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The IP Communications Authority Since 1998™

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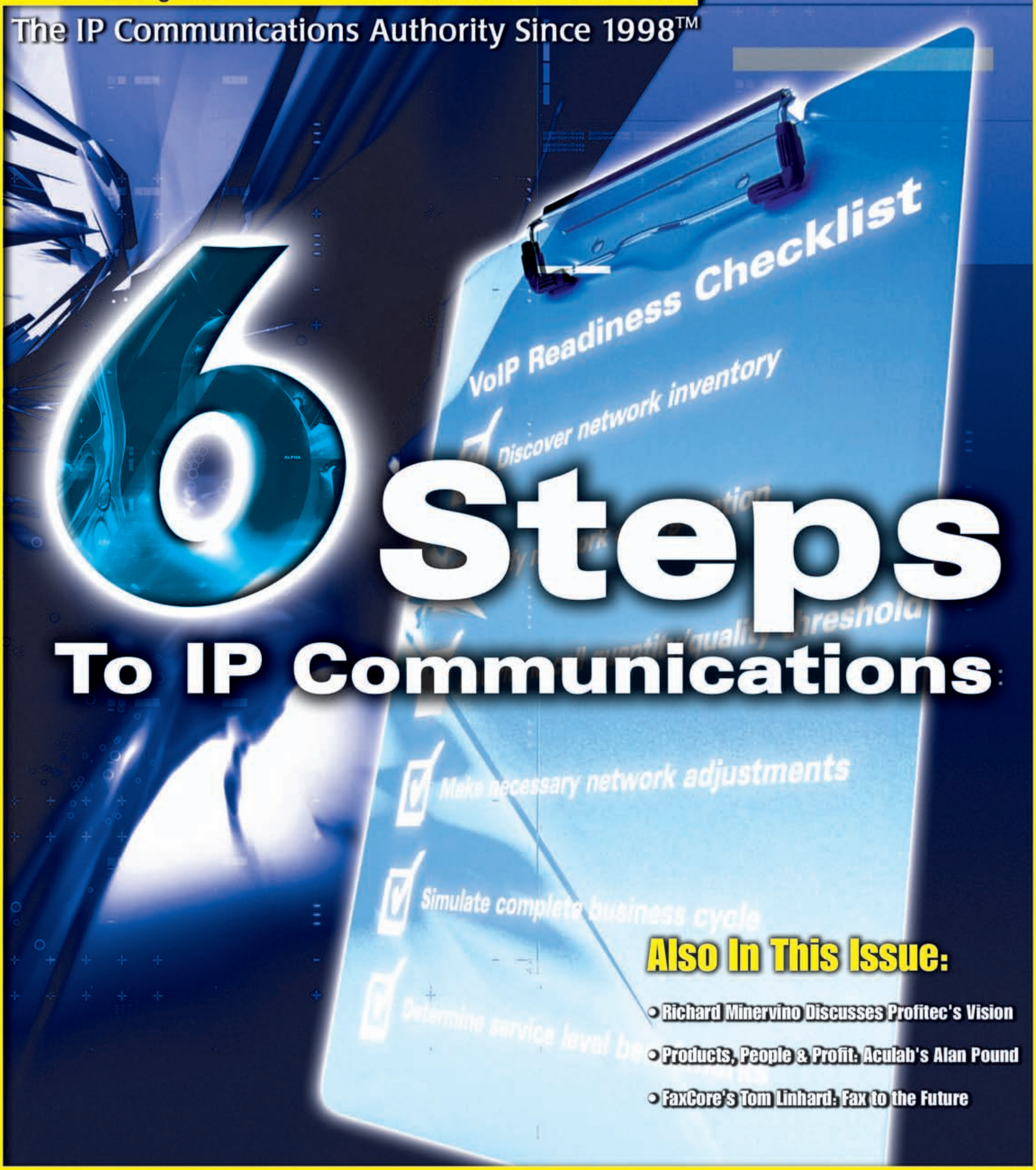

6 Steps To IP Communications

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- Richard Minervino Discusses Profitec's Vision
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- FaxCore's Tom Linhard: Fax to the Future

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Internet telephony is revolutionizing telecommunications through the convergence of voice, video, fax, and data, creating unprecedented opportunities for resellers, developers, and service providers alike. INTERNET TELEPHONY® focuses on providing readers with the information necessary to learn about and purchase the equipment, software, and services necessary to take advantage of this technology. INTERNET TELEPHONY® readers include resellers, developers, MIS/networking departments, telecom departments, datacom departments, telcos/LECs, wireless/PCS providers, ISPs, and cable companies.

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The VoIP Authority

By Greg Galitzine

Location, Location, Location

Let's take a trip back to the late 1990s, right about the time when every thing connected to telecom was seemingly awash in money, and there was WAP, and with WAP there was the early promise of location-based services.

Who remembers this WAP enabled application? You're walking down the street and, as you approach a Starbucks (or any other generic non-vendor-specific coffee house), suddenly, as if by magic, a coupon would appear on your cell phone's screen, enticing you to stop in for a 50% discount on an otherwise overpriced Latte. This, my dear reader, was going to drive the future of mobile telephony.

I scoff a bit, as I remember those heady days. But location-based services vendors, having spent a few years operating under the radar, are once again making headlines.

One such vendor, MapInfo Corporation, recently released its MapInfo Professional version 8.5, the first version of the company's location intelligence application with Web services connectivity. The solution is designed to enable users to access dynamic data available on the Internet and perform detailed analysis of information in a single environment to make better informed decisions about the location of assets, customers, constituents, and competition.

For example, retailers can view customer profiles in a selected location for improved trade area analysis. Service companies can receive real-time traffic conditions from Internet traffic feeds to help service personnel more efficiently reach customers.

Closer to home, MapInfo's solutions are very well received in the telco/MSO space. Using the company's tools, service providers can design networks using GIS (geographic information system) and can take advantage of various information, such as rate and boundary data or franchise perimeters, to show carriers exactly where they can and should be building out. Wireless carriers can map exactly where their best coverage exists and deploy new equipment secure in the knowledge that it's going where it's most needed.

And carriers, always adept at collecting, analyzing, and acting upon consumer demographic data, can now utilize this mapped data as they look to build out triple play services, for example. Where are the customers who are most likely to spend money on this service? Where can I build out new infrastructure? Where *should* I build?

Another vendor making location-based services news is Telenity. And, some of the elements of the solution they've deployed with India's state-owned telecom behemoth BSNL are eerily reminiscent of the applications we were talking about back in the late 90's. Of course there's some new stuff too.

Telenity's Canvas LES, (Location Enabling Server) and an initial suite of 14 location-based services will enable BSNL to offer personalized value-added location-based services to its mobile customers. This highly scalable location solution is expected to serve from approximately 14 million to 70 million subscribers, which would make it the world's largest deployment to date.

The solution is designed to enable subscribers can easily find, locate, or monitor phones and other assets based on their geographic position, points of interest, and securely fine-tune their privacy profile on-the-fly when they want it.

The following applications are among the first services:

• Real-Time Fleet and Asset Management — Enterprises can locate, monitor, and manage their mobile assets and employees using a Web browser.

• Friend Finder — Alerts subscribers when one of the contacts they designate on their buddy list is in close proximity to their location.

• Mobile Yellow Pages — Just like it sounds; allows subscribers to find the location of the closest service point of their interest.

• City Sightseeing — Subscribers can look up location information of a place of interest (restaurant, museum, theater, park, and so on).

Location-based services are once again on the rise. One simply has to look around to find a wealth of information on this resurgent segment of the industry. Or perhaps just start walking towards the nearest coffee house. Maybe the information will find you...


ARE TELECOM MEGA-MERGERS PUTTING THE SQUEEZE ON YOUR ABILITY TO PROVIDE FOR YOUR CUSTOMERS?
THERE IS AN ALTERNATIVE
The telecom landscape is changing dramatically. But is that good for you and your customers? They want IP networks that are faster, more efficient and more secure than ever before. But they also want them to work seamlessly with their legacy infrastructure. We can help. As a Partner in our Global Crossing Global Partner Program℠, you can market our Global Crossing Fast-Track Services™ portfolio of IP services under your name or co-brand with us. Either way, your customers get state-of-the-art, end-to-end IP services from the company that was the first to deploy MPLS. Learn more at www.globalcrossing.com
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QUOTE OF THE MONTH:

"IP telephony resellers are doing it. VoIP service providers are doing it. Enterprise IT organizations are beginning to do it. And, increasingly, IP PBX equipment manufacturers are requiring it. We are talking about performing a VoIP readiness assessment. However, in spite of its many benefits, the VoIP readiness assessment has yet to become standard practice for every VoIP project. Why not? It's a puzzling question that is difficult to answer."

— Dave Zwicker, page 94

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Top 10 Most Visited Channels on TMCnet.com

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10. Triple Play

WHAT'S ON TMCNET.COM RIGHT NOW

TMC's VoIP for SMB Community

The VoIP for SMB Community is the leading online resource for business-class broadband services that combine voice and data. This site is your primary destination for the most relevant news, education, and trends analysis. But that's not all. VoIP for SMB will help you navigate through the myriad of service providers and plans — Empowering you with the knowledge to make the right decision for your business. Come check it out at http://www.voipforsmb.tmcnet.com. Sponsored by Covad.

TMC's Hosted VoIP Channel

The Hosted IP Channel on TMCnet.com features the latest news and features in this growing space. To visit TMCnet.com's Hosted IP channel to learn more about this developing business area, go to: http://www.tmcnet.com/channels/hosted-ip/. Sponsored by Citel.

WHAT'S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With more than 16 million page views per month, translating into more than 1,000,000 visitors, TMCnet.com is where you need to be if you want to know what's happening in the world of VoIP.

Here's a list of several articles currently on our site.

USF Compliance Could Cause Major Headaches for VoIP Providers

Providers of VoIP services now have yet another regulatory concern as the Federal Communications Commission recently expanded the federal Universal Service Fund reporting and contribution obligations to cover "interconnected VoIP" providers.
http://www.tmcnet.com/329.1

Is Business Ready for Enterprise Communications Everywhere?

Getting enterprise voice services on a mobile phone is an important achievement toward the full world of enterprise communications everywhere. But, while enterprise adoption of IP telephony is gaining momentum, the industry has also made important strides towards a new kind of enterprise mobility.
http://www.tmcnet.com/330.1

Halftime Report: 802.16e/WiMAX Aided by 802.20 Penalty Box

The controversial suspension of the IEEE's 802.20 Working Group will bolster the momentum of the WiMAX camp, proving to be a "minor setback" for cellular giant Qualcomm.
http://www.tmcnet.com/331.1

Feds Working on New National Emergency Alert System

The U.S. Federal government is reportedly testing a new national alert system that could be used to send emergency messages to virtually any communications device — including your cell phone. The new all-digital system would replace the current one used to send emergency messages via radio and television which was developed during the Cold War.
http://www.tmcnet.com/332.1

Spectrum Sharing Test-Beds Proposals Could Lead to SDR, Cognitive Radios

Proposals outlining various trial test beds to explore spectrum sharing technologies submitted to the federal government could foster further development of software defined radios (SDRs) and cognitive radios, eventually leading to commercialization of ubiquitous radio access networks.
http://www.tmcnet.com/333.1

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By Rich Tehrani

Hitting the Road in the Name of IP Communications

I am a blogger. Alas, I have been blogging less often lately. There. It feels much better to get that off my chest. Sometimes it helps to get past something by confessing it. It feels even better to write it in a column that reaches hundreds of thousands of people.

I have a reason for my lack of blogging, and that is customer meetings and speeches. I have been traveling all over the country giving speeches and visiting customers, learning about what is happening in the world of IP communications. Generally, the more meetings I have, the more I have to write about. But, I may have exceeded the breaking point, as I have met so many companies that it has been difficult to write it all up. So, I decided to slow down for 24 hours and write up some of the things I have seen. Don't expect this column to have broad concepts about the industry, but instead lots of juicy tidbits about the most intriguing companies and people I have recently met.

Resellers

Last week, I keynoted at VoIP (define - news - alert) Sizzles in Miami. The event targets resellers, and at the show there was a great deal of optimism about the future of VoIP products and services. Resellers had a few questions and concerns and were trying to figure out the best products and services to sell. In the exhibit area there was great interest in the Allworx booth, among others. Word on the street is that Allworx is doing some very exciting things. The people at the company are more enthused than at any other time — I have known them for years.

