plan based in part on VoIP that his city has implemented in its struggle to recover from Hurricane Katrina more than a year ago. And while few if any VoIP-related projects can claim anything near as dramatic as a genesis, there remains little doubt these days that VoIP is where the majority of enterprise IT organizations are going.

This Network World Executive Guide to VoIP brings together a combination of news stories, trend analysis, test results and expert opinions – beginning with a detailed look at the lessons learned by Meffert and his team in New Orleans.

One story headlined "What you need to know about VoIP peering" delves into relationships between carriers in which they agree to exchange VoIP traffic to keep it on IP backbones instead of the public switched telephone network. Doing so allows carriers to cut costs, improve call quality and provide enterprise customers with new IP-enabled services.

"The numbers being thrown around in the industry are that only 2% to 4% of all VoIP traffic goes through a peering service," says Mario Galvez, vice president of marketing for Switch and Data, which in June announced VoIP peering exchange services for carriers. "We're going to see greater cost reductions in the future as the ratio of VoIP peering calls increases."

The guide includes a number of reports from the front lines of VoIP deployment – the voices of experience – and they address a variety of scenarios, including migrations from one VoIP provider to another, and even implementation of systems based on open source code.

Visiting Nurse Service (VNS) of New York and the public schools in Saskatchewan, Canada explained how they are saving big bucks by ditching leased Centrex lines and bringing telephony in-house using VoIP.

"We were spending quite a pretty penny per month on [Centrex, and] the flexibility wasn't there," says Randy Cleghorne, director of IT planning and management at the VNS. He says the move is saving almost $1 million annually.

Other stories address speech quality issues, and, of course, security. Included in the latter is a look at VoIP troubleshooting tools and the dangers they can pose in the hands of hackers.

The second section of this guide is devoted to a Network World Clear Choice Test of VoIP analysis tools. Six vendors accepted our invitation, submitting seven tools. The vendors included Apparent Networks, ClearSight Networks, Empirix, Fluke Networks, Touchstone Technologies and WildPackets.

Our testers write: "If you can't see into the VoIP traffic on your network, then you don't know whether it's good or bad. To know whether voice-quality or call-connect issues are related to your VoIP IP/PBX system or are tied to underlying network issues, you'll need to turn to the evolving class of network monitoring products called voice-over-IP analysis tools."

Finally, we conclude with a collection of Network World opinion pieces that address issues such as the best way to balance MPLS and VoIP, the fallacy of blaming VoIP for the trouble of Vonage, and an analysis of the major VoIP-related deal between Microsoft and Nortel.

All in all, there's a lot here to help get you where you're going.

Section 1

Industry trends

Eye of the hurricane: New Orleans prepares

Disaster-recovery plan following Katrina based on VoIP, Wi-Fi

■ *By Tim Greene*

If a Katrina-strength storm were to hit New Orleans tomorrow, again wiping out the entire traditional telecom infrastructure, CIO Greg Meffert says the city would be able to respond quickly, thanks to innovative disaster-recovery plans based on Wi-Fi and VoIP.

Four city-owned SUVs outfitted with Unisys Mobile Communications Hubs that include wireless gear and satellite dishes mounted on the roof would hit the streets as soon as the roads became passable. The SUVs, equipped with solar-powered batteries, would act as mobile network operations centers (NOC) designed to keep city officials connected to each other and the outside world over 756Kbps very small aperture terminal uplinks. "It's a NOC that can create a small Wi-Fi cloud around it for phones and laptops," Meffert says.

Residents with Wi-Fi-enabled laptops or VoIP phones would be able to access the Internet for free over EarthLink's 15-square-mile downtown wireless mesh network, a stipulation in the company's contract with the city that Meffert insisted upon. "We needed it to be free, a baseline free service. We need it to be there during a storm," he says.

The recovery plan also calls for a backup data center 450 miles away - well inland - in Austin, Texas. Through an arrangement with the city of Austin, key New Orleans servers will occupy a rack in Austin's data center. That standby recovery center will take over as soon as weather forecasters place New Orleans in the "cone" of future hurricanes, the area a storm is most likely to hit. "When we're in the cone, we're going to roll all our servers to Austin, and we'll do that every storm," Meffert says.

Because the plan relies on Internet connectivity, the city has contracts with local ISP Data Sync, plus Cox and Level 3 Communications, and each provider uses a different path to the Internet.

In addition, the municipal Wi-Fi network that covers about 4 square miles downtown, including the French Quarter, Central Business District and the Ernest N. Morial Convention Center, will connect to a tower on the tallest building in the city, One Shell Square, which will link via an optical point-to-point connection to a Level 3 Internet point of presence in neighboring St.

STEVEN VOTE

Charles Parish, Meffert says. The city plans to install generators stocked with weeks worth of fuel to keep the municipal Wi-Fi cloud running.

Lessons learned the hard way

Meffert's decision to base his disaster-recovery plans on Wi-Fi and VoIP is a direct result of the lessons learned after Katrina, when, for example, getting a simple phone connection so the mayor could talk to the president was a major challenge.

Meffert gained national attention immediately following Katrina when he went into a looted Home Depot, grabbed a Cisco router, found a single live connection at the Hyatt Hotel where the mayor was holed up and used a Vonage client to connect the mayor with President Bush.

But that was the least of Meffert's challenges. In Katrina's wake, he found that there was no plan to reestablish public communications, so residents could call out to let loved ones know how they were. And the city lacked the infrastructure to process the overwhelming volume of damage assessments, claims, permits and other bureaucratic forms needed to rebuild.

Not that the municipal government was caught napping when Katrina hit. It's just that nobody was prepared for a storm of that destructive magnitude, a storm that took down absolutely all networking, not just for days, but for weeks. "It was a shocker for me," Meffert says. "If somebody knows a storm is coming, you expect they'll have what they need to do their job. We didn't even have food and water and plumbing."

He adds, "You rely on telecom vendors to provide certain services, and you assume if they go out, they're going to come back," he says. "If people were working at 911 dispatch desks, they're going to come back. If you have a diesel backup that kicks in and will run for a day and a half - it's plenty if you're going to get more gas."

But nothing came back. Telecom switching offices were destroyed - at least one was washed away without a trace. Emergency workers lost their homes and didn't return to work. Generators that weren't flooded ran out of fuel, and there was no way to get more.

Putting it together on the fly

Awash in chaos, Meffert and his team improvised, enlisting whatever technologies they could find to patch together rudiments of a network that could immediately address some of the city's needs.

For example, before Katrina, the city had deployed Wi-Fi access points to support police with 130 surveillance cameras scattered in high-crime districts. When power was restored after the storm, many of the access points came back to life. The city then expanded the Tropos-based mesh, linked it to the Internet and opened it for residents to make IP phone calls via VoIP software from Skype and Vonage, as well as send-and-receive e-mail and instant messaging.

"It's obvious why it's so much more reliable than the phone network," he says. "We're sitting in 8 feet of water, and we look up and what's there? The telephone poles with the access points mounted on them."

In future emergencies, VoIP will take the place of traditional telephony for the city of New Orleans recovery effort, Meffert says. City calls will be routed from laptop voice clients or VoIP phones through a Cisco Call Manager server, which can connect to other IP phones or to the public phone network.

This will be particularly useful when outside agencies try to contact the mayor, for instance. As long as the mayor stays near the IP phone associated with his emergency contact number and the phone stays connected to the Internet, the network will find him. With the three new Internet connections, that shouldn't be a problem.

Because making these calls relies on the Call Manager, one priority is making sure the Call Manager server is running and accessible. During Katrina, power to the New Orleans city data center on the third floor of City Hall

New Orleans residents wait in line to use an automated system that speeds up the permitting process.

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was out for weeks. Fortunately, key data was backed up in a California data center run by ACS, and the city's Web portal was hosted on servers at Red Carpet Host in Dallas, allowing workers to carry out minimal city business.

But that arrangement still left public-facing applications normally accessible via the portal unreachable. They were on servers back in City Hall, without power and unconnected to the Internet, Meffert says.

Automating the permit process

Once the waters subsided, the city needed to get people to rebuild, which involved applying for federal funds and issuing permits for demolition, building and renovating, says Michael Centineo, the city's director of safety and permits. Improvements that came out of the emergency effort to meet those demands will streamline administration permanently, he says.

Before the storm, the city had set up a Web portal to allow residents to fill out permit applications online, using an application from software vendor Accela. Online or at touch-screen appliances in the hallway outside the permit office, applicants could enter their own data before talking to city staff and, depending on what they entered, be placed in the appropriate queue for personal attention. This procedure paid off by saving people from hours of needless waiting, because applicants discovered quickly whether they lacked information they needed to retrieve.

The problem was that the maximum load on the permit system pre-Katrina was about 35 per day, and demand peaked at about 500 per day afterward, Centineo says. At the same time, budget cuts reduced permit staff from 127 to 60, he says, and only by expanding permit automation did his office keep up. "This department would have been frozen and inactive, because it would have been overwhelmed by the volume of requests," Centineo says.

Immediately after the storm, city programmers customized the building-permit interface of the Accela software to make it easier for untrained citizens to use, says Peter Bodenheimer, a city IT project manager who oversaw the code writing. "We like to go with custom code, so we don't have to rely on others' timetables," he says, "and we needed this done. We had people who were screaming."

Meffert says the city tried to make the user experience require as close to zero training as possible, because the system simply had to work; there was no alternative. "We had to go to the extreme and have the user compatibility of Disneyland. You walk in, and you have a greeter to escort you to the kiosk box and explain how it works," he says.

Hitting the streets with Toughbooks

Eight weeks after the storm, the waters were gone and the city had to figure out the extent of damage and what buildings were safe to repopulate, Meffert says. That called for a whole new layer of bureaucracy for which the city had no infrastructure. "We had to inspect 110,000 homes in six weeks. There was only one way: create a more simplified inspection process," he says.

So the city wrote its own damage-assessment application for gathering information the Federal Emergency Management Agency would want and loaded it onto 34 Panasonic Toughbook laptops with wireless cards and global-positioning capabilities that inspection teams carried into devastated neighborhoods.

The laptops could call up digital records of individual properties via the Wi-Fi links, and inspectors verified and updated the data. The records had been geo-coded so records of property could be called up based on the location of the laptop as calculated by GPS. An inspector sent to a parcel of land where a building was washed away entirely, for instance, could still find the records of what had been there.

The updated database was made publicly available via the city Web portal and public kiosks. "That's how we told people if their house was inspected or not," Meffert says. "We had to create new [Wi-Fi] hot spots that would sync with the database." The database was also linked to a city map to differentiate inspected sites from uninspected ones, so inspectors could see where to go next.

Residents could use online access or public kiosks to find out the damage assessment for their property, and staff was available to resolve disputes. Residents could fill in a Web form from home and get a reservation number to get them to an agent at a set time the next day.

With its success automating and Webifying building permits and the damage assessments, the city plans to add electrical, mechanical, air-conditioning, heat and gas permits to the Web process, Centineo says.

Self-service kiosks for the French Quarter

Another goal is to place weatherized kiosks outdoors in the French Quarter. "We've seen what self-service can do," Bodenheimer says. Residents will be able to pay property taxes, parking tickets, research real estate transactions or find out what licenses have been issued for the construction going on next door. Whatever is available via the Web portal will be available at the kiosks. "Anything you can do that's city business online, we want to do with a kiosk. We want to make sure every citizen is served, not just those with computers," Bodenheimer says.

The program is being expanded to place kiosks in trailers in neighborhoods, so those without transportation or Internet access can get what they need within walking distance of home. The holdup is the trailers. "It seems like we're always waiting on trailers," Centineo says.

Since the storm, the city has also expanded the public interfaces to include a place to report new classes of crime, Meffert says. For instance, architectural theft - doors, mantels, windows, fences that are stolen from abandoned houses - is epidemic, he says. Residents can report this looting online to a city database. Using an administrative interface, police can log on to search reports of thefts in their districts and draw maps marking the hardest-hit neighborhoods and so better plan patrols.

As the new hurricane season plays out, Meffert says he hopes the disaster preparation the city has made proves successful, not only for New Orleans, but as an example for others. "It's actually a symbol not of where we've been, but where we're going," he says. "People can look back and say New Orleans was leading and not just in po' boys and beer. It was something pretty important."

What you need to know about VoIP peering

■ *By Carolyn Duffy Marsan*

As companies install VoIP systems, they're looking for ways to keep more of their voice traffic on IP networks to avoid the toll charges associated with the Public Switched Telephone Network. VoIP peering among carriers promises to make end-to-end VoIP calls a reality.

The VoIP peering market is still in its infancy, with carriers testing third-party services and joining exchanges to handle voice peering. In the United States, VoIP peering services for carriers are available from NeuStar, Stealth Communications, Switch and Data, and Tello.

Carriers are starting to offer end-to-end VoIP services to their corporate customers that take advantage of peering. Global Crossing announced in September that it was peering with VoIP service provider SunRocket. Global Crossing plans to sign other VoIP peering deals during the next year.

For corporate network managers, VoIP peering is promising, but it is still a year or more away from reality. That's because none of the top-tier carriers in the United States, including AT&T or Verizon Business, offer VoIP peering.

"The numbers being thrown around in the industry are that only 2% to 4% of all VoIP traffic goes through a peering service," says Mario Galvez, vice president of marketing for Switch and Data, which in June announced VoIP peering exchange services for carriers. "We're going to see greater cost reductions in the future as the ratio of VoIP peering calls increases."

VoIP peering refers to relationships between carriers in which they agree to exchange VoIP traffic to keep it on IP backbones instead of the PSTN. By keeping voice traffic on IP networks, carriers can lower costs, improve call quality and offer new IP-enabled services to their enterprise customers.

"The first step of the IP revolution is taking your traffic and making sure it stays on IP," says William Stofega, research manager for VoIP services at IDC. "If the traffic starts going through gateways and terminating on the PSTN, voice quality will degrade. You get a superior signal in a controlled, on-net call."

All-IP calls also are cheaper. "Using IP from an enterprise perspective, the savings can be in the 20% to 30% range in telecom costs and operations," Stofega says. "If you can consolidate and keep your calls on-net, especially if you're using a WAN, there is significant savings."

Additionally, carriers can offer new features, including presence and mobility for end-to-end VoIP calls.

> **"We're going to see greater cost reductions in the future as the ratio of VoIP peering calls increases."**
>
> Mario Galvez,
> vice president marketing, Switch and Data

Presence is going to have "tremendous value" to companies, Stofega says. He says enterprises also are going to be interested in services that use an IP PBX to provide "the same desktop features on a cell phone. Mobility and presence are where I'm expecting to see new and different capabilities."

Stofega says these features will be available to enterprises in the next year or two, thanks to the growth he's predicting in VoIP peering.

Corporations need VoIP peering to interconnect the islands of VoIP inside their own organizations and among their organizations and key suppliers and customers. Enterprises could establish VoIP peering arrangements among themselves, but few have done that. Instead, industry observers expect enterprises to take advantage of VoIP peering through carriers.

"It comes down to the same old question: Do I need to do this myself, or will my service provider facilitate this in such a way that it is not cost-effective for me to do it?" says John Longo, senior analyst with Heavy Reading. "Agreements like the Global Crossing announcement is the trend that I would expect to see."

Carriers have two options in VoIP peering: To develop multilateral relationships through such exchanges as Stealth Communications' Voice Peering Fabric and NeuStar's SIP-IX or to develop bilateral relationships with other carriers as in the Global Crossing and SunRocket deal.

Measuring traffic in the billions

One VoIP peering exchange that is gaining momentum is the Voice Peering Fabric from Stealth Communications. Shrihari Pandit, CEO of Stealth Communications, predicts that the 2-year-old exchange will carry more than 100 billion minutes of voice traffic in 2006, up from 18 billion minutes in 2005.

"We're growing at a very fast rate," Pandit says. "We don't have any direct competitors. The alternative is to use a service like NeuStar or VeriSign has to offer, which works like a SIP proxy or SIP redirect service. But the customers need to have a SIP gateway for that."

Carriers that belong to the Voice Peering Fabric include China Telecom, Telecom New Zealand, XO Communications and RCN as well as VOIP providers Net2Phone and SunRocket.

The Voice Peering Fabric includes a directory lookup service based on the Enum protocol, which matches a telephone number with a corresponding IP address (see "Why Enum matters to VoIP peering"). Stealth Communications has 18 million telephone numbers in its Enum database, up from 1 million in 2004.

Freedom Health Systems is using the Voice Peering Fabric to improve the quality of its VoIP calls. Based in Deerfield Beach, Fla., Freedom Health Systems provides telecom services to hospitals, pharmacies and other healthcare providers.

A year ago, the company began offering its clients VoIP services in addition to the data services it offers over its VPN.

"We have a primary data center in Miami, which we hooked into the Voice Peering Fabric for reliable, high-quality transit to other carriers," says Will Glynn, director of VoIP services for Freedom Health Systems. "The Voice Peering Fabric gets us jitter-free transport to a handful of other carriers."

Carrier inconsistency

Glynn says he tried running VoIP calls over the public Internet but found that some carriers provided better service than others. He says call quality is much better and more consistent using the Voice Peering Fabric.

So far, Freedom Health Services has signed up a handful of its pharmacy clients to its VoIP service. "It's a replacement for their existing phone system," Glynn says. "They toss out their local key systems and call it a day."

Freedom Health Systems also takes advantage of the Enum lookups offered by the Voice Peering Fabric. If a VoIP call that it carries terminates to a number in the Enum directory, Freedom Health Systems can keep that call on an IP network end-to-end for lower cost routing. Freedom Health Systems also enters all of the telephone numbers for its clients into the Enum registry so they can receive inbound calls via IP whenever possible.

Freedom Health Systems handles only a tiny number of end-to-end IP calls through Enum.

"Honestly, I was expecting more hits with Enum than we are seeing, but we're not charging extra for it," Glynn says. "Enum doesn't get us that many advantages today but it promises growth in the future."

While Freedom Health Systems takes advantage of VoIP peering through an exchange, Global Crossing prefers bilateral agreements with carriers such as SunRocket.

"That was our first bilateral agreement, but we have others in the works," says Pat Reilly, Global Crossing's senior product development manager for VoIP services. "Most of these conversations will happen in the next six months. Probably, within 18 months we'll be done."

Global Crossing isn't joining VoIP peering exchanges yet, but it is cooperating with the Voice Peering Fabric. This arrangement was announced last week.

"We've joined the carrier alliance to provide data access to the different Voice Peering Fabric points," Reilly explained. "We've got this global MPLS network. If any of our customers want to reach the Voice Peering Fabric, we can provide that connectivity."

Global Crossing has been aggressive in VoIP peering. In June, the company announced Enterprise VoIP Community Peering, which means the company will not charge for calls when its enterprise customers of VoIP outbound services are calling customers of its VoIP local services.

"We have dozens of companies using these two services," Reilly says. "They're mostly multinational corporations deploying IP PBXes in their own premises or hybrid PBXes or VoIP-enabled routers. Many are starting with their call center facilities."

Reilly says this service, which involves a small share of its traffic, saves customers around a penny and a half per minute for VoIP calls.

"Lots of folks are talking about VoIP peering, but we are trying to stand up and move this all forward," Reilly says. "One of the reasons is that by eliminating access charges we can provide savings to our enterprise and service provider customers. But really we're also trying to drive up the number of IP end-to-end calls."

VoIP peering isn't that useful to enterprises until it reaches a critical mass and most carriers participate. How long that will take is unknown.

"It's very, very early in VoIP peering," Galvez says. "It's going to be slow in developing when it comes to the larger providers because lots of this is still being driven by regulatory issues. At the end of the day it's about dollars. As long as there is still a financial benefit for local exchange companies to terminate voice traffic on the PSTN, they are not going to participating in peering."

Galvez predicts that it will be two or more years before VoIP peering reaches a critical mass for enterprise customers to see dramatic cost savings.

"When talking to their service providers, corporate network managers should be asking about their philosophy about peering," Galvez says. "They should ask about how much of their traffic is being terminated on-net through their network or through a peering partner. And they should ask about how that is translating to true cost-savings."

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Industry trends

Back to TOC

Why Enum matters to VoIP peering

■ *By Carolyn Duffy Marsan*

Behind the scenes in VoIP peering services is Enum, a protocol that translates telephone numbers to related IP addresses (see "What you need to know about VoIP peering").. Most VoIP peering services use Enum databases to connect telephone calls from one VoIP network to another and to keep the traffic on IP networks.