I also moderated a panel at the conference, with Alan Percy of Audiocodes and Michael Baus of Linksys sitting in with me, discussing how resellers can make money in voice over IP. The audience led the session and they were pretty savvy, asking lots of great questions. One of the better ones was related to Skype and the fact that products in the future will likely have to incorporate the Skype protocol so as not to ignore the massive user base of Skype callers in the world. I made a statement that, in the next few years, I expect all IP communications vendors targeting the enterprise to have Skype support. Even after further reflection, I think this comment is right on.

In the next few years, I expect all IP communications vendors targeting the enterprise to have Skype support.

Another theme echoed by Alan and Michael was that, if you are a reseller, the biggest pitfall you are likely to experience is trying to sell IP communications products and services without adequate training. So, make sure you get trained before you sell something.

From there I went to the Synnex Corporation annual conference, which was held in South Carolina. I met with many more resellers that had some great questions. Most of the questions centered on how to pick a company whose products to resell and also how to sell solutions and not merely push boxes.

For my part, the presentations really focused on this theme. Box pushers will be squeezed out of the reseller business in the next few years so, if you aren't focusing on solutions, you might be history.

Also, there are some traditional interconnects out there. The PBX resellers who refuse to embrace VoIP. Your days are numbered if you don't evolve.

I really enjoyed meeting all these resellers and I was happy to see that most of the people at these conferences were readers of TMC publications and go to TMC events. It is always great to meet the TMC community in person.

Cisco

I also had a chance to spend a day with Cisco (quote - news - alert) in San Jose and I learned a great deal about the company's products and services. Cisco's cable offerings are doing well and they have actually built their own head end on campus, which was extremely impressive. We took a tour of their labs and got to see some of their testing procedures that ensure quality. We then got to see a real live home (of course, this was in the lab) laden with Cisco and Linksys products. This was the home of the future, if you will, with streaming audio over WiFi, HDTV, and more.

Cisco has always done a good job of branding and one of my favorite taglines from the company is "empowering the Internet generation." Their new tagline may be "the network is the platform for experience." I may be paraphrasing, but this is the general concept. The point is that podcasting, online dating, and other activities are increasingly network-based and, as Sun's Scott McNealy famously said, "The network is the computer." Really though, "the network is the experi-

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ence," or at least it provides the experience.

Another interesting tidbit from the meeting is that consumer Internet traffic has surpassed business Internet traffic for the first time. This is not good for service providers, as consumers pay much less for Internet connectivity. The point is that service providers had better start selling new services soon or they are in for some serious trouble.

One of the ways to ensure this revenue is generated is by allowing service providers to take advantage of Cisco's service exchange framework, or SEF, allowing providers to provide the identity of users and, subsequently, allowing for things like content filtering, IPTV, and newer concepts like a "turbo button," which would allow a customer to have a speed and QoS bump on their broadband connection for a few days. This could be useful for someone who plans on downloading movies on the weekend, for example.

Gizmo

I also had a chance to meet with Michael Robertson who founded MP3.com and who founded SipPhone/Gizmo Project. Gizmo Project is a SIP-based solution that is very similar to Skype (news - alert) and has been gaining lots of momentum lately. Oh, by the way — the Nokia 770 tablet uses the SipPhone software to provide VoIP service.

Robertson has also founded a company doing AJAX work and, as such, you can expect to see a Web-based version of Gizmo Project soon and the solution will be heavily integrated with AJAX. AJAX is a technology allowing a Web browser-based application to seem like it is software running locally. Outlook Web Access and Google Maps are examples of AJAX implementations. Expect VoIP to go the way of the Web browser and AJAX and lose that bulky software.

Switchvox

From there we go to another company that has a number of MP3.com alumni, called Switchvox. (news - alert) The company makes an easy to use interface for Asterisk phone systems. In my opinion, the biggest barrier to greater Asterisk adoption is the difficulty in setting it up. Switchvox helps solve this problem and its interface is really slick and will have more and more AJAX elements over time. I am told practically anyone will be able to manage the system once it is up and running.

Another interesting Switchvox feature is the integrated IVR that is programmable, meaning you could build an application allowing customers to call in to check on their account balance and then pay via credit card. The IVR also has Web integration, which allows customers to track packages via the phone as well.

Wireless

I recently met with a CopperCom executive with whom I had a long discussion about how rural LECs are in dire straits if they don't wake up and smell the wires — or lack thereof. You see, the next generation is going wireless and if you aren't figuring out how to keep your customers using your service, you are in real trouble. Attention all regional phone companies! Start selling WiFi networking installation and mainte-

Talking IMS with NMS Communications' Mike Katz

RT: Mike, what are your general views on IP Multimedia Subsystem (IMS)?

MK: Today, IMS is in the early adopter stage, however we are starting to see adoption plans from at least the Tier 1 carriers and service providers. Right now there are key deployment and business issues being decided by the market makers. One trap IMS followers do not want to fall into is the belief that IMS will solve everything. In fact, IMS raises the need to differentiate using new applications, while enabling subscribers who are using these applications to roam seamlessly to a competitor's network. The advent of IMS raises tough technical and business issues!

RT: What are some of the types of new IMS applications we can expect to see?

MK: Truly new applications will come from the combinational effect you get when reusing IMS technology elements from different core applications. For example, a video content store used by a SIP application server for a core offering like videomail could be using IMS, easily repurposed to support a video blog, video share application, or a "rich voice call," not necessarily created by the original vendor but more likely by a new third-party vendor using IMS's SCIM (service capability interaction manager) layer. This creates a new ecosystem that takes out the "vertically integrated" silos of the past and fundamentally changes the time-to-market for network-based applications.

RT: What are some of the future problems of the all-IP network in relation to the carriers, the customers, the solutions providers, and so on?

MK: The problems are two-fold: technical and business. The technical issues are mostly around clarity of IMS definition in the applications space and interoperability. The business issues in IMS surround applications, what it means to offer one, how the users perceive more value from it (being IMS or not) and how the operators "play nice" among themselves and decide how they will allow subscribers and their applications to roam freely between operators in an all-IMS universe.

RT: Please comment on the importance of interoperability to the all-IP network.

MK: If IMS simply creates bigger and newer "vertically integrated silos" then IMS will have failed, and failed badly. To win in the future telecom market means to make money while providing accessibility and choice to consumers and businesses. Hence interoperability of IMS implementations is a must. Carriers should be able to "mix and match" IMS components within layers for the best possible return on their investment. It's also true at the application layer as well, requiring a similar interoperability effort from the developer ecosystem. I'd define the unspoken interoperability need between legacy networks

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nance, anti-virus support, file transfer services, and backups. And that is just the starting point. If you aren't looking for new revenue today, you won't be around tomorrow.

Don't be afraid to be creative. Get into the photo archiving business. Get into IPTV and carry local stations that aren't accessible via cable. Broadcast high school football or archived high school lessons.

Coupled with the fact that these companies might soon lose their Universal Service Fund financial support, things are looking bleak for those providers that choose to stand still.

SyncVoice

I also met with a company called SyncVoice, (news - alert) which focuses on unified management of VoIP networks in IP communications and especially hybrid networks, where vulnerabilities lurk. The company allows an enterprise CIO to manage voice as if it is data and, moreover, to TiVo the network to play back any problem events if needed.

Unimax

I got a chance to go to Minnesota and spend time with my old friends at Unimax, (news - alert) who tell me that business is going very well, and they are in the great position of advising their customers on what VoIP systems to install. The company is well known for its collaboration tools as well as its software for making moves, adds, and changes easy to perform. Basically, if you are migrating to IP communications in your enterprise, you will benefit from talking to these guys before you migrate. They really have the experience to help you make migration much easier.

Commetrex

I also had a chance to stop in the Atlanta area and see Mike Coffee from Commetrex. (news - alert) His company has been instrumental in making enabling technologies for voice and fax. More recently, the company is playing in the IMS space and I expect to see a number of innovative products soon. What I like about Mike is that he studies markets thoroughly. I am not aware of that many CEOs who study a new market like Mike does. I am looking forward to getting him to write for our sister publication, *IMS Magazine* soon.

If you read my columns or if you've heard me speak, then you are aware of how I harp about voice communities. Well, one of the best community building sites I have seen is radiohandi.com, which was launched by telecom veteran Brian McConnell. You can SMS a message to a group, leave a voicemail that is converted to e-mail and sent, and basically interact with a group in real time via a variety of communications modes. This is the ultimate product for a Soccer coach, for example. There are myriad business applications here as well.

Miles to go Before I Sleep...

Ironically, as I dropped this article onto the desk of my editor, I was headed out the door for another meeting! This industry is enjoying some tremendous success right now, and the level of innovation and creative energy coursing through the community certainly makes this an interesting trip. Speaking of trips, my team is waiting for me in the car. You

and IMS and between operators' IMS implementations as a major pothole in the road to IMS success.

RT: What is the role of NMS Communications (news - alert) in providing interoperability based on their communication platforms?

MK: IMS comprises 1,500 documents covering 60 network elements across architectural layers. NMS desires to create simplicity and value for its customers. NMS and its partners will provide IMS solutions in service offerings, service components and through its partners' service test equipment that enable the best of breed and mix and match IMS solutions to exist. This is exemplified by NMS's pre IMS solutions for rich voice calling (using GSMA standards) from our Mobile Applications group, our Vision server family (IMS-ready) and partners like Empirix building IMS interoperability test equipment with our Open Access family of products.

RT: What revenue generating IMS services we might see evolve over time?

MK: IMS services will initially evolve from simple "legacy replacement" voice applications to more robust and rich offers that include: video sharing, video blogging, mobile TV, interactive gaming, mixed mode messaging (IM to voice/video etc...), presence and location applications (such as playing coupon ads to your 3G mobile when your in range of a particular advertiser in a mall), streaming video service tutorials for fleet service personnel to enable faster, cheaper service repair strategies for commodity products (i.e., copier repairs, mobile Centrex for business, networked-based directory services)... The list goes on.

RT: Will there be enterprise applications or will IMS applications target mostly consumers?

MK: The mix of applications will start with consumer because the vast majority of carriers' subscribers are consumers. While much has been said about IMS consumer applications, historically consumers have had the least incremental cash to spend on "new applications." Operators will need to develop more of an enterprise focus for their new applications. Areas to examine include mobile Centrex, (call completion, bridging IP and TDM networks), networked-based directory services (bridging enterprise contact data into a network service) and fleet support services (streaming video tutorial content over IP or 3G-324M services to 3G phones). Consumer applications to watch are video blogging and video sharing.

RT: What do you think of the acquisitions being made in the DSP resource board space?

MK: Some companies come and go, but NMS has been a stalwart and will continue to be one in the DSP resource board market. Our position in this market, along with the hundreds of NMS application partners developed over the last 20+ years puts us in good stead as we move forward to address the challenges I mentioned earlier. It has been said that after a battle the dust will clear, and you need to look to who's left standing to determine the winner. IT

Mike Katz is director of video products at NMS.

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News Analysis By Robert Liu
TMCnet Wireless and Technology Columnist

Intel… What Were You Thinking?!?

At first glance, Intel's (quote - news - alert) decision to invest in Clearwire Corporation (news - alert) may baffle some observers. In an era characterized by the likes of Vonage (quote - news - alert) bleeding red ink, why is the world's largest chipmaker shelling out $600 million to an unprofitable service provider that only serves approximately 100,000 subscribers? What is Intel thinking?

Some analysts believe Intel's decision is a smart move in more ways than one. The chip giant isn't just looking at Clearwire to boost the Worldwide Interoperability of Microwave Access (WiMAX) standard. If Clearwire's management team (led by the cellular pioneer that locked out Ma Bell of the cellular market in the 1980s, Craig O. McCaw,) can execute properly, Intel could even succeed in bringing industry heavyweight Sprint into the WiMAX fold.

"This is a very good move on Intel's part," said Phil Solis, senior analyst at ABI Research. "It's one more reason for Sprint to choose mobile WiMAX. This is something that could further push Sprint toward mobile WiMAX."

Sprint intends to make an announcement this summer about which wireless technology platform will be used for its next-generation network. According to Sprint spokesman John Polivka, the company has conducted a number of trials using a variety of modulation specifications, including two WiMAX alternatives. (*No decision has been made public by Sprint as of this publication date*.)

"If Intel is successful with this strategy, it not only captures a new market, but it undermines some of the competitors out there," explained Daniel Meron, analyst at RBC Capital Markets. "Incumbents themselves are reluctant to make a move because they don't have any pressure. Having Clearwire out there may have the same effect that Covad had in 1999."