"Enum is one example of a directory service that is essential to the industry as it deploys IP throughout the world and delivers a whole variety of voice, data, video and other content services," says Jeff Ganek, CEO of NeuStar. "Enum is particularly important because it enables the delivery of all these advanced services, which are possible because of the billions of dollars of investment the industry has made in IP technology, to billions of endpoints on the worlds' networks that are addressed by telephone numbers."

NeuStar has offered Enum services for three years and has 1.3 billion telephone numbers in its Enum database. Half of those telephone numbers are corporate and the other half are residential.

NeuStar's Enum service is bundled into SIP-IX, a new service that allows carriers to cooperate and keep traffic on IP networks.

"We're seeing a very significant explosion of additional services that are now being rolled out, like instant messaging, push-to-talk, video sharing, video telephony as well as premium content services," says Mark Foster, CTO of NeuStar. "Carriers need additional service elements to enable that traffic to work between IP networks. That's the suite of services of SIP-IX."

NeuStar is running a trial of SIP-IX with mobile, cable and wireline carriers. NeuStar won't identify the number or names of the carriers participating in the trial.

"Part of the value proposition of SIP-IX is the interoperability among different carriers, such as cable to wireless and cable to wireline," Foster says. Interoperability is "very important for fixed/mobile convergence."

Foster says that all the services in the SIP-IX trial use Enum and most involve VoIP.

"A few involve large enterprises as well as consumer services," Foster says. "The enterprises are looking for such productivity enhancements as dynamic call routing across multiple networks through multiple types of devices."

NeuStar executives predict carriers will introduce new end-to-end IP services based on the interoperability features of SIP-IX in the first half of 2007.

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> Mark Foster,
> CTO of NeuStar

University dumps Cisco VoIP for open-source Asterisk

Sam Houston State University replaces Cisco CallManagers, Nortel PBXs with Linux-based VoIP and messaging servers

■ *By Phil Hochmuth*

Some organizations consider taking the plunge off of big iron PBX platforms into IP telephony as being pretty daring, but that's nothing compared to what Sam Houston State University (SHSU) is doing. The south Texas school is boldly moving thousands of users off a Cisco VoIP platform to an open-source VoIP network based on Asterisk.

SHSU is in the process of moving its 6,000 students, faculty and staff off of Cisco CallManager IP PBXs and a legacy Nortel Meridian PBX over to Linux servers running Asterisk, which includes call processing, voicemail and PSTN gateway functionality. The driver for this project was cost, says Aaron Daniel, senior voice analyst at Sam Houston State University.

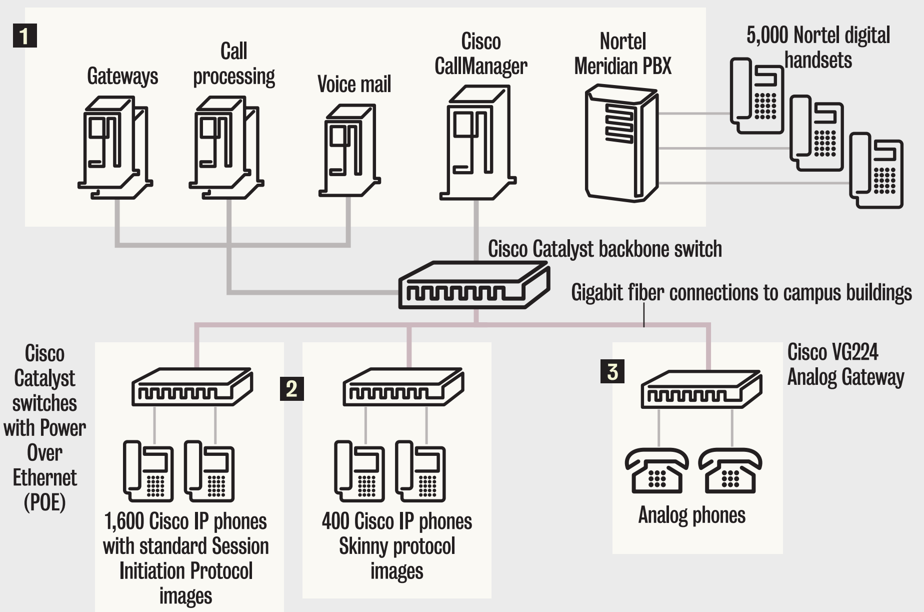
"We thought that it will be more cost effective in the long run to go with an open source solution, because of the massive amounts of licensing fees required to keep the Cisco CallManager network up and running," says Daniel, who this week gave a presentation on his migration project at the VON show in Boston. In the Cisco model, each phone attached to the CallManager required a separate licensing fee to operate, Daniel says. In SHSU's Asterisk/Cisco model, where it will keep its existing Cisco phones but attach them to Asterisk servers on the back end, the phone licensing costs are eliminated.

SHSU so far has moved 1,600 IP phones from Cisco CallManagers to Asterisk, which runs the IETF-standard version of SIP. The Asterisk functions are spread across six redundant Dell servers: two act as redundant PSTN gateways (and are outfitted with four-port T-1 cards from Digium, which commercially distributes Asterisk); two more servers handle call processing; another set provides voicemail.

The Cisco 7940 and 7960 IP phones the school had deployed were updated with a standard SIP software image replacing the proprietary Cisco Skinny Call Control Protocol (SCCP, or "Skinny"), which was used to connect the phones to the CallManagers. When the IP phones were upgraded with the SIP image about a month ago "all we had to do was reboot the phones," in order to register

Big VoIP on campus

Sam Houston State University is migrating its phone network off of a Nortel PBX and a Cisco IP PBX onto the open source Asterisk VoIP platform.



1 Redundant Asterisk servers handle call processing, public switched telephone network gateway and voice mail services in the data center.

2 Campus builidings are connected to the backbone via fiber. Cisco PoE switches support IP phones connected to Asterisk and Cisco voice servers.

3 Analog phones connect via a Cisco VoIP/analog gateway, while dedicated copper lines support the legacy Nortel handsets.

them with the Asterisk server, he says.

More control over the IP PBX software and servers was another reason SHSU made the Asterisk jump, Daniel says. "We felt we were more susceptible to hacks," since only Cisco-approved servers updates and patches could be installed on the Windows Server 2000-based CallManagers, he says.

Besides the phones, Cisco gear still comprises a large chunk of the IP telephony infrastructure at SHSU. The entire WAN and LAN is based on Cisco routers and switches. The Catalyst switches already installed support power over Ethernet (for powering IP phones) as well as QoS for voice traffic. All voice traffic on the campus network runs separate from data traffic in its own VLAN segment. Additionally, Cisco VG228 gateway devices, which can connect up to 24 copper/analog phones to an VoIP network, is used in dormitories and other areas where just a basic phone is needed instead of a more costly IP handset, Daniel says.

So far, SHSU has been able to operate the Asterisk/Cisco IP phones at one-third the cost of CallManager/Cisco IP phones, Daniel says. When the digital Nortel handsets are migrated to SIP-based Cisco phones, or analog sets, another large chunk of savings will come just by shutting down the electrical and cooling resources required to keep the old PBX running. "The Meridian takes up an awful lot of power itself. The room it's in has to be cooled to 60 degrees, and it has to have its own generator," Daniel says.

While Asterisk and the SIP protocol lack some of the more extensive features on the Cisco CallManager, the university community has handled the transition with few glitches. The only major feature missing in the Asterisk/Cisco phone network is secretarial functions, which allow an administrator to manage and answer phone extensions for multiple end-users. To fix this, Daniel is looking into extensions to the SIP protocol that allow for multiple-line handling, he says.

In another potential issue with open-source VoIP, SHSU loses the technical support from Cisco with its Asterisk migration. But Daniel says he has so far been able to keep up with support issues through mailing lists and the online community that develops and supports Asterisk. Dell provides support on the server hardware, and Digium supports the T-1 cards installed in the boxes.

"We try to have checks and balances," among the IT staff that supports the Asterisk system, Daniel says. "We try to keep the [the Linux and Asterisk server images] as pristine as possible." Daniel has also created copious documentation on all the Asterisk configurations and changes he's made to the software. "Basically if someone were to have to come in and take over my job, they'd have a pretty quick turnaround on learning what needs to be done," he says.

"We have a lot more peace of mind with the open-source system. If a bad exploit is found in SIP, we can fix it ourselves."

Aaron Daniel,
senior voice analyst, Sam Houstan State University

VoIP converts say good riddance to Centrex

Interop session details users' cost savings, productivity gains from VoIP switch-over projects.

■ *By Phil Hochmuth*

NEW YORK - Organizations still leasing Centrex phone lines could be wasting money and holding back advanced telephony and collaboration features from employees, said a pair of IT professionals speaking at Interop this week.

Two very different government/non-profit organizations - the Visiting Nurse Service (VNS) of New York, and the Public Schools in Saskatchewan, Canada - were both able to hack down their telecom and IT costs by bringing telephony in-house using VoIP instead of hosted Centrex services. The IT executives also said that the productivity gains and advanced features delivered by IP telephony and VoIP blew away what Centrex could do for them.

"We were spending quite a pretty penny per month per year on [Centrex, and] the flexibility wasn't there," said Randy Cleghorne, director of IT planning and management at the VNS. Over a year ago, the organization moved almost completely off Centrex to an Avaya-based VoIP system. The VNS now runs dual Avaya S8700 IP PBXs, which support 3,000-plus IP endpoints - mostly IP phones, with PC-based softphone clients mixed in. The IP PBX servers, which run a hardened Linux operating system, operate out of a centralized data center and serve VoIP clients in over 143 locations throughout the five boroughs of New York.

Cleghorne said the organization is saving around $900,000 a year after eliminating its Centrex costs. She added that along with renegotiating its service contracts for data and voice, the VoIP move brought VNS' IT budget down from around $4 million per year to nearly $2.5 million.

In New York City, where unexpected events are the norm, the flexibility the VoIP system offers - such as the ability to relocate phones and extensions quickly - is another key asset, Cleghorne said.

"We could move people to alternate locations in minutes vs. days," on the VoIP system, as opposed to Centrex, which required a technician call for every move or change, she said. The flexibility came in handy this summer, when a heat wave caused power outages in Manhattan and forced the VNS to move some of its workers to locations with more reliable electricity.

> **"We were spending quite a pretty penny per month per year on [Centrex, and] the flexibility wasn't there."**
>
> Randy Cleghorne,
> director of IT planning and management,
> Visiting Nurse Service

As for the Canadian school district, it moved off of a Centrex system, as well as dozens of key telephone systems in separate schools, to a Nortel-based VoIP network. At the core, the school chose the Succession Communication Server 1000 IP PBX - a VxWorks-based call server that uses a real-time, embedded operating system.

The district has 51 schools and two administrative sites, with more than 2,000 employees (more than 23,000 students are in the system). The burden in switching to VoIP was unifying all the separate phone networks that were built out over the years across 53 sites, sometimes by subcontractors who never considered someday linking all of the phone extensions together, said Daryl Koroluk, general manager of information systems at the school district.

"We had a growth rate that was huge," Koroluk said. The VoIP deployment, which took place a year ago, expanded the number of desktop phones from 900 to around 2,000 throughout the district. Voice mail, which was a rare luxury or a shared resource for teachers and staff in the past, also exploded - from 123 mailboxes to more than 2,300.

Koroluk has run into a couple of trouble spots with the new system.

For one, pricing can be complicated. Centrex costs were pretty straightforward, Koroluk said, as phones, extensions and voice mail were billed on a monthly basis. There are more surprises when it comes to licensing for VoIP phones, the software that runs the IP PBXs and the various features that can be added to parts of the system such as voice mail boxes, he said.

Another issue was the added electrical and cooling requirements that came with deploying power-over-Ethernet switches in wiring closets and IP phones in classrooms.

"In some locations, we're not operating what others might call an ideal environment for this kind of network and [VoIP] equipment," Koroluk said.

The school's IT staff tracks the environmental data on the Nortel switches and VoIP gateways deployed through agents built into the hardware's operating systems. These agents monitor the temperature and humidity of the gear and send alerts to administrators if things get too hot in the wiring closets.

VoIP: an invaluable asset

A converged net means unprecedented productivity for these bankers.

■ *By Beth Schultz*

When it comes to using advanced technology, 120-employee Goldsmith Agio Helms could teach the megasized investment banks of the world a thing or two.

Consider the sophisticated, converged network infrastructure over which the Minneapolis-based firm conducts its multinational business. Because it supports such advanced applications as instant messaging/presence, unified messaging, video calling and VoIP, the network lets bankers stay in touch and up-to-date at all times.

Unified messaging has proved particularly beneficial for Goldsmith Agio Helms, says Chris Ferski, vice president of IT at the firm. Using a simple baseline calculation, Ferski figures the new voice mail system saves 80% of people 10 to 20 minutes a day. That's because no one ever has to miss a message again. Employees get notices on their new handhelds when voice mail arrives. Such integration has elevated the level of customer service to an all-time high.

In essence, Goldsmith Agio Helms has become a virtual company capable of unprecedented employee productivity and customer responsiveness. As such, it earns a 2006 Enterprise All-Star Award.

A much-studied choice

Ferski began the network-upgrade project shortly after joining Goldsmith Agio Helms in the fall of 2002. "We needed to shave off time in everybody's workday and be more reliable. Those things are of significant value to us, and our old system couldn't support them," he says.

His goal was building a rock-solid foundation for voice, given the company's phone-centric nature. "VoIP was definitely in my mind from the get-go [even though] it was pretty new at the time," he says. "I wasn't sold on the idea that it had to be VoIP, but as we got into the project I realized it was the only choice."

To boost bankers' productivity, Ferski had to be able to integrate desk phones and computers, and enable on-the-fly mobility for phone extensions. That would have been tough without a true IP platform, he says.

To find that desired system, Ferski put five vendors through the wringer, rating them on their ability to deliver 24 critical items. Those included four-digit dialing, call routing, collaboration, network integration and management, as well as product longevity and long-term investment protection. Nortel blew away the competition, he says, scoring 21 out of a possible 24 points.

Sound investment

Last year, Ferski oversaw the firm's migration to an all-Nortel network. Besides traditional Ethernet switches for data, the network comprises the Nortel Multimedia Communication Server 5100 and the Nortel Communication Server 1000 system, which supports CallPilot unified messaging.

The firm spent between $200,000 and $400,000 on the overall infrastructure, Ferski says.

Based on productivity gains alone, Ferski says the firm will see a return on its infrastructure investment in less than three years.

"I wasn't sold on the idea that it had to be VoIP, but as we got into the project I realized it was the only choice."

Chris Ferski,
vice president of IT, Goldsmith Agio Helms

Wi-Fi/cellular convergence presents numerous challenges

■ *By Phil Hochmuth*

The convergence of desktop and mobile phones into a single, go-anywhere gadget that works on multiple wireless networks may seem just around the corner; however, real, large-scale deployments are still a long way off, according to IT professionals and vendors at Interop.

The concepts seem simple enough - a Wi-Fi/cellular device that lets you keep talking as you walk out of a building (using a Wi-Fi VoIP network) and into the parking lot (covered by cellular), or vice-versa. Complex hand-off technology between networks, varying standards in handset technologies and a reluctance by carriers to give up billable cell phone minutes to customers with their own VoIP-enabled wireless LANs could present some barriers.

What is known as Fixed/Mobile convergence (FMC), "will be a big plus for end-users, because they'll be able to just communicate," says Craig Mathias, principal of the Farpoint Group, a consulting firm, who led Interop's wireless conference track.

As users move in and out of Wi-Fi and cellular network zones, calls would be passed off among corporate WLANs and carrier cellular networks. "You won't really know what particular network you're using," he says. This will also radically change how enterprise communications networks are built, and the types of services enterprises buy to support them.

"You won't need a desk phone anymore," Mathias says, "or the copper running too the desk, or a pure-wireline voice carrier, or even a PBX."

This potential shakeup of the traditional business/carrier services relationship is where the high-minded concept of FMC meets reality, others say.

"We're still probably three years away before this starts to become a common service," says Alan Cohen, senior director of mobility solutions at Cisco. Right now, carriers are hesitant to help out in this area because there isn't much in it for them, he says. "Moving a call out of the cellular [cloud] and onto a campus Wi-Fi VoIP network takes billable minutes away from the carrier." This will take a huge push from users to get going. A Ford, GM or an IBM will have to start demanding this kind of service on a large scale.

One user who attended Interop this week is ready for fixed-mobile convergence now, but sees little support for it among carriers and equipment makers.

The campus at Northwestern Memorial Hospital in Chicago is as ready as any network for fixed-mobile convergence. The hospital recently installed a converged radio antennae infrastructure from Mobile Access, according to Dan Curran, IT director for the hospital. This technology combines cellular, 802.11, RFID and any other type of over-the-air communications the hospital may want to deploy. Cellular network providers - Sprint and Verizon among them - come in loud and clear through the hospital's halls, as the Mobile Access antennae amplify the cellular signals internally. The Mobile Access antenna infrastructure also feeds the hospital's Cisco WLAN access points, which are deployed centrally in wiring closets instead of spread though the campus.

> "You won't need a desk phone anymore, or the copper running too the desk, or a pure-wireline voice carrier, or even a PBX."
>
> Craig Mathias,
> principal, farpoint Group

To cut its cell phone bills, Curran recently started giving out Cisco 802.11 IP phones to doctors, nurses and staff. "They loved it," Curran says. "But one of the issues was that if they made a phone call in the building, as soon as they step out, it goes dead." On top of that, end-users we're not pleased with having to carry two handsets - cellular and 802.11 - when roaming around the campus.

"We just need that simple device," that ties a desk extension to a mobile device, and changes from 802.11 to cellular as end-users roam outside buildings.

There are no shortages of simple devices that provide dual-mode connectivity. Dual-mode devices are made by Motorola, Cisco, Samsung, LG and other makers. ABI Research estimates that shipments of these wireless handsets will exceed 300 million units by 2011.

Various products also exist that provide the cellular/VoIP handoffs, or some degree of dual-network tie-ins. Avaya offers an extension-to-cellular technology, allowing Motorola cell phones to log into a PBX network and impersonate in-office phones. Avaya is also working with carriers such as BellSouth on a hand-off service. BlackBerry-maker RIM has the Ascendant Voice Mobility Suite, which rings desk and cell phones simultaneously. Start-up DiVitas Networks offers an appliance that ties mobile and desktop phone extensions into a single device.

For other Interop attendees interested in fixed-mobile convergence, the devil is in other details for now.

The Visiting Nurse Services (VNS) of New York, which runs an Avaya VoIP network and a Cisco WLAN, uses the extension-to-cellular option, but Avaya's nascent Wi-Fi/cellular handoff had one problem. "It's all GSM-based," said Randy Cleghorne, director of IT planning and management at the VNS. All of the cell phones used by VNS employees run on CDMA mobile technology, which is not supported in the hand-off technology. "I would really like to see that cell option come along," she says.

Wideband audio boosts VoIP quality

■ *By Michael Ward*

Companies have rapidly deployed VoIP technology to reduce operational costs and increase productivity by integrating telecom services with other enterprise workflows. However, the codecs most widely used today that let speech be transmitted over IP data networks do not reproduce speech faithfully for a variety of dialects.

Typical human speech, while consisting of a wide range of frequencies, is very well reproduced by sampling frequencies up to 10KHz. Unfortunately, most VoIP codecs capture well under a half of this frequency spectrum. The G.729 voice codec, one of the most common codecs used in VoIP systems, is a narrowband codec that samples across a frequency range from about 200Hz to just under 4KHz. As a result much of the frequency spectrum of typical human voice is not captured.

To minimize the amount of data required to encode the sampled speech to limit the requirements of network bandwidth as much as possible, algorithms such as G.729 use a variety of encoding techniques that make certain assumptions about speech.

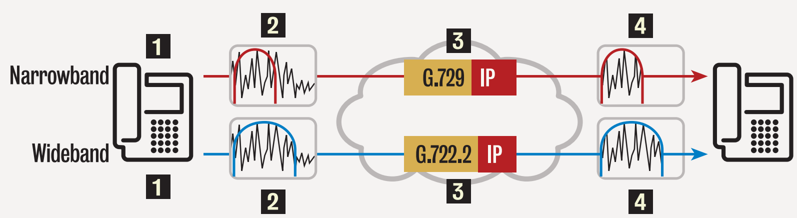
In many cases these assumptions have been made based on a hypothetical Western male voice. These predictive encoding techniques coupled with the limited frequency sampling result in a voice codec that poorly recreates Asian speakers because of the higher frequency content of many Asian dialects. These narrowband codecs also do a poor job in the transmission and encoding of music.