In the five years that WiMAX (define - news - alert) has evolved into the 802.16-2004 (Fixed) and 802.16e-2005 (Mobile) specifications, carriers have shown some interest in incorporating the technology to deliver wireless broadband service over a wide area network (WAN) or metro area network (MAN). But that interest has been mired by the vast number of competitive options, which has clouded the technology roadmap.

Ironically, the amount that Intel paid to Clearwire is the same as the amount Qualcomm paid for Flarion Technologies in August 2005. At that time, Qualcomm and Flarion argued that their proprietary Orthogonal Frequency Division Multiplexing (OFDM) technology called Flash-OFDM was superior to WiMAX and lobbied to build support for the spec through the IEEE standardization process. But recently, the IEEE took the highly unusual step of suspending the activities of the 802.20 Working Group, pending an investigation into "irregularities" — a move that Flarion officials have characterized as disruptive and unfair. Flarion and Nextel had an extensive Flash-OFDM trial in North Carolina until June 2005.

In addition to Flash-OFDM, Sprint has been conducting ongoing trials with UMTS TD-CDMA supplied by IPWireless, which has also received $14 million in combined funding from Sprint and Nextel. IPWireless officials dispute the notion that Intel's Clearwire investment creates a partner for Sprint to embrace WiMAX.

"I think, looking behind it, we've seen some things that don't make sense," said Jon Hambidge, Vice President of Global Marketing at IPWireless. "The whole idea that this creates a roaming partner…is that really the case? It's pretty clear that this creates more of a competitor than a roaming partner."

IPWireless, in fact, had conducted a trial with Clearwire's predecessor company. Despite what is claimed in a Securities and Exchange Commission filing, Clearwire actually wasn't founded by McCaw in October 2003. As early as October 2002, Clearwire had an agree-

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ment with IPWireless to deploy its TD-CDMA standard, but that deal collapsed once McCaw's NextNet Wireless, a provider of wireless networking equipment, was merged into Clearwire.

"One frustration we've had as a company is we're showing real technology in real trials, not PowerPoints," Hambidge told *INTERNET TELEPHONY*.

But Clearwire wasn't fully WiMAX-compatible either. When the company first launched its wireless broadband service in August 2004, the 802.16e-2005 standard was still evolving. That meant the network and gear supplied by the NextNet subsidiary weren't fully compliant with today's Mobile WiMAX spec.

"The equipment is very similar to what Mobile WiMAX is, but it's not Mobile WiMAX," ABI's Solis said.

So, to ensure future compliance, Clearwire will now get all of its wireless broadband equipment from Motorola, a strong Intel ally that shrewdly acquired the very profitable NextNet subsidiary. Nearly three-quarters of Clearwire's 2005 revenue came from the production and sale of base stations and customer premises equipment (CPE), which netted a bottom line of $46.6 million.

That leaves Intel with a business that only accounts for a quarter of total revenue. Despite serving 4.8 million people in 27 markets covering over 200 municipalities in the U.S., as well as a few markets in Europe, Clearwire only has 88,000 subscribers in the U.S. and 11,500 more in Europe. Along the way, it racked up nearly $175 million in net losses since its inception and expects to realize significant net losses for the foreseeable future.

As a result of Intel's investment, Clearwire was spared the agony that engulfed Vonage during its initial public offering. The infusion allowed the company to shelve its IPO plans.

"Intel didn't want that IPO to go poorly. Intel has to make sure there's at least one 2.5 band WiMAX network in this country," Hambidge explained.

"Intel looks at it as a loss leader, but they are trying to change the make-up of the industry. Intel is creating a market because the market is not there. Intel is taking its own fate in its own hands," Meron added. IT

Robert Liu is Executive Editor at TMCnet and a regular contributor to Internet Telephony Magazine. Previously, he was Executive Editor at Jupitermedia and has also written for CNN, A&E, Dow Jones and Bloomberg.

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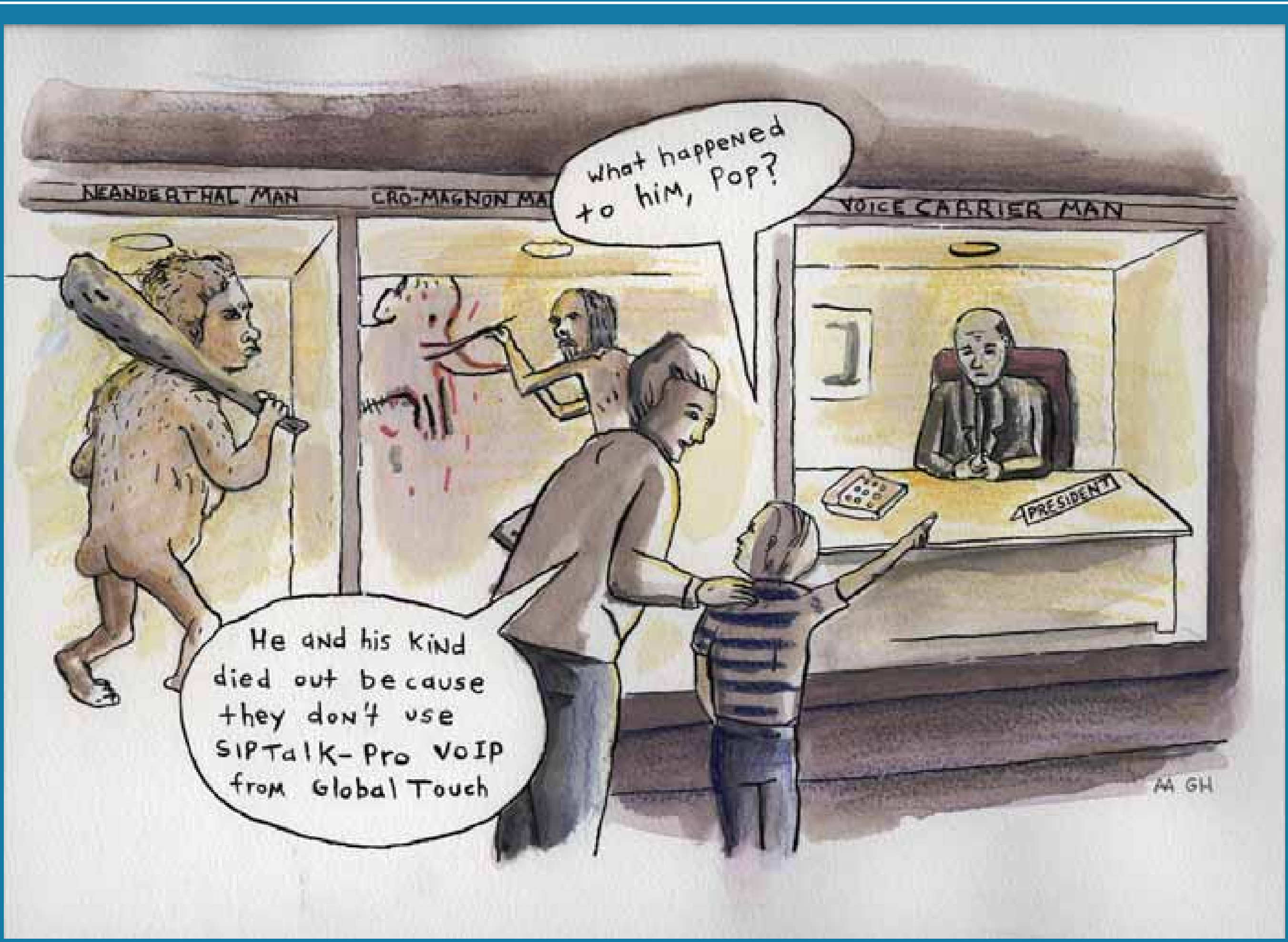

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Hosted VoIP Provider ZivVa Buys Vocalocity

By Erik Linask

ZivVa (news - alert) is a provider of hosted VoIP applications looking to simplify the creation and deployment of VoIP services and OEMs; Vocalocity (news - alert) provides OEM software for standards-based telephony solutions. Now, ZivVa has announced it has magnified the power of its product portfolio by acquiring Vocalocity.

The combined firms, operating under the name Vocalocity — because of the market leadership position of the Vocalocity brand — will provide enterprise customers and OEM partners two powerful VoIP platforms under a single brand: an on-demand service-based platform and an OEM-driven standards-based VoIP platform.

The combination is a natural mix; ZivVa's hosted VoIP applications for enterprises, such as its hosted PBX, complement Vocalocity's technology, which accelerates the migration to open telephony platforms by simplifying the development and integration of VoiceXML, CCXML, SIP and other evolving standards.

With the joint product set, the new company can serve both sides of the VoIP market. As a result of the acquisition, enterprise customers will be able to easily build automated voice response applications using standards-based creation and integration tools, such as the Vocalocity App Center, and run on-demand VoIP applications, like the office PBX, from a hosted services platform.

http://www.zivva.com
http://www.vocalocity.net

Foundry Networks Announces the Latest in Network Security

By Erik Linask

Foundry Networks, (news - alert) a performance and total solutions provider for end-to-end switching and routing, announced its latest, the new SecureIronLS secure LAN switches. The new switches feature embedded Layer 2 through 7 security for enterprise-wide protection against internal threats. They are meant as a security value-added alternative to traditional LAN switches in the distribution layer as an internal firewall; or they can be deployed at the network edge as a personal firewall with direct desktop and server connectivity.

The SecureIronLS secure LAN switches are designed for next generation enterprise networks, in which mobility, wireless connectivity, and convergence have depleted the effectiveness of traditional perimeter-only security solutions.

The SecureIronLS switches provide complete security throughout the life of network access by users, devices, and flows with authentication, access control, and intrusion/anomaly detection and prevention. Prior to granting network access to any user or device, the switches are set to first authenticate credentials, using secure Web transactions to authenticate the user against enterprise user databases.

Optional features include a hardware SSL acceleration module to scale the performance of secure Web authentication to a large number of users. This is critical for enterprise networks looking to deploy the latest in network security, especially since they have the largest numbers of mobile workers and a security breach could be devastating. In addition to user and device authentication, access is controlled through deep application inspection of user traffic against known threat profiles.

http://www.foundrynet.com

8x8 Awarded Patent for Hosted IP Telephony Technology

By Johanne Torres

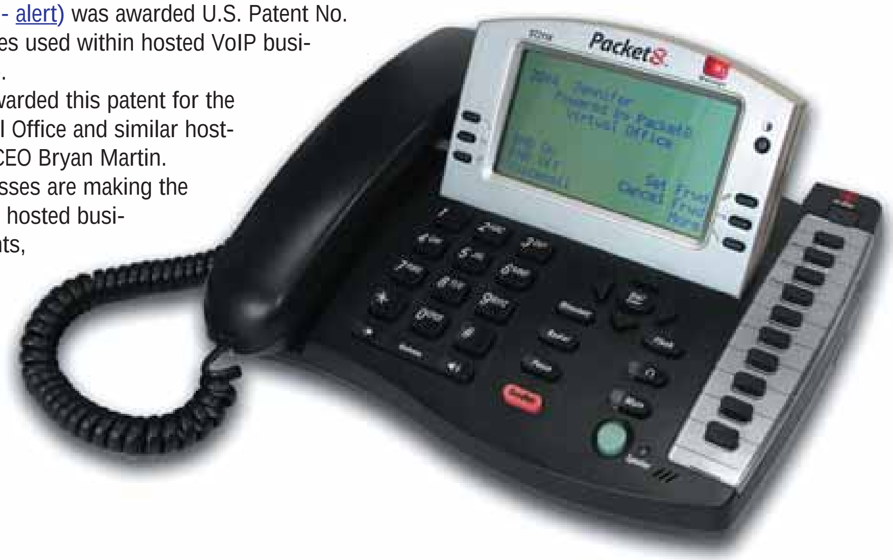
VoIP and video service provider 8x8 (news - alert) was awarded U.S. Patent No. 7,035,935 for IP telephony routing technologies used within hosted VoIP business services, including Packet8 Virtual Office.

"We are pleased and proud to have been awarded this patent for the technology incorporated in our Packet8 Virtual Office and similar hosted IP-PBX services," said 8x8 Chairman and CEO Bryan Martin.

Martin continued: "More and more businesses are making the switch from traditional PBX phone systems to hosted business VoIP services. Technological achievements, such as those in this invention, that provide scalable, cost-effective, user-friendly control over communications networks, are vital to the successful deployment of hosted VoIP business phone services."