In an attempt to solve these problems, other codecs were defined that double the audio-frequency spectrum encoded. Wideband codecs sample frequencies up to 8KHz. By doubling the frequencies that are encoded, a truer representation of the speed or audio can be encoded and recreated on the remote end. As most speech can be represented with only 10KHz of frequency, these wideband codecs are able to more faithfully reproduce the original speech than narrowband codecs. Music transmission also becomes possible using wideband codecs.

The G722.2, or Wideband Audio Modem Riser, wideband codec was defined initially by the European Telecommunications Standards Institute/Third Generation Partnership Project as Wideband AMR for use in cellular and mobile applications and then ratified by the International Telecommunication Standardization Sector as G.722.2 for use in VoIP and other applications. G.722.2 supports nine bit rates from 6.6K to 23.85Kbps. Bit rates as low as 12.65Kbps still can deliver acceptable voice quality.

HOW IT WORKS: WIDEBAND VOICE CODEC

Wideband voice codecs double the frequencies that are encoded by narrowband codecs, providing a truer representation of speech.



Narrowband

1. Individual speaks into VoIP phone configured for G.729 (narrowband voice codec).
2. G.729 codec samples and encodes voice at 8KHz.
3. Voice is transmitted across IP network at 29.6Kbps (including headers).
4. G.729 packet is decoded at receiving end, with significant loss in original audio fidelity.

Wideband

1. Individual speaks into VoIP phone configured for G.722.2 (wideband voice codec).
2. G.722.2 codec samples and encodes voice at 16KHz.
3. Voice is transmitted across IP network at between 28.2K and 45.45Kbps (including headers).
4. G.722.2 packet is decoded at the receiving end with over twice the original frequency spectrum as the G.729 narrowband codec, resulting in significantly improved audio fidelity.

Real vs. perceived difference

G.729 requires only 8Kbps to encode voice, but this is before any additional data for Real-time Transport Protocol (RTP), User Datagram Protocol (UDP) and IP packet headers. When taking the RTP, UDP and IP packet headers into account, G.729 requires 29.6Kbps of effective network bandwidth and G.722.2 requires 28.2K to 45.45Kbps, depending on the rate used. With the largest G.722.2 data rate, just over 50% of network bandwidth is required to achieve twice the transmitted frequency and the associated audio fidelity benefits.

To deliver the superior audio quality of wideband codecs, the algorithms used to implement these codecs typically are more computationally intensive and thus require more processing capacity for the VoIP endpoints.

As such, many existing VoIP devices may not be capable of supporting wideband codecs. Fortunately, as processor capacity continues to increase, the benefits of wideband VoIP are becoming an option in an increasing number of devices at lower prices.

Ward, director of product line management at Trinity Convergence, can be reached at mward@trinityconvergence.com.

Open source router tempts company to bid firewall farewell

Vyatta airs Linux-based routing appliance based on Dell hardware.

■ *By Phil Hochmuth*

As open source router start-up Vyatta formally launches its first appliance this week, one early adopter is using the Linux-based gear to replace Cisco technologies.

Vyatta is offering its Open Flexible Router (OFR) software preinstalled on Dell server appliances. One of these devices is scheduled to come online this week on the network of Adify, a small San Bruno, Calif., company that provides advertising services for Web publishers and advertisers.

"We're taking several measured steps with the Vyatta product" to replace some pieces of Cisco infrastructure, says Charles Stewart, vice president of operations for Adify.

The first step took place this summer, when the company moved its Asterisk VoIP system - which serves 35 internal users and 10 remote users - to a Vyatta box on a separate network segment. Adify was running VoIP and data traffic through its Cisco PIX firewall and a 2600 access router, but decided to segregate the networks for better voice quality.

The company's next step, which is set to take place this week, involves deactivating its Cisco PIX and replacing it with a Vyatta OFR appliance, which includes stateful packet inspection firewall features. The company's network manager, Thor Myhrstad, built its first Vyatta box from scratch, but says, "We wanted to move quickly on (removing the PIX), which is why we chose to get Vyatta as an appliance this time."

The new Vyatta appliances are based on Dell PowerEdge 850 servers - a single-rack-unit box with a 2.8GHz Intel Celeron processor, 256MB of memory and dual Gigabit Ethernet ports. A single PCI-X slot is available for the various WAN interface cards that can be preinstalled on the router. (Vyatta uses PCI-X-based T-1 and T-3 cards from Sangoma). The $1,800 package includes a one-year support subscription for OFR, which is based on the eXtensible Open Router Platform, an open source routing stack, and a hardened Linux operating system.

Adify also plans to build a backup WAN based on an OFR router and DSL links. This network will be used in the event of a failure on the dual T-1 lines it rents from provider XO Communications, Myhrstad says. Because Vyatta supports the Virtual Router Redundancy Protocol, the backup network can take over immediately in case of an outage, Myhrstad says. For now, the Cisco 2600 router will stay, connecting dual-T-1 links, because that box is provided and maintained by XO.

> **"Commodity hardware is really an effective alternative for a certain class of the market."**
>
> Dave Roberts,
> vice president of strategy, Vyatta

Down the road, the company is considering Vyatta/Dell products as a possible replacement for the Foundry Networks load balancers it uses in its co-location facility, where the company's revenue-generating ad servers and billing systems operate. Adify's Stewart says he is talking with Vyatta developers about a load-balancing module for OFR for some time in the future. If such a module were available now, the company would be using it, he says.

While small firms such as Adify can often move more aggressively than larger organizations into open source network technology, Vyatta executives acknowledge the technology is not ready to take over the networked world yet.

"Commodity hardware is really an effective alternative for a certain class of the market," says Dave Roberts, vice president of strategy for Vyatta. "I'm certainly not going to put one of these Dell (servers running Vyatta) into the Internet backbone over at UUNET," he adds, but the Vyatta/Dell product has more than enough horsepower for users requiring T-1/T-3 WAN speeds.

Stewart agrees with this notion.

"I remember the first time I opened up an F5 box, maybe seven years ago," he says, referring to an older version of the load-balancer company's product. "I said, wait, this is just a PC … all these devices are just a network interface and an application doing some stuff behind it."

The Dell/Vyatta package costs $1,800, which includes a one-year subscription to Vyatta's online technical support and software updating/patch service. One-year contracts for software support alone cost $500, or $650 with online and phone support. OFR also can be downloaded for free from the company's Web site.

VoIP security still spotty, says Juniper expert

■ *By Network World staff*

A frank assessment of the state of VoIP and security technologies came from Gregory Lebovitz, technical director and solutions architect at Juniper Networks, at an Interop session on Tuesday.

"No [intrusion-prevention system] or firewall vendor supports all VoIP protocols and technologies," said Lebovitz, whose company's products claim to offer a measure of security for VoIP nets. "If [security vendors] are telling you that they support all VoIP technologies they're lying. There just isn't anyone who supports everything today."

Lebovitz said users must ask security product vendors what specific VoIP equipment and protocols are supported on their intrusion detection/prevention systems and firewall; product names and numbers should be asked, and users should test these combinations of security and VoIP gear in a lab before buying.

"It's not that we're not trying; we want to get there," Lebovitz said. "But there's only [so little] resources being devoted to write [so much of the] code that will be needed to get there."

Cisco's Chambers moves into world politics

While CEOs from Juniper and CA gave keynote talks the Interop conference in New York this week, Cisco CEO John Chambers apparently has better things to do - such as rubbing elbows with world leaders at the Clinton Global Initiative conference, which is taking place in the city this week. Chambers was expected to join First Lady Laura Bush, Bill and Melinda Gates, Afghanistan President Hamid Karzai, Lance Armstrong and 900 other invited guests at Bill Clinton's high-powered powwow, with topics such as world poverty, climate change and religious and ethnic conflict on the agenda. Chambers and his entourage were seen Wednesday morning entering the Sheraton New York Hotel and Towers, where the conference was held - and where a significant portion of Interop attendees were staying. Show-goers said getting a cab to Interop's venue at the Javits Centerl was challenging, as security personnel and black SUVs made a tight ring around the 7th Avenue hotel.

WAN optimization vs. thin clients

One Interop session pitted the optimization and thin client options against each other as means to "slim down branch offices" and reduce poor performance to distributed locations, especially for companies that have consolidated data centers or centralized their applications. Despite a discussion that had panelists debating the merits of tunneling traffic, compressing encrypted content and caching dynamic data, no technology came out on top.

> **"No [intrusion-prevention system] or firewall vendor supports all VoIP protocols and technologies."**
>
> Gregory Lebovitz,
> technical director and solutions architect, Juniper Networks

The panel included speakers from Silver Peak, Expand Networks, Riverbed and Citrix.

One audience member representing Cisco was able to add his thoughts on the growing market, albeit with some healthy competitive digs from panelists. Cisco recently joined more than a dozen vendors battling for WAN optimization dollars with its Wide-Area Application Service – if with some healthy competition. While some criticized Cisco for coming a bit late to the market, others pointed out the market is immature and bound to be dramatically different within three years.

Eric Siegel, a senior analyst with research firm Burton Group, moderated the panel and said WAN optimization technologies will help users. "The payback on the technology will blow your socks off. Socks will be in the air," he told attendees. "Even if it's obsolete in a few years, the results you can get today with these technologies will give you much happier end users."

Application networking, not the other way around

A majority of network managers polled today would probably reveal that along with their responsibilities to keep network available and performing well, they must also keep an eye on application performance. A panel at Interop Tuesday featured representatives from Crescendo Networks, Foundry Networks and Cisco, each discussing how their respective product has the ability to accelerate, secure and optimize applications on the network. The vendors detailed the challenges network managers face today trying to keep sophisticated multitiered applications running smoothly.

Hooman Beheshti, Crescendo vice president of technology, discussed how technology that understands application behavior - which his company loads onto application front-end boxes - will speed processing times.

"If networks can better understand specific applications and anticipate application requests, then networks can help speed content generation," he said.

Foundry supported adding intelligence to the network, but more for security purposes.

"The past few years, we really shot ourselves in the foot by deploying many application and networking technologies that are complex and can be uncontrollable. Things like wireless, mobile, VoIP and Web applications must have been wondering how do I control my network," said Gopala Tumuluri, director of product marketing at Foundry. Tumuluri said to get the control, or security, needed on advanced networks supporting such applications, "you are going to need more than just a packet-forwarding switch" and must embed intelligence into the network devices.

"If you put more application intelligence in the network, you reduce the number of servers you have to maintain and you can increase productivity in a few areas, your own and your employees' productivity," said Mark Weiner, director of data center solutions for Cisco.

Despite the difficulty in understanding application behavior on today's networks, the panelists agreed that network vendors should load their gear with more intelligence to better support applications - but not have them take on the entire burden of a server.

"You do not want network-centric devices to move up to where they will inherit the server problems," Tumuluri said. Nor do you want application developers getting the privilege of tweaking networks.

"I don't think any of us want application developers looking at router configurations," Beheshti said. "I don't want application code residing in the network, but there are things that can be done to make the application run better on the network."

VoIP troubleshooting tools can cause hacker trouble, too, expert says

■ *By Tim Greene*

A range of free network troubleshooting tools, some specifically designed for VoIP, can also be used as hacker tools that pose threats to IP-voice deployments, according to an expert speaking at a VON seminar.

These tools can imperil the confidentiality of VoIP calls and open them to being altered along the way, says Mark Williams, vice president of Tactical Security, which advises and trains businesses in IT security.

Taken as a group, threats to VoIP confidentiality make it more vulnerable than traditional phone technology. "The idea of a private line really does not exist in VoIP," he says.

Tools meant to analyze traffic can zero in on real time protocol packet streams for eavesdropping or be copied to files that can be listened to later, he says. Voice over misconfigured IP telephony is a tool designed specifically to do this, but the analyzer Cain can do the same and has recently been updated with a VoIP-specific tool.

Similarly, Wireshark captures traffic and via a more cumbersome route produces the same results, he says. "Every tool can be misused," he says.

In addition to capturing conversations, these tools can capture signaling that can yield addresses of IP phones. This data can be used to divert calls or to enable unauthorized use of the VoIP system, Williams says.

> "Every tool can be misused."
>
> Mark Williams,
> vice president, Tactical Security

Most VoIP threats are threats to data networks in general and warrant the same types of protections. Putting VoIP traffic on a separate virtual LAN (VLAN) can insulate it from certain denial-of-service attacks, in addition to freeing it from competition for bandwidth on VLANs also used for data traffic, he says.

Customers that own VoIP gear should carefully go through configuration to make sure the default settings - designed for ease of getting the VoIP system running - are changed to secure the system as much as possible, he says.

Classes of users should be identified and templates granting each the fewest privileges they need will help guard against inadvertent disruption of the network through user error as well as reduce the opportunities for attacks that compromised phones present to hackers, Williams says.

Phones should be chosen to give the least functionality required, as well. For instance, phones with a switch built in to support plugging a PC and a phone into the same Ethernet jack should not be deployed in places where there is no need to plug in a PC. They are a vulnerability that could be exploited, he says.

Williams urges VoIP users to consider using secure protocols on their VoIP networks such as H.325 for IP-conferencing security, secure RTP for protecting against eavesdropping and unauthorized users, and security for session initiation protocol, used for signaling, via the IEEE recommendation RFC 2543.

Businesses also should train employees to recognize social hacking attempts in which attackers manipulate individuals into revealing information about the business VoIP network or actually granting them physical access to it, he says.

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Section 2

Clear Choice Test: VoIP analysis tools

Digging deep into your net with VoIP analysis tools

■ *By Anthony Mosco, Robert Smithers and Robert Tarpley*

If you can't see into the VoIP traffic on your network, then you don't know whether it's good or bad. To know whether voice-quality or call-connect issues are related to your VoIP IP/PBX system or are tied to underlying network issues, you'll need to turn to the evolving class of network monitoring products called voice-over-IP analysis tools.

CONTINUED ON NEXT PAGE

NetResults

Product	ClearSight Analyzer	Fluke OptiView Protocol Expert and OptiView Link Analyzer	Touchstone WinEyeQ	WildPackets OmniPeek Enterprise with VoIP
Vendor	ClearSight Networks www.clearsightnet.com	Fluke Networks www.flukenetworks.com	Touchstone Technologies www.touchstoneinc.com	WildPackets www.wildpackets.com
Price	$20,000 for all software and includes one year of updates.	From $9,500 for software; $20,000 for Gigabit-capacity probe appliance.	$21,600 for Software Professional version.	$26,490 for one console and five engines.
Pros	Ease of use, intuitive layout, excellent navigation and drill down capabilities; top monitoring and threshold response reporting.	Acquisition/interface cards allow effective physical insertion into available network connection types; good reporting filter and template availability.	Strong capture-and-analysis tool with good real-time monitoring capabilities; easy ability to drill down to more detail with minimum navigation.	Ability to perform multiple captures with unique filter settings simultaneously with many and detailed filter options; excellent enumerated capture Peer Map display.
Cons	Little or indirect WAN status monitoring.	Although detailed, has limited structure in navigation and view.	Limited historical reporting.	Displays are generally clear but static, not tailorable.
Score	4.3	3.9	3.75	3.75

Product	Fluke Visual UpTime Select VoIP Module	Hammer Call Analyzer	AppCritical
Vendor	Fluke Networks www.flukenetworks.com	Empirix www.empirix.com	Apparent Networks www.apparentnetworks.com
Price	$36,000.	$24,975.	$50,000, includes predeployment assessment.
Pros	Single point of management; rich assortment of monitoring options and reporting.	Excellent RTP capture and playback analysis; demonstrated supports of both IP & ISDN.	Ease of deployment, transparent to network boundaries; outstanding Reporting and Help system.
Cons	Limited capture-and-analysis capabilities.	Limited reporting of live traffic, limited report templates; limited alerting.	No live traffic capture capabilities; no live packet analysis.
Score	3.7	3.7	3.45

The Breakdown

The Breakdown	ClearSight Analyzer	Fluke OptiView Protocol Expert and OptiView Link Analyzer	Touchstone WinEyeQ	WildPackets OmniPeek Enterprise with VoIP	Fluke Visual UpTime Select	Empirix Hammer Call Analyzer	Apparent AppCritical
Configuration & deployment **20%**	4.5	3.5	4.0	3.5	4.0	4.0	3.5
Display **10%**	4.5	3.5	3.5	4.0	4.0	3.5	3.5
Traffic capture **10%**	3.5	3.5	4.0	4.0	2.5	3.5	4.5
Real-time **10%**	4.0	4.5	3.0	3.5	3.0	3.5	0.5
Diagnostics **20%**	5.0	4.0	4.0	4.0	4.0	4.0	4.5
Reporting **20%**	3.5	4.5	3.5	3.5	4.0	3.0	3.5
Advanced features **10%**	5.0	3.5	4.0	4.0	3.5	4.5	3.0
Total score	4.3	3.9	3.75	3.75	3.7	3.7	3.45

Scoring Key: 5: Exceptional; **4:** Very good; **3:** Average; **2:** Below average; **1:** Subpar or not available

Clear Choice Test: VoIP analysis tools

Since our last test of these tools, VATs have picked up more monitoring power and offer a deeper level of detail in their displays regarding the VoIP activity of your network. Degradation of your VoIP traffic can be monitored, investigated and resolved before users are aware of it.

In this year's Clear Choice VAT test, six vendors accepted our invitation, submitting seven tools. The vendors included Apparent Networks, ClearSight Networks, Empirix, Fluke Networks, Touchstone Technologies and WildPackets.

All products were tested in Miercom's lab using a detailed methodology to assess the tools in six categories (see How we tested VoIP analysis tools). The categories are configuration and deployment options, display and interface usability, traffic capture and real-time monitoring options (see story discussing the benefits of these data-gathering methods), diagnostics and trouble-shooting measures, and reporting capabilities and advanced features.

ClearSight's Analyzer garnered the Network World Clear Choice Award for its second year in a row. It topped our charts because of exceptional diagnostic tools and its advanced navigation and display features. Fluke's OptiView tool earned second place, showing strong in its real-time monitoring and reporting features. Here below is a product-by-product breakdown of how each tool fared when we plugged it in, turned it on and set it to watch our test network.

There is a general trend in this class of product to increase protocol coverage, redesign interfaces for improved visual highlighting and navigability, and include advanced features to quickly pinpoint and detail problem areas of VoIP activity and call quality.

Consider with this improvement that there is no one right or wrong way to implement a VAT. Most of the tools tested here could measure up in your network depending on the level of expertise required, the immediacy of problem resolution needed, the desire for proactive call-quality management, and the level of detailed capture data necessary.

To see how each product fared, please view the product summaries.

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Clear Choice Test: VoIP analysis tools

The right tool for the right network

■ *By Lynn Haber*

We found, and users also need to keep in mind, that VoIP analysis tools are oriented toward particular VoIP protocol environments such as SIP, H.323, Cisco's SCCP (also known as "Skinny"). Interpreting different call-control protocol sequences is difficult for any one VoIP analysis tool in general, because the messages vary considerably with the particular protocol used.

Additionally, there are two main ways of providing VoIP analysis - through real-time monitoring in which data is viewed on the fly or through data-capture processes where data is analyzed after the fact. Real-time monitoring shows current and/or summarized statistics on active VoIP systems and best serves as a sort of security guard, watching over VoIP traffic as it moves across the wire.

Data-capture and analysis methods record network statistics on VoIP traffic and save them for future, detailed review through the use of tools designed to enhance the analysis process. Filters - the selection criteria for catching and storing VoIP data for subsequent analysis - have often provided focus for the examination of the statistics. Most of the products tested here offer a mixture of real-time monitoring and data-capture processes.

VoIP tools can operate in one of two modes. It can be a passive "listener" to your network traffic, gathering statistics in real-time (or near real-time). Or it can monitor the network "actively" by sending out its own packet traffic, gathering and comparing the results of the test transmissions across the network.

Both approaches have advantages and disadvantages. Most of them directly relate to the network the VoIP analysis product will need to monitor. A passive tool adds no overhead to your network while in use and can monitor it for prolonged periods.