8x8 was founded in 1987 and, since that time, has been awarded 61 United States patents for its voice and video communications technologies.

http://www.8x8.com



Polycom Supports VoIP Features on Microsoft's Platform

By Johanne Torres

Polycom (news - alert) announced its plans to develop and market business-class UC SIP voice end-points to include desktop devices that integrate the presence awareness, instant messaging (IM), and new telephony and VoIP capabilities of Microsoft's (quote - news - alert) new unified communications platform. The move will enable users to quickly find and communicate with customers, partners and co-workers within a more secure environment.

Polycom made compatible the Polycom VSX video systems, its SoundStation and SoundPoint IP phones, and the V2IU network address and firewall traversal system with the Microsoft Office Live Communications Server 2005 and Microsoft Office Communicator 2005. The move will allow users to have presence and buddy lists on the Polycom VSX video systems and the SoundStation and SoundPoint phones to simplify rich media calling.

"To support Microsoft's unified communication vision, we are developing integrated desktop devices that bring Polycom voice quality, ease of use, and performance to customers in Microsoft environments," said Sunil Bhalla, senior vice president and general manager of voice communications at Polycom. "Through our expanded integration with Microsoft, we are able to offer solutions that bring the advantage of real-time presence information, name-based dialing, and the simplicity of using directories and buddy lists to the forefront of the telephony experience."

http://www.polycom.com
http://www.microsoft.com

Inter-Tel Releases New Software Version

By Cindy Waxer

SMBs looking to increase the capacity of their IP-based communications system can now do so with just a software change, thanks to Inter-Tel. (news - alert) A provider of voice and data communications for businesses and an Intel Communications Alliance member, Inter-Tel has released its Inter-Tel 5000 Version 2.0 software. That's good news for today's SMBs, many of which lack the manpower and financial resources to accommodate sweeping IP-based communications system upheavals.

"The flexibility provided by this latest software is yet another reason IP-based communications systems are quickly becoming the standard for small and medium-size businesses," said Craig W. Rauchle, president and chief operating officer for Inter-Tel. "Version 2.0 was designed to give Inter-Tel 5000 customers a way to boost the capacity of their existing system and expand their communications capabilities as their business grows."

Designed to provide sophisticated management of system resources, Inter-Tel 5000 Version 2.0 software expands the capacity of the Inter-Tel 5000 family of Network Communications Solutions. Depending on configuration, v.2.0 can expand the existing capacity of the CS-5200 from a limit of 25 IP endpoints to support as many as 75 IP endpoints, and increases the flexibility of the CS-5400 to grow from a previous maximum of 110 IP endpoints up to as many as 175 IP endpoints.

"The less efficient days of hardware-based changes and dedicated resources are quickly becoming a thing of the past, thanks to solutions such as the Inter-Tel 5000," noted Jeff Ford, Inter-Tel's chief technology officer.

http://www.inter-tel.com

Convergys Intros Convergys Testing Solutions

By Stefania Viscusi

In a bid to help businesses further enhance the quality of their products and strengthen their software testing capabilities, Convergys Corporation (news - alert) announced today a new suite of testing solutions, Convergys Testing Solutions.

Convergys Testing Solutions combine professional consulting services with an automated software application testing tool, to offer businesses the ability to test and verify all changes made to their mission-critical systems, like billing and customer care.

"Convergys developed the Easy Test software to address our unmet needs from existing testing products on the marketplace. After client requests to make this solution commercially available, we created Convergys Testing Solutions, including the Convergys Easy Test application testing tool," said Andrea Ayers, Convergys president of government and new markets.

Easy Test's natural language scripts make it easy to use and give time and focus back to the area of mission-critical application development. With the concurrent testing functionality of Convergys Easy Test, users can write test cases as soon as the requirements are defined, ultimately resulting in cost savings and quicker time to market.

http://www.convergys.com

INFONXX to Provide Enhanced 411 Service Through VPF

By Erik Linask

INFONXX (pronounced "INFO – N – X – X"), (news - alert) provider of directory assistance and enhanced information services to both enterprises and service providers, announced this week a partnership with Stealth Communications, owner and operator of the Voice Peering Fabric (VPF). The partnership will provide VPF members access to a new range of voice directory services.

Through the partnership, INFONXX becomes the preferred provider to the VPF of directory assistance and enhanced information services via its IP network, which currently handles more than a billion calls worldwide.

INFONXX can be very flexible with what services it offers to its users. Each customer can customize their own services, a definite benefit since enterprises tend to want different services than service providers, and wireline customers' requirements vary from those of wireless customers.

In addition to its standard directory assistance, which is the basis for its entire product line, INFONXX makes available 14 additional product sets to its users. These include, enhanced services (e.g., weather, traffic, movies, etc.), text direct, Spanish language services, concierge services, location-based services, personal address books, and much more.

INFONXX will enhance the VPF peer-to-peer connections for VoIP telephony traffic with premier directory services. These services will be delivered over private and direct connections that increase the services the VPF provides to its members.

In addition to its current service offerings, INFONXX is currently exploring several other new services and technologies, including the ability to more efficiently implement voice recognition software to enable more voice-enabled services, especially for wireless customers.

http://www.infonxx.com

JDSU Intros NetComplete VoIP Service for Cable Operators

By Johanne Torres

JDSU (quote - news - alert) introduced its VoIP NetComplete Service Assurance Solution, the company's new VoIP deployment monitoring application for cable operators. NetComplete allows cable operators to expand service penetration without having to scale operations through data collection, consolidation, and correlation.

"NetComplete addresses the challenges cable operators face when deploying VoIP," said Roger Lingle, vice president of marketing for JDSU's Service Assurance Solutions business unit in a statement. "NetComplete gives operators an invaluable advantage, accurate visibility into the end-user's VoIP experience and a clear understanding of where, when and why the voice signal clarity needs improvement. Its ability to immediately isolate VoIP problems results in increased quality of service (QoS), lowered operational costs, drastically reduced truck rolls and a decrease in mean-time-to-repair (MTTR)."

The test system measures hybrid fiber coaxial (HFC) characteristics, the cable modem termination system (CMTS) and IP network. These features provide visibility into problem areas that affect voice signal clarity.

"On an individual cable VoIP customer basis, it correlates the data and presents a complete analysis to a cable technician on one screen to quickly pinpoint the root cause of a service problem," noted the company's news release.

NetComplete can also perform real time analysis of mean opinion score (MOS) to measure voice quality, and of simultaneous customer calls and correlate this to the signal's radio frequency (RF) performance from a single, centralized location.

http://www.jdsu.com

Alcatel Provides Time Warner Cable with Microwave Solution

By Stefania Viscusi

In a bid to stay in the competition among cable operators, Time Warner Cable's Southwest Division has chosen Alcatel to help it deploy premium triple play services in Southwestern Texas.

Alcatel (news - alert) will provide Time Warner Cable's (news - alert) Southwest Division with their microwave radios, which, combined with TWC's current fiber deployments, will help the cable operator to further its reach with an alternate transport route.

As competition to reach more customers increases in the MSO market, Time Warner will be able to use Alcatel's wireless technology, turnkey approach to deployment, and microwave platform to provide more customers with the services they demand.

Alcatel's portfolio of point-to-point wireless transmission solutions includes low, medium, and high capacity microwave radios for voice, video, and data (IP, ATM, Fast Ethernet) communications. Supporting a full range of frequency bands and network configurations, Alcatel's wireless transmission systems are managed by a unified and fully integrated network management platform, as well as through the simplified network management protocol (SNMP).

Tom Eggemeier, Senior Vice President and General Manager of Alcatel's wireless transmission activities in North America commented, "This contract with Time Warner Cable is an example of Alcatel's ability and commitment to leverage our technology and thought leadership to address the specific needs of MSO customers."

http://www.alcatel.com
http://www.timewarnercable.com

Minacom Adds MTA Loopback VoIP Testing to DirectQuality R7 Platform

By Mae Kowalke

VoIP solutions provider Minacom (news - alert) announced the addition of a new feature to its DirectQuality R7 service level test automation platform. The platform now allows operators to perform direct VoIP service quality testing of subscriber-installed multimedia terminal adapters (MTAs). Minacom has a patent pending for this new functionality, which allows operators to use PowerProbes installed in hubs and/or the network operation center to perform both analog/audio and IP/RPT-based VoIP quality tests.

The company explained that IP/RTP loopback testing "measures VoIP media streaming performance and quality at the packet level" and that analog/audio loopback testing "provides true user-perceived speech quality analysis, including the impairments audio codecs introduce when performing digital/analog and analog/digital speech conversion and compression." The latter is compatible with DOCSIS 2.0 (and later)-enabled devices from Arris, Motorola, and other manufacturers.

The DirectQuality R7 platform now also offers operators the following features:

- Loopback testing from the PSTN-side of media gateways and softswitches
- Measuring of MOS, R-factor, packet losss, jitter, latency, echo, noise, call volume, DTMF, and common network times
- Validation of Internet/video and data transmission performance
- Measuring of dial-tone, delay, post-dial delay, billing-duration, call completion ratio, and other connectivity functions

Minacom's CEO Michel Nadeau commented that, "No other test technique can measure VoIP service quality directly to subscriber's homes without requiring field-based test equipment; loopback testing to MTAs can greatly reduce the need to dispatch technicians to assess and resolve customer-reported service issues."

http://www.minacom.com

Tektronix Monitors BellSouth's VoIP Service

By Johanne Torres

Communications monitor Tektronix (news - alert) has announced that BellSouth Corporation (news - alert) will deploy the company's GeoProbe VoIP monitoring system from the company's Unified Assurance suite throughout its IP network. GeoProbe will provide BellSouth with performance monitoring, which enables troubleshooting, as well as identification of service degradations and network problems that may impact VoIP service quality.

"Real-time, end-to-end monitoring of VoIP calls throughout our network is an essential element of BellSouth's VoIP service assurance strategy," said BellSouth's chief technology officer Bill Smith in a statement. "The Tektronix Unified Assurance suite will help us to deliver the quality and reliability that our customers have come to expect from BellSouth, leading to increased satisfaction and customer loyalty."

Tektronix' GeoProbe monitoring system will collect all signaling info related to VoIP traffic and will provide the ability to trace a call across multiple protocols and networks. The system proactively detects network problems and will enable BellSouth to quickly isolate and correct problems for greater levels of service quality.

"While VoIP offers a solid case for reducing service provider operating costs and increasing service revenues, these advantages could be offset by the higher degree of complexity in monitoring service quality if not managed properly," stated Doug Dickerson, vice president, Performance Monitoring at Tektronix.

http://www.tektronix.com
http://www.bellsouth.com

Convedia Enables Enhancement of West's VoIP Conferencing Service

By Erik Linask

Convedia Corporation (news - alert) announced that West Corporation, (quote - news - alert) a provider of outsourced communication solutions, will deploy Convedia's CMS-6000 and CMS-1000 Media Servers at InterCall, a West subsidiary. Leveraging Convedia's expertise in media processing infrastructure, InterCall will look to further enhance its VoIP audio conferencing and collaboration services, while providing a versatile, reusable platform for West Corporation's next-generation IP services.

InterCall's (news - alert) Senior Director of product strategy, product management, and product development, Herb Pyles, explained that InterCall is investing in a proprietary IP platform to deliver reservationless conferencing — which, he says, is really what has helped the conferencing industry take off — and a key component of that platform are the Convedia Media Servers that that deliver the conferencing applications.

The Convedia Media Servers will provide a scalable, IMS-compliant media processing infrastructure to enable reservationless conferencing and collaboration services developed usingSIP) and Media Server Markup Language (MSML) protocols to integrate and control the Convedia media servers.

"InterCall customers use conferencing solutions for numerous applications beyond everyday meetings, so it is critical that we deliver the best available technology," elaborated Craig Webster, senior vice president of systems development for West Interactive Corporation. "West and InterCall evaluated a number of IP media server vendors, but Convedia's reliability, track record, broad range of multi-service processing features and superior economics proved decisive in our organization moving forward with Convedia."

InterCall also favored Convedia because its media servers are IMS-compliant and, with the heavy industry focus on SIP standards and IMS readiness, InterCall, too, ensured it had a product capable of handling an IMS environment when it becomes necessary.

"Companies like West/InterCall are capping their investment in traditional conferencing bridge technology and are shifting the majority of their new capital purchases towards IMS-compliant technology and IP services," said Marc Beattie, Partner & CSP Practice Manager, Wainhouse Research.

http://www.convedia.com
http://www.intercall.com
http://www.west.com

Fusion Seeking Patent to protect its New Peer-to-Peer VoIP Service

By Patrick Barnard

Fusion Telecommunications (news - alert) announced it has applied for a patent for its worldwide Internet Area Code, eNumber, and an automated eNumber ownership verification process — components that make the proprietary service unique in the marketplace.