An active tool can gauge results precisely by comparing what it knows about the packets it created and transmitted compared with the results captured at the far end of the transmission. The passive monitoring mode is used by ClearSight, Empirix, Fluke OptiView, TouchStone and WildPackets. Fluke's Visual Uptime Select can operate in both active and passive whereas Apparent is strictly active.

How we tested VoIP analysis tools

The test bed was configured with two subnets simulating a headquarters and a branch location. The network infrastructure at both sites consisted of Extreme Networks Summit 48 L2/L3 switch/routers. The two sites, labeled A and B, powered by Zultys MX250 IP PBX systems, were connected by a T-1 IP WAN link.

Each VoIP analysis tool tested was configured and inserted, one at a time, into the test bed. The WAN connection was simulated using a Hurricane IP Network Emulator from PacketStorm Communications. The PacketStorm Emulator let us vary our network environment, simulating various impairments that included numerous latency, jitter and packet loss scenarios. A mirrored port was configured on the headquarter subnet for the vendors to insert their respective analysis tool.

An average of 5,000 channels of simulated SIP traffic was generated and delivered from the site A by Touchstone Industries WinSIP (2.4.0) Generator. An average of 128 channels of simulated SIP traffic was also generated from a Spirent Abacus 5000 (v3.2) SIP traffic generator from site A. These site A calls were terminated at respective matching endpoints across the WAN link, at the B site.

Each of the products was evaluated on its ability to capture, monitor and analyze these calls. All of the participating products varied slightly in the overall results on MOS, latency, jitter and packet loss, but all reported accurately on the effects of the network and the VoIP calls in progress.

Apparent Networks' AppCritical

Apparent Networks' AppCritical tool, an active-mode monitoring system, has an intuitive, linear interface that provides very efficient access to information.

Collecting information on network activity and reporting on these captured statistics is one of the product's strengths. Based on applying a very small amount of traffic consisting of hybrid Internet Control Message Protocol (ICMP)-style queries, it operates unobtrusively from a single installed location.

In most instances a single site installation is sufficient to monitor the entire network. In some highly secured networks that limit ICMP activity, additional software probes called remote sequencers might be necessary to overcome restricted boundaries. They are installed on the remote hosts. With its unique architecture, this was far easier to deploy in an enterprise network compared with the other VAT products tested.

In our traffic-capture category, which assesses each product's awareness of network conditions and VoIP endpoints, AppCritical leads the pack, because it was very strong in collecting the data necessary to identify certain network conditions (such as the loss of an IP WAN connection, call controller or gateway) that would affect VoIP applications. It also had a high degree of success in pinpointing the cause of degradation in call quality, and its expert-commentator-like interface helped diagnose network issues.

AppCritical has one of the best and highly developed help and analysis support interfaces we have tested. Reports and quality-assurance threshold alerts contain links to background information to assist in explaining the contents.

Where this product fell down was in its lack of real-time analysis tools.

ClearSight Networks' Analyzer

ClearSight offers both stand-alone and distributed versions of its Analyzer product. The distributed version allows for multiple sites to be simultaneously monitored, either individually or in aggregate, a condition that earned it high marks in the deployment category.

Analyzer's interface is intuitive and unique, displaying more network activity in one place by default than any other product tested. In addition, it is designed for drilling down to greater detail, without opening multiple separate windows in a desktop-type interface. The tool's default interface displays a summary graphic view of all active and nonactive protocol sessions on the network.

The VoIP protocols tracked by ClearSight include Session Initiation Protocol (SIP), Skinny Call Control Protocol (SCCP), H.323, Megaco, Media Gateway Control Protocol (MGCP), to name a few. Unknown protocols are displayed in a generic traffic category. The default navigation tabs include summary, detailed and combined-flow views.

Its reporting for monitoring and threshold responses is generally very good. We did have difficulty in detecting a duplex mismatch between sites at the router/switch level. That aside, Analyzer excelled in its ability to monitor our test network and was able to perform all the diagnostic tasks we required of it.

What distinguished the performance of this product was its ability to provide the administrator with the top-level information and then to drill down into and fix a reported problem. For example, in a summary view, the RTP traffic report can be used to display the detail of the RTP stream, playback any captured file, codec and/or call-quality detail without losing the visual or logical context of the tool navigation.

ClearSight Analyzer stands very strong on the scope of the audio and video codecs it can recognize and analyze, and also the ability to assess mean opinion scores through generated, simulated traffic and by monitoring actual user traffic.

In addition, Clearsight's Real Time ladder-view with TCP/IP and application-anomaly detection makes it easy to make changes and see the effect of the change without recapturing and recomparing traffic.

Clear Choice Test: VoIP analysis tools

Empirix's Hammer Call Analyzer

The Hammer Call Analyzer (HCA) is best used as a traffic-capture and diagnostic tool, not a real-time monitoring one. Its excels at detailing captured data for further investigation. The HCA's mode of operation is passive - listening to traffic rather than generating packets for transmission and comparison itself.

The Call Analyzer has the ability to display the actual sound waveform for both sides of the call, allowing visual analysis of problems between the various devices.

The HCA provides the most customizable, detailed level for setting up triggers. A trigger is a set of predetermined conditions that will start the capture of session data automatically. Triggers can be set for pre- and postevent actions on a given threshold or level of session activity. The fine degree of control for setting up triggers gives the tool a pseudomonitoring capability, such as continuous real-time.

The HCA offers a display of VoIP sessions, which serves up an effective capture view. The user can correlate and visualize call flows among any combination of the following protocols: SIP, Megaco, MGCP, ISDN, Signaling System 7 (TCAP and SCCP), H.323, T.38 and Simple Traversal of User Datagram Protocol. The Call Merge capability allows you to follow the flow from end to end.

Empirix also offers an optional ISDN card, which gives you a good look at your ISDN traffic, including such details as call setup and teardown. Additionally, the ISDN card allows the use of the Call Merge Map, which gives managers the ability to associate calls that change in protocol, such as a call that starts as ISDN, is converted to H323 and then back to ISDN. This is useful in a hybrid environment.

Empirix was a bit off in our reporting category, because it does not create preformatted reports of system activity, as do the other tools tested. What it does export are in .csv and .txt files, which are more like formatted data files than a report.

Fluke Networks: OptiView, Visual Uptime Select

Fluke Networks submitted two products for testing, its OptiView suite and the Visual Uptime Select tool, which was acquired in a recent acquisition of Visual Networks.

The OptiView product suite, as tested, consisted of the Protocol Expert and the Link Analyzer. The Protocol Expert is a software tool designed for use on lower-speed links (10/100Mbps) using a laptop. Typically, the Protocol Expert is deployed to capture and analyze VoIP traffic at the endpoint location.

The Link Analyzer tool is installed on the main uplink or core server-farm link where aggregated VoIP/data traffic traverses. Different network environments can be accommodated easily by built-in connections for the Link Analyzer, including 10/100Base-T, 1000Base-SX, 1000Base-LX and 1000Base-T.

The OptiView suite stood out in our Real-Time Features category, in which we assessed the level of real-time session detail that can be reported. OptiView has the ability to identify key nodes in the network by address and role, IP endpoints, call encryption recognition and the vocoder of a specific call session.

As far as diagnostics are concerned, OptiView automatically detects and identifies such network problems as loss of a gateway, controller or specific endpoint, and can detect call-quality degradation in latency, packet loss and MOS call-quality level.

Fluke's Link Analyzer tool also features escalating notification processed and customizable alarms when network conditions reach predefined conditions.

OptiView offers many preformatted reports of the VoIP statistics collected and offers links to third party reporting tools like Crystal Reports.

Fluke's Visual Uptime Select is a traffic-analysis and network-monitoring application capable of displaying real-time activity. It requires software agents to be installed at network monitoring points software agents to be installed at network-monitoring points to report VoIP traffic between the monitored sites back to a central administrative console. Overall, it has a strong inherent ability to report outages and error conditions on the network.

Visual UpTime is able to detect and report the loss of a WAN link, call controller or gateway with active alerts sent to its service summary screen, as well as report degradation in call-quality conditions (latency, jitter, packet loss).

The reporting capabilities are extensive, comprising a large library of customizable template reports. One of these template reports has basic statistical or metric fields, for all or specific sites or IP ranges, but it can then be altered to display just the information the administrator requires.

Its straightforward, simple interface is efficient in determining and highlighting any issues with the network. This interface also offered quite a bit of flexibility in filtering and sorting the collected data.

Clear Choice Test: VoIP analysis tools

TouchStone's WinEyeQ

TouchStone Technologies submitted the WinEyeQ Professional tool for testing, but the company also offers Lite and Probe editions, which scale to different levels of concurrent sessions that can be monitored. In addition to the SIP and H.323 focus it held last year, TouchStone has added to its list of supported protocols. They are MGCP, Megaco, HTTP, SMTP, POP3, FTP, real-time streaming protocol, SNMP and Telnet. WinEyeQ also can capture video as well as RTP streams.

Designed for monitoring only VoIP networks, WinEyeQ provides the most efficient and direct product layout to facilitate analyzing a VoIP environment. Using a tabbed layout for navigation, one can progress left to right to high-level network activity by category to tabs with more in-depth information on active calls, registrations, recent errors and user alerts, to name a few. Also new to WinEyeQ is a command-line interface allowing for script execution.

WinEyeQ offers a unique real-time SIP device interface that can contact, query and even control SIP device settings. Called the Test Peering Fabric, this application can send a message to SIP endpoints and query them for an XML-formatted real-time status on the device, or pull down a call-summary file containing 160 metrics (also in XML format) at the completion of the call. This Test Peering Fabric lets an administrator broadcast status requests to multiple endpoints and change option settings for SIP devices so they can be remotely controlled and reconfigured.

WildPackets' OmniPeek

WildPackets' OmniPeek data-analysis tool provides an optional module, Enhanced VoIP Option, to provide VoIP analysis. OmniPeek has distributed capabilities that comprise software probes installed on remote-network subnets that report back to the main analysis engine.

OmniPeek's strength lies in the deployment flexibility of its data-capture filters - the selection criteria for catching and storing VoIP data for subsequent analysis - and the detailed level of information they supply. OmniPeek has multiple options for setting what data to capture and the VoIP conditions that cause a preset capture to begin recording data.

The precision of the filters avoids the collection of a voluminous log that could add time and overhead to the debugging process in the VoIP environment. OmniPeek allows multiple unrelated captures to execute simultaneously, with different filters and initiation conditions set on each.

One of OmniPeek's best capture-and-analysis features is the graphical Peer Map display. This shows a diagram of VoIP sessions visually with both endpoints enumerated along with visual representation depicting the relative percent of network throughput used by each session over the span of the capture. Hovering over objects in the Peer Map will generate pop-up boxes with further statistics about the object.

A unique analysis feature for captured .wav files lets an administrator replay the statistics of a captured call with the replay of the call in the same screen.

A captured call under analysis can be examined step by step as the captured metrics are synchronously displayed on the screen. If the jitter or latency degrades in midcall, the replay can be stopped, in snapshot fashion, and all associated metrics examined for insight to the call and network environment.

OmniPeek did not show as strong in real-time monitor mode. The amount of VoIP data available in real-time monitoring is limited, because standard statistics such as packet counts, network throughput percent and detected error are available only in summary aggregation by protocol. There are no drill-down capabilities to further explore issues with a particular device or IP address.

Section 3

Opinion

Making the most of MPLS

■ *By Johna Till Johnson*

Most of my clients are making the move to MPLS. They're either actively considering MPLS-based services, testing them or embarking on a migration. I've learned quite a few lessons:

❏ Expect to save 25% to 40% compared with your existing voice, data and video services - but with some caveats. First, the greatest savings accrue to folks who combine all three (thereby eliminating such things as redundant ISDN networks for video). Second, the greatest savings apply to highly distributed, relatively low-bandwidth networks (that is, dozens or hundreds of T-1 sites). A network with relatively few high-speed sites (a dozen OC-3s, for instance) may not see the same savings. Finally, your mileage may vary: The savings aren't guaranteed. Some companies have found that because of previous competitive negotiations or unusual network configurations, MPLS doesn't provide any savings.

❏ If you're planning, along with rolling out MPLS, to converge voice and data via VoIP, tackle each project separately. Carriers recommend starting with MPLS and pursuing VoIP later; I wouldn't say you need to hold off on VoIP while launching your MPLS network, but make sure to devote enough resources to both. Keep in mind the impact VoIP will have on your service contract. Don't commit to a voice-only minimum annual revenue commitment, for instance, if you're planning to convert voice to VoIP during the duration of the contract. Plan the anticipated capacity of your sites appropriately - if you need to increase bandwidth to accommodate voice and video traffic, make sure your access circuits can handle the load and your contract doesn't include clauses that make increases prohibitively expensive.

❏ Get the scoop on network-to-network interfaces (NNI). Try to keep your sites on a single carrier's network - with the possible exception of having multiple carriers link to critical sites for redundancy - because effective MPLS NNIs are lacking. For starters, carriers have different service class definitions, so you'll have to map provider A's platinum class to provider B's Tier 1 class - and each has different characteristics. That's if provider B is offering MPLS at all: In some cases, MPLS NNIs are really handoffs to frame or ATM. Keep in mind that few providers will offer end-to-end service-level agreements to off-network sites. More broadly, the most effective cross-carrier connections are between providers that have previously established working relationships. In other words, each carrier has a short list of providers it works with - ask for that short list, as well as the details of the technical and business relationships. Plan to drill down, asking questions such as: What is the certification process for adding providers to this list? What service guarantees can be made, and how are they enforced? How is troubleshooting handled? The more you know, the more effectively you can manage crosscarrier connections.

MPLS, NAC shine at Interop

■ By John Dix

Although the New York version of the Interop conference didn't draw huge crowds last week, those that made the trek seemed pleased with educational sessions covering everything from network access control to MPLS (see complete Interop coverage).

The MPLS session was hosted by Network World columnist Johna Till Johnson, president of Nemertes Research, and featured speakers from Qwest, Sprint and Verizon.

Johnson said the main application driving her clients to MPLS is VoIP, a sentiment backed by speakers from Sprint and Verizon. But Martin Capurro, Qwest director of Global Product Management for IP Access, said it's broader than that: "We see customers wondering, 'How do I take frame, private line and voice and integrate them on one backbone?'"

Johnson told the crowd they could expect WAN savings of 25% to 40% by moving to MPLS, with the larger savings going to companies that bundle voice, data and video, have international sites and have the carrier do the management.

But a member of the audience who works for an investment firm said the savings disappear if you need big T-3 or OC-3 pipes. "SONET is still the cheapest alternative for us."

A second strike against MPLS for this buyer: His security group regards MPLS, which is a Layer 3 service, as being akin to the Internet and so requires encryption, which is demanding at the speeds in his backbone.

In terms of prepping for migration to MPLS, Qwest's Capurro told the audience the shift typically involves migrating frame to MPLS, private lines to Ethernet services, and voice to VoIP. "Make sure you have a view across all of the components," he said.

Another hot topic at the show was network access control (NAC). Network World Lab Alliance member Joel Snyder, a senior partner with Opus One, led a panel on the topic featuring representatives from Microsoft, Cisco, Juniper, StillSecure and the Trusted Computing Group.

Asked to venture a guess on what percentage of large companies will be doing full-fledged NAC in five years, all of the speakers agreed that it would be common by then.

So the question was, how do you best prepare for the arrival of NAC? Thomas Howard, security solutions engineer with Cisco, said you need to develop policy: "If you don't know what you want to do, how are you going to know what you need?"

But Dave Greenstein, chief architect with StillSecure, recommended a piecemeal approach. "There is so much bureaucracy involved in policy; I say start with your highest risk. Often that's laptops that come and go."

Steve Hanna, distinguished engineer with the Trusted Computing Group agreed: "Roll it out gradually, starting with users that are working with high-value assets."

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Opinion

Back to TOC

Don't blame VoIP for Vonage's belly-flop

■ *By Johna Till Johnson*

Remember Vonage? Back in May, the Internet VoIP company held its much-anticipated IPO, which investors and various pundits were predicting would showcase the triumph of "next-generation" VoIP services over the old, tired offerings of the Bells.

What happened instead was that Vonage's shares, which were initially offered at $17, promptly tanked (Vonage is trading at about $8 per share now). That's not all - Vonage is now apparently suing its own customers for failure to purchase roughly 1 million shares of its stock. The company recently sent "pay up or face the consequences" letters to roughly 9,000 of its customers who had promised to invest but backed out in the face of the disappointing IPO.

Talk about hilarious!

Not even the Bells have been boneheaded enough to think of suing their customers for shorting their stock. (And please, let's not give them any bright ideas, OK?) Unless you're one of those Vonage customers who got the letter, you're probably chuckling.

But there's a serious point here, too. It's a mistake to view Vonage's market belly-flop as a thumbs-down on VoIP technology. As I've pointed out in many previous columns, VoIP can potentially lower costs and improve agility for enterprises that deploy it correctly.

The real lesson behind Vonage's VoIP belly-flop is that service companies - regardless of the technologies they deploy - are fundamentally providers of, well, a service. In the telecom case, the service involves connecting people effectively. That means a great deal more than "using the latest and greatest technology." It means providing top-tier customer service and support.

Vonage apparently hasn't entirely figured that out. Complaints about its customer service abound. "Vonage is the roach motel of phone companies," writes one disgruntled ex-customer. "They have salespeople working around the clock but intentionally don't put their customer service extension in the menu on their phone system." Analyst David Andrews details a similarly negative experience (see www.nwdocfinder.com/5345), concluding that Vonage "builds negative brand equity" in the way it treats customers.

There are early indications that the company is addressing these problems - sort of. A few weeks ago, CEO Jeffrey Citron said the company was "improving customer service to help retain subscribers." Great move - too bad it didn't occur to the folks in charge until recently.

So there are several take-aways here. First, Vonage's competitors should keep in mind that the company's current rocky path doesn't mean VoIP-based services are down for the count. On the contrary, it's increasingly clear that VoIP will become the baseline infrastructure enabling a whole host of next-generation communication and collaboration.

Second, providers of all stripes (and technologies) should focus on offering world-class service and support. Technology's just table stakes. To cement long-term customer loyalty, providers need to meld cutting-edge technology with tried-and-true service and support.

And finally . . . Just for the record, Citron, suing your customers (however justified legally) isn't exactly a great way to win their undying loyalty. What's next? Tacking unspecified charges on to phone bills? Oh wait - that's Verizon.

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Tit for tat in convergence war

■ *By John Dix*

Microsoft last week climbed into bed with Cisco's largest competitor, Nortel, as it continues to battle Cisco for dominance in the emerging unified-communications market.

In a multifaceted deal, Microsoft and Nortel formed the Innovative Communications Alliance and committed to develop and sell unified communications products jointly. Although details were light, the companies said the four-year agreement calls for cross-licensing technology and forming a Nortel systems integration division to support the partnership.

Early development efforts will focus on integrating Nortel's VoIP products with Microsoft's client software. Nortel CEO Mike Zafirovski says he expects the deal will let Nortel generate $1 billion in new revenue through 2009.

Microsoft's battle with Cisco has been becoming progressively more pronounced. Last March Cisco put the companies on a collision course when it announced the Unified Presence Server and Unified Personal Communicator client software, which are analogous to Microsoft Live Communication Server and the Microsoft Office Communicator client. Cisco also released CallManager 5.0 - the heart and soul of the company's key VoIP product - and offered it on Linux. This is the first time CallManager has been available on anything but Windows.

In addition, just last month Microsoft announced a broad unified-communications strategy that hinted at IP PBX features Microsoft needs a telephone partner, however, and it didn't have to look far to find a suitable alternative to Cisco. Nortel is in desperate need of some fresh direction.

Like other VoIP players, Nortel is looking to make its IP PBX a software-only product; it can't hurt to team with Microsoft on that. Nortel's plan is to deliver telephony features as software modules that can be used with Microsoft's unified communications software platform.

Although vendor partnerships rarely deliver what is promised upfront, one reason to believe this one will be different is the competitive forces at work. Given the increasingly combative relationship between Microsoft and Cisco, the Redmond giant has real incentive to see this through.

Certainly Nortel also has something to gain. This relationship is deeper than those Microsoft has with other IP PBX vendors - which tend to deal with product integration and distribution - and should be significant in Nortel's efforts to expand its core VoIP business. Gaining access to all those Microsoft desktops will be a huge win.

Then there is the big "but": Regardless of motivation and intention, Microsoft can't get around the fact that Cisco owns a staggering percentage of the enterprise infrastructure market. Partnering with Cisco competitors can do only so much.