The company only recently announced the launch of the new service — sold under its efonica brand — which lets subscribers keep their existing phone numbers and call each other for free using the Internet Area Code and either a PC or other compatible device. Fusion has selected "10" as the Area Code, meaning all a user has to do is dial "10," plus the area code and phone number they wish to reach, in order to place a free call. Users can call each other for free anywhere in the world, regardless of their physical location.

"We believe that the introduction of the worldwide Internet Area Code marks a significant development in the VoIP industry," said Roger Karam, president of Fusion's VoIP division. "We've simplified the way subscribers call each other. Because efonica supports the current dialing habits of consumers worldwide, customers should adapt to efonica seamlessly."

"We believe our approach to delivering VoIP services is a revolutionary advance in the industry and deserves patent protection," Matthew Rosen, president and CEO of Fusion Rosen said. He added that the main selling point for the new service is that it "avoids many of the flaws found in competitors' offerings, where consumers have had to compromise on many of the capabilities that landline subscribers have taken for granted for years."

http://www.fusiontel.com

VoIP Inc. Selects Envox CT ADE

By Cindy Waxer

Fast on the heels of having released the latest version of its communications development platform, Envox 6.3, Envox Worldwide (news - alert) has announced that VoIP Inc. has selected its Envox CT ADE interactive voice response development tools package to power its VoiceOne Enhanced 911 Verification Service.

The VoiceOne Enhanced 911 service offered by VoIP Inc. is an FCC-compliant solution for VoIP-based 911 calls. The VoiceOne Enhanced 911 Verification Service is a self-service IVR solution that empowers subscribers to ensure information accuracy. By calling 555-9191, VoiceOne Enhanced 911 subscribers can hear the address information that is on file and then either confirm that it is accurate, update the information, or instruct the system that they will be traveling to a particular area during a specific period of time.

"VoIP Inc. (news - alert) has created an innovative service that fills a huge gap for many VoIP-based service providers," said Mark D. Flanagan, president and CEO of Envox Worldwide. "We are proud that Envox CT ADE enabled VoIP Inc.'s programmers to develop such a cutting edge service — one that will surely accelerate the adoption of VoIP-based services and increase subscriber confidence."

The VoiceOne Enhanced 911 service provides five redundant entry points for IP 911 calls to enter the VoiceOne Network through the Internet or private peering, which are strategically placed geographically to provide the shortest path for a phone call originating anywhere in the U.S. These first levels of redundancy, and the strategic geographic locations, provide the best practical implementation to achieving the best possible QoS connection to the VoiceOne Network from Internet voice subscribers.

http://www.envox.com
http://www.voipinc.com

ClearMesh: The SMB Fiber Alternative

By Erik Linask

Optical wireless mesh technology can offer the quality and the cost-effectiveness that both providers and SMBs seek. It is more resilient than WiFi or WiMAX technology, which were designed for reach and mobility rather than the massive requirements of a primary converged network. Mesh technology also has the feature of becoming stronger as it grows — the more nodes are added, the greater the network's capacity becomes.

To take advantage of this opportunity, ClearMesh Networks (news - alert) has announced the launch of its next generation wireless optical mesh solution, the ClearMesh Metro Grid. The solution enables service providers to roll out business class services at 5-100 Mbps to the SMB market while leapfrogging legacy copper and economically rolling out a fiber-grade wireless infrastructure without trenching new fiber.

ClearMesh Metro Grid technology extends the reach of carriers' metro Ethernet networks, where lateral expansion of fiber would require significant up-front investment and months for deployment. Metro Grid can also complement existing WiFi and WiMAX service deployments by infusing additional bandwidth in high density metro areas, expanding the customer base for wireless networks.

The ClearMesh 300 nodes can be mounted on telephone poles or rooftops provides the interconnectivity that creates the mesh network. This integrated networking platform combines high-capacity Ethernet switching and low-cost wireless optical transport with business-grade Ethernet service delivery interfaces.

Using secure optical transmission, each node can distribute up to 300 Mbps of wire-speed, ultra-low latency, and full-duplex service capacity via three optical transceivers that can connect to three other nodes or other optical devices. This allows service providers to deliver enough capacity to concurrently serve thousands of VoIP calls, video streams and Internet sessions.

Each CM300 node has a range of 800 feet, or about two city blocks, and is designed not to be affected by weather, as long as they are deployed within the suggested range. Because the optical signal is a line of sight technology, the solution has an inherent security feature in that it cannot be intercepted or disrupted by radio frequency devices. To that, ClearMesh has also added a scrambling feature that secures the transmission, even if it were captured by an intruder in the line of sight.

This technology also provides access to a community that ClearMesh feels has been traditionally underserved and cannot afford the high costs and long rollout periods associated with laying new fiber. In fact, ClearMesh estimates the cost of running six new fiber laterals at approximately $180,000. Those same six buildings can be connected for about $3,500 with the ClearMesh solution — and would take only two day to install.

http://www.clearmesh.com

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RUN WITH IT

"Emergent enabled our new company to hit the ground running with leading-edge software solutions at a tremendous value."

James Ballard Smith
Chief Operating Officer
InZon Corporation

InZon, a new global communications corporation utilizing VoIP technologies, needed proven, reliable software solutions that out-performed competitors in capability and value.

"As a growing company we required an adaptable, innovative and flexible solution. Emergent was the clear choice," says InZon COO James Ballard Smith.

Emergent's ENTICE solutions closely matched InZon's start-up demands. And Emergent's products are flexible and practical, able to keep pace with InZon as the company grows into new service areas.

"Emergent is seasoned, with people who have been developing leading-edge software for years. I work with people I know, who can solve our problems. Emergent offers common sense cost structures coupled with out-front software solutions.

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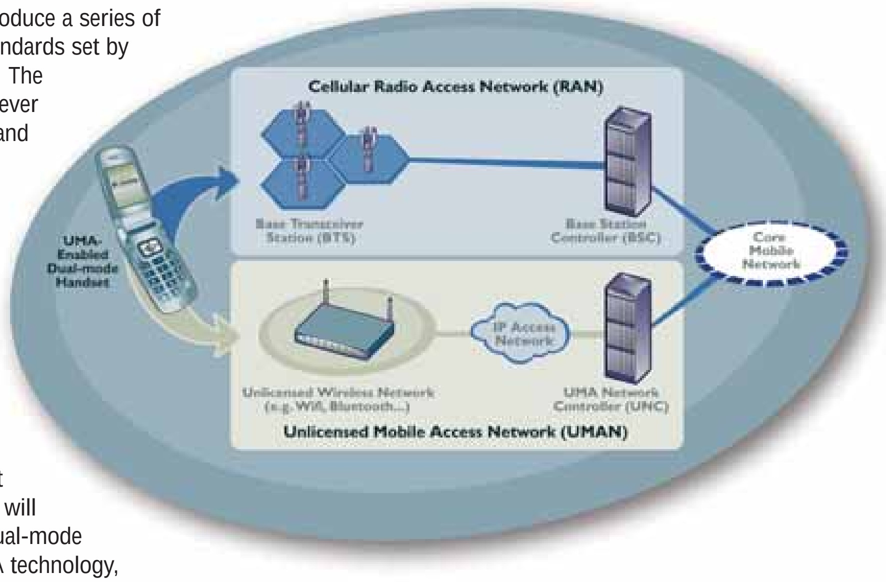
Kineto, Simcom to Build GSM/WiFi Dual Mode Handset Using UMA

By Mae Kowalke

Kineto Wireless (news - alert) and Shanghai Simcom, (news - alert) a subsidiary of SIM Technology Group, announced a collaboration to develop a dual mode (GSM/EDGE and WiFi) mobile handset using Unlicensed Mobile Access (UMA) technology.

Simcom, a China-based company that designs mobile handsets, will work with Kineto and its UMA client software to build the handsets. Kineto, an innovator in the UMA technology field, is providing Simcom with development, testing and integration tools as part of its Device Developer Program.



Working together, the two companies will produce a series of mobile phones that conform to global UMA standards set by the 3rd Generation Partnership Project (3GPP). The phones will allow users to access service wherever they are by seamlessly roaming between WiFi and cellular networks.

The two companies announced their goal is to make high-performance, low-cost UMA-enabled devices available to operators worldwide. Simcom's platform supports advanced technologies, such as EDGE, TDS-CDMA, and WCDMA. When combined with WiFi and UMA, these technologies "deliver a very attractive product portfolio for service providers globally," according to the firm.

Mark Powell, vice president of Kineto's client business unit, said that "working with Simcom will help meet the increasing market demand for dual-mode mobile/WiFi devices." He added: "Kineto's UMA technology, together with Simcom's excellent mobile handset design capabilities, will result in effective and desirable UMA products coming to market."

http://www.kineto.com
http://www.sim.com

BTC Using Nortel to Triple the Capacity of its Wireless Network, Deliver IMS

By Patrick Barnard

Are you planning a business trip or vacation to the Bahamas this fall? If so, you'll probably want to bring along your PDA or cell phone, because the wireless service there is likely to be greatly improved.

That's because Bahamas Telecommunications Company (BTC), (news - alert) the primary telecom operator in the Bahamas, is expanding its network using Nortel's (quote - news - alert) wireless technology. The GSM/GPRS network enhancements will triple BTC's capacity, thus facilitating delivery of next generation services including high-speed wireless broadband, m-commerce, multimedia messaging, and the ability to download large email attachments, videos, music and games. As a result of the improvements, the Bahamas will become an even more desirable destination for vacationers and business travelers. BTC also plans to roll out enhanced roaming capabilities in order to better accommodate international visitors.

"Our aggressive schedule to complete this expansion by the end of September is a direct response to the great investments being made by the Bahamian government in an effort to attract high-end hotels and resorts, as well as housing sub-divisions for affluent vacationers," said Leon Williams, acting CEO of BTC.

BTC is gearing up to offer other new services as well, including an advanced SIP-based prepaid voice service and a phone number service geared for expatriates, targeting the large community of Bahamians in the United States. BTC will be providing its GSM customers with SIP access, thus enabling it to target roaming wireless users or enterprises with WiFi infrastructure. BTC will also consolidate its prepaid services for wireless, wireline and VoIP.

http://www.btcbahamas.com
http://www.nortel.com

Mindspeed Announces Wireless Software Suite for Comcerto VoIP Processor Family

By Laura Stotler

Mindspeed Technologies (news - alert) has announced a wireless software suite for its Comcerto VoIP processor family. The new wireless solution with integrated Comcerto processors offers an optimized platform for implementing GSM/W-CDMA and CDMA media gateways with support for IMS and fixed/mobile convergence applications.

The Comcerto software suite also offers full support for VoIP over wireless access networks, high-speed wireless data access and multi-purpose 3G and WiFi/WiMAX services. The processor platform was designed to support wireless services and integrates packet and signal processing into a flexible architecture. It features advanced voice signal processing and high-quality echo cancellation and offers all necessary media stream processing and transport layer building blocks.

The Comcerto platform enables telecommunications equipment to seamlessly transmit secure, carrier-class-quality voice, video and data over wireless and wireline networks, which frees equipment manufacturers to focus on product differentiation at the application layer.

The new software suite builds on earlier offerings and the company's wireless codec portfolio, and also adds necessary protocol stacks and other packet-processing software. This ensures interworking with 3G, 2G+ and WiFi networks. The software also features voice quality improvements and offers support for ATM and IP networks while offering a roadmap for future wireless codecs.

Interworking with 3G networks is accomplished through the lu/NbUP protocol stacks and 2G+ inteworking is supported through TFO and circuit-switched data. WiFi interworking is supported through software for UMA, and Mindspeed's patented intelligent transcoding technology eliminates transmission delays for improved wireless voice quality.

http://www.mindspeed.com

Verizon Wireless Taps Lucent for VoIP Rollout

By Johanne Torres

Verizon Wireless (quote - news - alert) announced it has tapped Lucent Technologies (quote -news - alert) to deploy the company's CDMA2000 1xEV-DO Rev. A technology into its nationwide network. Lucent's Rev. A system, consisting of incremental hardware and software upgrades to Verizon Wireless' existing Lucent-supplied base stations, will enable Verizon Wireless to introduce a set of new services such as enhanced push-to-talk, messaging, and VoIP.

CDMA2000 1xEV-DO Revision A is an enhanced version of CDMA2000 1xEV-DO used to increase efficiency, data speeds and capacity of existing EV-DO networks. EV-DO Rev. A allows users to receive data (forward link) at speeds up to 3.1 Megabits per second (Mbps) and send data (reverse link) at speeds of up to 1.8 Mbps. These increased forward and reverse link data speeds reduce data latency and enable operators to deliver VoIP and other multimedia services.

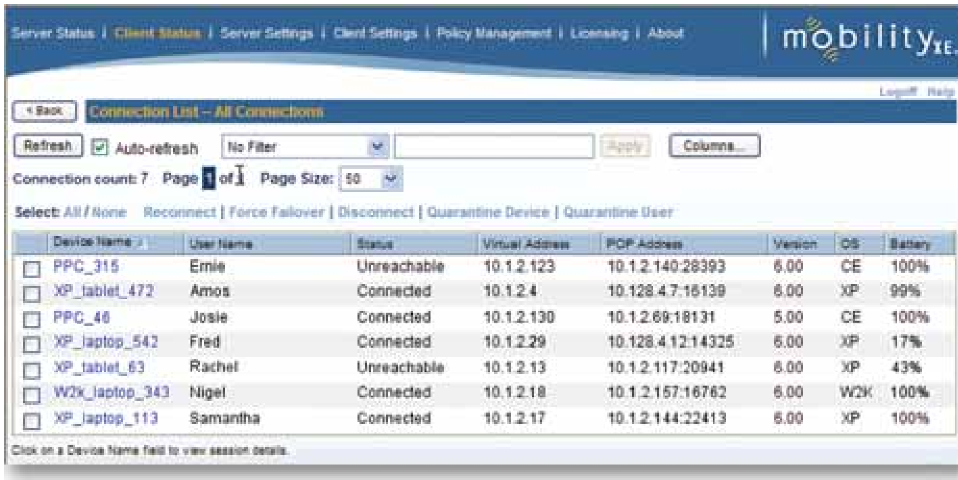
"Lucent has long been a key partner and has helped us provide our customers the most satisfying wireless experience possible. Going forward, we're going to count on Lucent as we roll out services, like Voice Over IP, that will help advance the way people communicate," said Ed Salas executive director of network strategy.

According to the companies, the first set of live, over-the-air calls using EV-DO Rev. A technology was completed in August, last year. The companies have, since then, conducted live VoIP and video telephony calls using the EV-DO Rev. A QoS feature.

http://www.verizonwireless.com
http://www.lucent.com

NetMotion Wireless Mobile VPN Now Microsoft Certified

By Erik Linask



NetMotion Wireless (news - alert) offers a solution to ensure mobile workers are able to stay connected to their businesses and colleagues while away from their desks. Importantly, its flagship mobile VPN solution, Mobility XE, has now achieved certification through Microsoft's Mobile2Market program.

Mobility XE — which enables hundreds of thousands of mobile workers in more than 900 organizations worldwide to get connected and stay connected to critical business applications over wireless networks — is the first VPN to be certified by Microsoft as a "Designed for Windows Mobile" solution for Windows Mobile 5.0 Smartphone.

The Mobility XE Mobile VPN keeps enterprise workers securely connected to critical data and applications, even as they roam in and out of coverage gaps or from network to network. Mobility XE turns a multi-network environment into a single unified network, maintaining application sessions so mobile workers can work without disruption.

To enable continuous connectivity, Mobility XE establishes a virtual IP address for each user, so that, even as clients move and are assigned new IP addresses by various networks, the enterprise application servers always see the same, unchanging virtual IP address. Mobility XE also incorporates security features to protect both the user and the network from intrusion.

"Thanks to advancements such as Windows Mobile 5.0, mobile computing is becoming a common part of mainstream enterprise IT practices," said Bob Hunsberger, president and CEO of NetMotion Wireless. "But true enterprise mobile computing is about more than e-mail and the Internet — it's about reliable access to enterprise applications, protected by rock-solid security and supported by centralized management that incorporates every device type an organization uses. That's our mission and it's what we provide through Mobility XE."

http://www.netmotionwireless.com

Carnival Cruise "Fun Ships" Offer Cellular Calling

By Stefania Viscusi

If you thought going on a cruise meant sailing out into the sea to get away and rid yourself of communications like cell phones, think again.

As the importance of being connected increases for business users and families, vacationers sailing on cruise ships are now offered the convenience of making calls from their personal cell phones while out in the middle of the ocean or at the port — without the need for additional dialing or software.

A provider of wireless maritime cellular services and a joint venture between Cingular Wireless (quote - news - alert) and Maritime Telecommunications Network, (news - alert) Wireless Maritime Services (WMS) (news - alert) is now bringing this convenience to travelers aboard all Carnival Cruise "Fun Ships" via an exclusive agreement between Carnival Cruise Lines and WMS.

For this offering, WMS, which provides roaming in GSM and CDMA to the cruise industry, will use their state-of-the-art high-bandwidth technology to provide all Carnival Cruise "Fun Ships" with a cellular service compatible with any cellular phone and enables guests sailing on Carnival to make and receive calls anywhere in the world from any cruising region.

"Whether making a voice call, responding to email, sending a picture message, or accessing the Internet to check on local happenings back home, what a great convenience for Carnival guests to enjoy the same features of their wireless service at sea as they would roaming on land," said Leighton Carroll, Chairman of the Board of Directors of Wireless Maritime Services.

http://www.cellularatsea.com, http://www.cingular.com, http://www.mtnsat.com

NETGEAR Provides High-Speed VoIP and Video Networking Across Home Powerlines

By Erik Linask

NETGEAR (news - alert) has announced the availability of its 200 Mbps Powerline HD Ethernet Adapter (HDX101) and Kit (HDXB101). The Powerline HD Ethernet Adapter is what enables users to seamlessly connect their Ethernet-ready devices to the home network. It distributes high-speed, high-performance — and, perhaps most importantly, affordable — broadband throughout the home, without any new wiring.

NETGEAR's Powerline HD Ethernet Adapters turn any AC electrical outlet into an HD-streaming, high-speed Ethernet broadband connection for real-time high-quality video, gaming, and VoIP. The adapter offers built-in Video Quality of Service (VQoS) for consistent performance with encrypted security, at unprecedented data rates, to seamlessly stream HD video throughout the entire home.

"We've seen great demand for this technology, as its simplicity is unbeatable," stated Kartik Gada, NETGEAR's product line manager for connectivity devices. "Consumers are requiring faster, more reliable, convenient, and uncomplicated broadband connections throughout their homes, in order to support a growing number of applications, like online gaming, audio distribution, and HD video streaming, all of which require copious bandwidth. By working with DS2, we've developed the ideal solution for successfully bridging the gap in areas of the home network that aren't conducive to wireless, without requiring a clutter of cords."

DS2 200 Mbps technology also supports advanced features, like QoS management, multicasting, and network isolation, making it the ideal application not just for in-home consumer electronics, but also metropolitan wide area networking.

http://www.netgear.com

Eurotech Brings Star Trek Technology to Earth

By Erik Linask

Eurotech (news - alert) has unveiled its space age Zypad wrist-worn computer. With an eye on revolutionizing the way millions of people work, the Zypad's high-tech design is the compilation of circuit miniaturization, a wide range of computer hardware functions, optimized power consumption, and ergonomic considerations. Indeed, weighing only about 10 ounces, the device could conceivably function as a common everyday tool, not unlike a mobile phone.

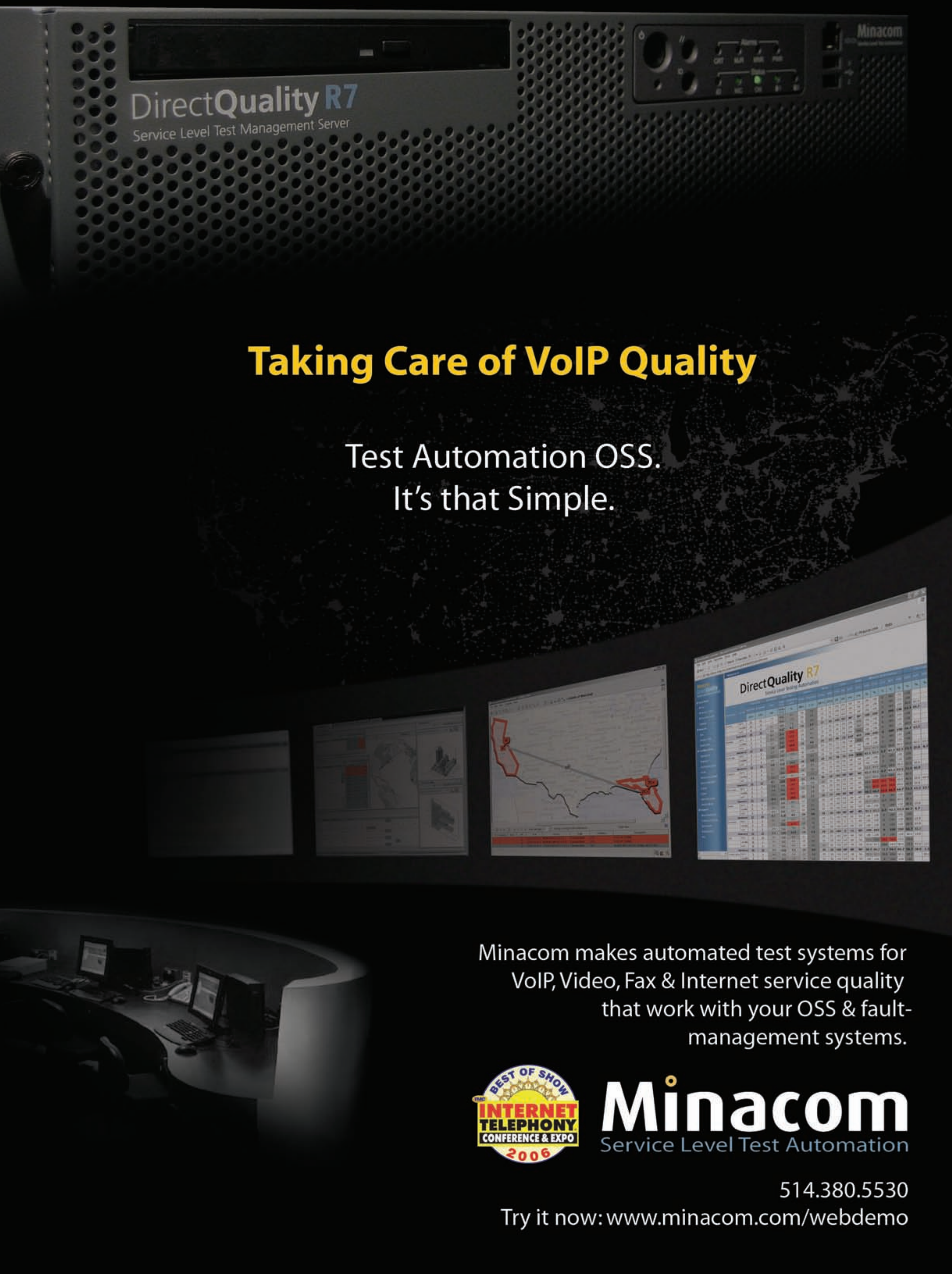
"The idea behind the wrist-worn computer project," explains Eurotech President Roberto Siagri, "is to leave the user's hands free. The Zypad enables users to work with a network connection without being tied to a fixed position. These wearable computers will give us a new freedom of movement, radically changing the way we work, and for the better."

While the primary target market for the wrist-PC will be demanding — perhaps hostile — environments where computers have become essential, yet the use of traditional PCs or other handheld devices is still impractical. In these instances, wearable computers will provide a means of access to integrated network resources without unnecessarily inconveniencing users.

The flexibility of the miniature device is noteworthy, as it adds to the range of possible uses. Largely, the controls are dependent on users' requirements and the intended application of the device. Both touch screens and navigation keys can be used, and voice recognition software is in the process of being developed for the Zypad.

Really, the uses are nearly limitless, once the functionality of a full-size PC is places in a small, wireless, wrist-sized device, in environments where hands-free and always-on connectivity are particularly important.

http://www.eurotech.com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BroadLight, AudioCodes and Legerity Enable VoIP Over PON

By Johanne Torres

BroadLight, (news - alert) a supplier of ITU Passive Optical Network (PON) semiconductor and software, announced it is partnering with VoIP technology companies AudioCodes (news - alert) and Legerity (news - alert) to deliver an integrated reference platform that will allow customers to add VoIP services on GPON.

The GPON VoIP reference platform is comprised of the BroadLight BL2000 Integrated BPON/GPON System-on-chip (SoC), AudioCodes' AC48x Voice over Packet Processor family and the Legerity's VE880 VoicePort Series telephone line interface systems.

The combined system has already been adopted by XAVi Technologies, a Taiwanese broadband products supplier.

JL Lin, chief technology officer of XAVi, said: "We chose to use the BroadLight and AudioCodes reference design because we needed to easily incorporate VoIP services in order to compete in major GPON contracts in review. Following our evaluation of the VoIP reference design, we believe it is a powerful and cost effective platform that will assist us in achieving design wins within the marketplace."

The joint reference design will offer PON OEMs and ODMs a hardware and software implementation of two or four VoIP channels, including full media processing (vocoders, echo cancellation, jitter buffer, etc.) and SIP signaling, supporting 3-way conferencing, call forward, call hold, call waiting and more.

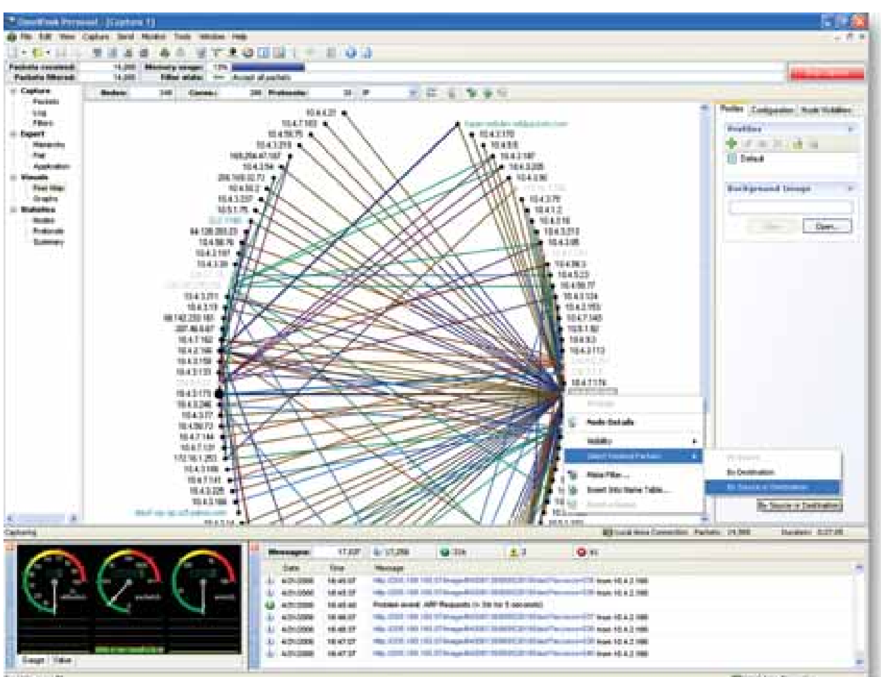
"We are pleased to empower BroadLight's end-to-end GPON solution with our leading voice over packet technology," stated AudioCodes' Shaul Weissman, vice president of VoIP processors. "GPON is one of the most promising next generation access technologies and as a pioneer in voice technology we recognize the importance of VoIP in this space."

http://www.broadlight.com
http://www.audiocodes.com
http://www.legerity.com

WildPackets Launches Free, Web-based Network Analyzer

By Patrick Barnard

WildPackets (news - alert) has launched a Web-based network analyzer, OmniPeek Personal, which is the first commercially developed expert network analyzer available free for personal use. Based on OmniPeek Analyzer, a component of WildPacket's OmniAnalysis Platform, OmniPeek Personal "gives individual users the chance to experience the power and extensibility of WildPackets' network analysis technology at no cost."



With OmniPeek, users can troubleshoot network performance and security problems; view "top talkers" on the network and drill down to see which nodes are communicating, which protocols and sub-protocols are being transmitted, and which traffic characteristics are affecting network performance; change filters on the fly without having to stop and restart packet captures; and view packet-stream based analytics by conversation pair, instantly locating network events, such as SQL queries and DoS attacks.

OmniPeek Personal, which runs on Windows, is available for download at the firm's Web site. Users also can extend the functionality of the analyzer by using a number of available plug-ins. OmniPeek Personal offers "best-in-class" wired and wireless analysis in one application, as well as an intuitive real-time graphics and display; Visual Expert and Packet Visualizer toolsets; and application analysis and Apdex scoring that reflects end user satisfaction with applications.

http://www.wildpackets.com

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Narad Unveils New Ethernet Switch Set Delivering up to 100 Mbps over Cable

By Patrick Barnard

With the likes of AT&T (quote - news - alert) and Verizon (quote - news - alert) aggressively rolling out new, high speed FTTH networks for the delivery of digital video, cable operators are now faced with the challenge of affordably upgrading their networks so that they can deliver next generation services with the same level of quality — or better.

Narad Networks (news - alert) announced it has developed a new modular switch product line capable of delivering 100 Mbps simultaneously to every home on a cable operator's network. Cable operators can use the new switch in a variety of network architectures serving both residential and business customers. For example, the switch can be deployed in a fiber to the curb architecture for high capacity residential broadband.

Narad says the new Ethernet switch set will allow cable operators to upgrade their networks to deliver the same speeds or better as a FTTH network, but with less expensive electronic components and far less fiber construction, thus significantly lowering the overall cost. In fact, the company claims operators can achieve network speeds that are faster than those delivered by Verizon's FiOS or AT&T's Lightspeed.

"In contested markets, cable operators now have a new HFC cable solution with superior performance and economics than their competitors," said Michael Collette, chief executive officer of Narad Networks. "With this new platform, cable operators can future-proof their network to support more advanced services by delivering higher capacity at lower costs and in less time than anything offered by Verizon or AT&T."

http://www.naradnetworks.com

Sunrise Telecom Introduces New Network Analyzer

By Patrick Barnard

Test solutions provider Sunrise Telecom (news - alert) introduced its new network analyzer, the AT1800RQS. This portable device — which was rated a near-perfect 4.5 in Broadband Gear Report's annual Diamond Technology Review — is said to deliver "the power and flexibility to perform all required RF and DOCSIS field testing." It is also reportedly one of the only RF testers to use a Windows CE operating system. This makes the AT1800RQS unique in its ability to run third party software applications with Windows or Web-based WFM/NMS systems.

The AT1800RQS includes a true spectrum analyzer function, a QAM analyzer (the integrated dual band 6-8MHz IF filters and Annex A, B, C compliance make it well suited for global applications), as well as a built-in filtering system and a non-interfering bidirectional sweep platform. Sunrise says it is the only true field spectrum/QAM/sweep analyzer that integrates a DOCSIS 2.0 modem w/BPI+ manufacturer digital certificates. In addition, the built-in upstream QAM 16/64 generator option, when used in conjunction with the AT2500RQ QAM receiver, characterizes the return band, by quickly finding existing RF linear and non-linear network impairments.

"MSO's finally get the most powerful and flexible test instrument they have been seeking," said Bernard Cadieux, vice president of marketing and business development for Sunrise Telecom's Broadband Products Group, in the press release. "Only the AT1800RQS can deliver the complete functionality required to correctly and quickly diagnose, locate and help resolve problems — a true spectrum analyzer, QAM analyzer, cable modem and sweep transmitter/receiver."

http://www.sunrisetelecom.com

Microsoft CRM For Palm Treo 700w Smartphone

By David Sims

X10DATA Corporation, (news - alert) developer of the x10DATA Mobile Platform, has announced that it has mobilized Microsoft's (quote - news - alert) Dynamics ERP and CRM applications on Palm's (quote - news - alert) Treo 700w smartphone.

When powered by the x10DATA Smart Client, Treo users can navigate financial, sales, service, and project management applications in real time over Verizon's EV-DO broadband network using 10 intuitive, single-handed commands.

"The Palm Treo powered by x10DATA enables sales managers, service reps, and CEOs to do what they do best — from servicing customers and taking orders to managing projects and cash flow," says Doug Migliori, President of x10DATA Corporation. "It means being 100 percent effective, 100 percent of the time, wherever you are."

After Palm introduced the Treo smartphone, it quickly became one of the most popular devices on the market, integrating a mobile phone with e-mail, organizer, Web access, camera, and more. Now, the Palm Treo 700w combines the functionality and ease of use Treo smartphones are known for with the power of Windows Mobile.

And with the x10DATA Smart Client installed, it's the perfect balance of ease and power, delivering everything needed to be fully productive while away from the office, with one-handed operation.

http://www.x10data.com
http://www.microsoft.com
http://www.palm.com

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www.spiritDSP.com

LatiNode Using SIP Peering Solutions from NexTone, Pactolus and Cantata

By Laura Stotler

LatiNode, (news - alert) a facilities-based VoIP service provider, is now using integrated solutions from Cantata Technology, NexTone Communications, and Pactolus Communications Software Corporation. The company is leveraging the solutions to achieve service improvements in speed, simplicity, security and economy through the use of SIP peering.

The solutions enable LatiNode to expand trans-nationally and attract new distribution and retail customers through voice-over-broadband offerings, prepaid calling card and conferencing services built on an IP core network. The SIP peering strategy is helping the provider extend international voice services access economically.

LatiNode is using the RapidFLEX Service Creation Environment and SIPware Services to launch new subscriber services. It is also using NexTone's IntelliConnect System to make intelligent IP and IMS network interconnects to the Cantata IMG 1010 integrated media and signaling gateway. This will enable LatiNode to expand services throughout the Americas and Europe.

The SIP peering scheme will enable LatiNode to implement IP customer and peer interconnects while also turning out new services and speeding up revenue recognition. The SIP Peering and SIP voice services architecture enabled by the new partners offers service differentiation and reduces carrier CAPEX and OPEX required to deliver voice services. The setup simplifies IP core network interfaces and the service architecture, reducing the costs of interconnecting with the PSTN for VOBB direct inward dial through LatiNode's CrossFone services.

"Cantata, Pactolus and NexTone form a uniquely comprehensive, simple and scalable service delivery solution," said Jorge Granados, CEO and founder of LatiNode. "They are strategic partners in driving our triple-digit millions-of-minutes growth, and they share our mission of enabling LatiNode to expand telecom accessibility, usability and affordability among Hispanic and other communities around the world."

http://www.latinode.com

Ubiquity Software Selected by Global Crossing as SIP Platform Preferred Supplier for Global IP Network Expansion

Ubiquity Software (news - alert) and Global Crossing (news - alert) announced the execution of a multi-year, multimillion dollar agreement whereby Ubiquity Software will become a preferred supplier for SIP Application Server technology in Global Crossing's global IP network.

Under the agreement, Global Crossing will use Ubiquity's SIP Application Server (SIP A/S) and Service Oriented Architecture (SOA) service creation environment to develop, deploy, and manage a new generation of IP-based services and to migrate existing legacy applications to Global Crossing's global IP platform. Global Crossing currently plans to deploy several applications including VoIP Interactive Voice Response and VoIP Network Transfer.

In connection with the agreement, INUK Networks, a partner of Ubiquity, entered into a separate agreement to purchase VoIP services from Global Crossing (UK) Telecommunications Limited (GCUK). The services will be part of a triple play offering to be launched by INUK Networks later this year, providing university students broadband Internet access, IPTV and VoIP.

As a worldwide provider of IP voice, video and data services to more than 600 cities in 60 countries and six continents, Global Crossing is responding aggressively to the demand for new and innovative IP-based applications and services. Global Crossing selected Ubiquity's SIP A/S for its rapid service-creation capabilities and advanced SOOF-based SOA, operational cost savings potential, and the ability to address new customers with new and innovative services that extend its market leadership as an IP carrier.

"The Ubiquity SIP Application Server platform strengthens our IP architecture and supports our strategy to rapidly offer new and innovative IP services to enterprise and carrier customers worldwide," said Dan Enright, EVP Operations, Global Crossing.

http://www.ubiquitysoftware.com
http://www.globalcrossing.com

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CounterPath Delivers BT's New VoIP Softphone

By Johanne Torres

VoIP and SIP phone provider CounterPath Solutions (news - alert) just inked a licensing agreement with BT Retail, (news - alert) a division of UK-based telecom provider BT, to deliver the company's softphone. BT Softphone is a tool that allows retailers to offer consumer broadband customers enhanced VoIP service, including more secure Internet voice calling, and video messaging.

"Our aim is to help consumers enjoy a wide range of multimedia information, services and applications, as we move towards a converged communications landscape," stated Gavin Patterson, group managing director, consumer, at BT Retail. "The BT Softphone allows consumers to experience all the benefits of VoIP, such as cost-effectiveness and increased flexibility, and CounterPath enabled us to provide an intuitive, user-friendly solution."

"As more consumers experience the benefits of VoIP and Instant Messaging, service providers are looking to maximize the value of IP-based communication for their customers," said Donovan Jones, president and COO at CounterPath. "BT is one of the world's leading telecommunications brands, and a leader in the UK broadband market, so we are excited to be working with the company as part of its commitment to increasing widespread VoIP adoption and to ongoing innovation in this space. The BT Softphone is the first phase of this approach and we look forward to working with BT in the future on the development of new multimedia applications and services."

http://www.counterpath.com
http://www.bt.com

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NICE Completes Acquisition of IEX Corporation

NICE Systems, (news - alert) a global provider of advanced solutions that enable organizations to extract Insight from Interactions announced the completion of its acquisition of IEX Corporation, a provider of workforce management, strategic planning, and performance management solutions for the contact center. The acquisition of IEX Corporation, originally announced on April 28th, 2006, was an all-cash transaction for $200 million.

The IEX and Performix acquisitions, the latter of which was completed on May 22nd, 2006, are part of NICE's strategic vision to provide a comprehensive range of solutions for all aspects of contact center business performance and analytics. The combination of solutions from NICE, IEX, and Performix enables contact centers to gain the first holistic view of contact center performance, addressing stakeholders at all levels, and providing enterprise users with critical insights on strategic business issues.

http://www.nice.com

Interactive Intelligence Releases Enhanced Multi-Site Call Routing Software

Interactive Intelligence, (news - alert) the global developer of business communications software, has made available a new version of its multi-site call routing software, Interaction Director, which adds generic object routing to enable distributed contact centers and enterprises to more effectively route and process virtually any type of work task — from customer service trouble tickets and loan applications to catalog orders and address updates.

"The latest version of Interaction Director marks a breakthrough by extending network-based, multi-site call routing to nearly any type of work request imaginable," said Ken Landoline, senior analyst for Yankee Group. "This unique generic object routing across sites is yet another step in the trend we're seeing toward the increasing interest in remote agent support, the blending of internal and external workforces, and overall multi-site performance optimization."

Interaction Director was first released in 1999 as a network-based pre- and post-call routing product designed to work with the company's contact center automation software, Customer Interaction Center(CIC). Interaction Director helps organizations improve operational efficiencies by more evenly distributing calls across sites based on real-time information, such as agent availability, skill-sets, and other pre-configured rules. This includes the ability to route faxes, e-mails, and other multimedia interactions from ACD queues across multiple sites and groups.

http://www.inin.com

ING Comercial America Uses Avaya Technology

When ING Mexico, (news - alert) a provider of insurance, pension benefits and financial services, needed a way to enhance customer service by connecting customers to the right agent and at the right time, the company selected an Intelligent Communications solution from Avaya. (quote - news - alert)

The company established a contact center in Mexico City with 305 employees and 229 agent positions that attend to an average of 220,000 calls each month. Car accident reports, health and life insurance inquiries, pension information, and tele-sales are just some of the services provided via the contact center, in addition to services for ING employees country-wide, such as IT and HR support.

To manage call routing, Avaya Customer Interaction Suite contact center software allows incoming calls to be automatically distributed based on preset parameters. As a result, callers are more quickly and efficiently connected to the agent who can best attend to their needs. For example, 80% of calls related to accidents are quickly identified and attended to in less than 10 seconds and, in the case of customer service, in less than 20 seconds.

The new system provides routing and reporting tools to help better balance calls among the staff, which, in turn, can be scheduled more efficiently to match higher-volume calling times. From the Avaya MultiVantage Communications Applications portfolio, Avaya Communication Manager IP telephony software serves as the core of the new communications network at each site. Avaya Call Management System software provides reports and management tools needed to monitor and analyze contact center performance, showing where improvements are needed and where to take fast effective action.

http://www.avaya.com
http://www.ing.com

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Cistera Extends VoIP App for Call Centers

By Johanne Torres

IP phone application provider Cistera Networks (news - alert) introduced two new products, the Cistera ConvergenceServer 7500 and Cistera CallCenterEnterprise v1.7.

The CisteraConvergenceServer (CCS) 7500 is designed for enterprise VoIP installations with large numbers of users and monitoring requirements. It can support up to 100,000 directory users or 600 simultaneous recorders. The CCS 7500 also supports four Intel Xeon Processors and up to 36GB of memory. The server runs the Cistera v1.6 platform and will support the CallCenterEnterprise v1.7 (CCEV1.7).

The Cistera CallCenterEnterprise v1.7 will offer Call Center admins and supervisors additional features, including remote monitoring, screen capture, integrated instant messenger (IM) client with better presence support, coaching, and desktop remote control.

CCEv1.7 also offers reporting, remote Web-based monitoring and QA sampling. The CCS integrates seamlessly with Cisco's IP Contact Center Express and Enterprise.

http://www.cistera.com

CosmoCom Completes Interoperability Testing with BroadSoft

Contact center on demand specialist CosmoCom (news - alert) announced it has successfully completed interoperability testing between its CosmoCall Universe all-IP contact center software platform and BroadSofts (news - alert) BroadWorks VoIP application server. The interoperable solution empowers service providers to offer a complete hosted communication service to their enterprise customers with a unified PBX and Contact Center.

CosmoCall Universe (CCU) enables service providers to offer a unified hosted contact center suite that includes ACD, IVR, IVVR, CTI, multimedia recording, and administrative tools, all within a single, high-capacity, high-availability, multi-tenant platform. BroadWorks enables service providers to offer hosted IP PBX services that have all the features of traditional PBX systems. Both services allow customers to benefit without incurring the capital investment and operational overhead of premise-based systems.

Service providers hosting the two platforms can now deliver all of the telecommunications needs of an organization as a unified whole. A single end user device, typically a SIP-enabled IP telephone, serves as the terminal for both PBX extensions and call center agents. Calls flow seamlessly between call center agents and other staff members. The systems share status and presence information, facilitating customer service when call center agents need to seek out subject matter experts in the larger organization. And CosmoComs advanced IVR capabilities are available to provide automated attendant services and other applications not related to the contact center.

http://www.cosmocom.com
http://www.broadsoft.com

Envox Accelerates Transition to Next-Gen IVR

Envox Worldwide (news - alert) announced that it has launched the Envox IVR Upgrade Program that will enable organizations to replace their legacy IVR systems with next-generation solutions. These new Envox-based VoIP and speech-enabled solutions can increase automation rates by 100% or more, reduce total cost of ownership by 50% or more and ensure outstanding customer interactions.

"The world is filled with dead-end IVR systems that are undermining customer satisfaction and retention objectives," said Mark D. Flanagan, president and CEO of Envox Worldwide. "These legacy IVR systems carry exorbitant maintenance fees and add unnecessary cost to an organization's overhead. It has become apparent that these legacy IVR systems, which are based on proprietary hardware, software and interfaces, have failed both the organizations and customers that they were designed to serve."

Envox created the IVR Upgrade Program to fill a significant void in the marketplace. Competitive disruptions in the market, as well as the ongoing evolution of technology, have left many enterprises and service providers with IVR systems that have become a "dead end" rather than a pathway to the future.

The Envox IVR Upgrade Program will enable any enterprise or service provider to upgrade its legacy IVR system to a next-generation voice solution customized to address their specific needs, regardless of the application, the industry, the network type, or the user interface.

http://www.envox.com

VoIP Peering: TransNexus and Emergent Partner

By Johanne Torres

TransNexus (news - alert) and Emergent Network Solutions (news - alert) certified the Entice Session Border Controller and NexSRS certificate authority and settlement server, which, the companies said, are now fully interoperable for secure VoIP peering.

The system combo uses Public Key Infrastructure (PKI) services for a more secure peer to peer access control and settlement accounting among interconnect peers. This security feature allows VoIP wholesalers, VoIP clearinghouses, IP transport providers, and network co-location providers to “increase customer loyalty and generate new revenues from peer to peer VoIP traffic.”

Secure VoIP peering for Entice now includes two new features used to reduce interconnect fraud risk and settlement disputes. “First, digitally signed tokens included with each call setup ensure secure peer to peer access control. Then, real time collection of encrypted call detail records from both source and destination peers eliminate interconnect settlement billing disputes,” explained the companies.

In order to enable secure VoIP peering, the Entice Session Border Controller communicates with the NexSRS settlement server using the OSP peering protocol, an operations and billing support (OSS/BSS) protocol standard. The OSP peering protocol is supported by VoIP platforms such as Cisco, Veraz, Asterisk, SIP Express Router and OpenSER.

http://www.transnexus.com
http://www.emergent-netsolutions.com

Vodavi and Speakeasy Join Forces

By Cindy Waxer

Vodavi, (news - alert) a provider of traditional and next-generation business telecommunications solutions and Speakeasy, (news - alert) a broadband provider, announced their new marketing partnership targeting the SMB space. As part of the agreement, Speakeasy will market Vodavi’s 6800 Series IP Terminals in conjunction with its flagship business VoIP solution aimed at small business customers seeking a single provider for hosted voice and data connectivity. Speakeasy business VoIP offers SMBs the benefits and functionality of a sophisticated PBX phone system without having to invest in complex and expensive on-site hardware.

The partnership gives Speakeasy’s direct sales and certified indirect solutions providers a hosted business VoIP solution that delivers Speakeasy’s broadband connectivity, local and long distance phone service, along with Vodavi’s 6812 and 6830 carrier-grade desktop terminals. The 6800 Series Hosted IP Terminals include a library of features and functionality business customers are accustomed to in a traditional desktop telephone. Unique features that make Vodavi’s IP Endpoints attractive for Hosted solutions include Enhanced Shared Line Appearance, Line Status Monitoring, and the first true hosted IP attendant console that can manage and monitor 24 line appearances with its unique single button direct station select (DSS) and busy lamp field (BLF). DSS and BLF are ideal for attendant console or secretarial call coverage positions that need extensive single button feature access.

http://www.vodavi.com
http://www.speakeasy.net

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VoIP Provider IPcelerate Chooses Nexus IS as First Platinum Partner

By Mae Kowalke

Dallas-based VoIP provider IPcelerate (news - alert) announced the launch of its new Platinum Partner program. The first company to earn that designation is Nexus IS, a growing provider of converged communications solutions.

IPcelerate explained that its Platinum Partner program is reserved for only the most valued and best performing companies. Members gain access to training, marketing services, and new sales channels.

Now that Nexus IS (news - alert) is one of IPcelerate's Platinum partners, the two companies are working together to provide customers with compelling and impactful VoIP solutions. Nexus IS' CEO Deron Pearson said that the Platinum designation is very beneficial to the company.

"We will now be able to separate ourselves further from the competition by developing custom solutions to solve our customers' problems," Pearson said in a statement.

http://www.ipcelerate.com
http://www.nexusis.com

Fujitsu and Airspan Team to Deliver Carrier Class WiMAX Solution

By Patrick Barnard

Is WiMAX (define - news - alert) the "best" solution for the last mile dilemma facing U.S. service providers seeking to deliver high quality VoIP and IPTV to their customers?

The answer to that question, increasingly, is "yes," but to say WiMAX is the only logical solution — for the now and in the future — would be stretching things a bit. It appears that widespread adoption of the WiMAX standard 802.16e is more than a year off, but in the meantime, U.S. service providers are gravitating to the current, 802.16-2004 standard in droves as they frantically upgrade their networks for the delivery of next generation video and voice services to their customers.

Adoption of the technology has reportedly been more pronounced in recent months – particularly among operators working in either dense urban or extremely rural regions (in an urban setting, a single WiMAX node can be used to deliver next generation services to dozens, if not hundreds of customers simultaneously – while in rural setting, it can be used to provide access to homes which are too costly to reach via a fixed line).

Networking solutions provider Fujitsu is playing a big role in driving WiMAX adoption rates. The company announced it is partnering with WiMAX equipment maker Airspan Networks to deliver a new, end-to-end, turnkey WiMAX solution for U.S. service providers. Fujitsu will become a reseller for the full line of Airpan's AS.MAX products, and will provide support for the new carrier-class solution through its Network Life Cycle Services.

Airspan's (news - alert) AS.MAX products support a range of end-user devices that integrate both WiMAX and WiFi technologies. Fujitsu (news - alert) plans to enrich these products with advanced network features that will make it possible for carriers to deliver quality VoIP services, including integrated SIP gateway messaging to enable call control and capacity reservation for voice traffic. With the combined solution, voice capacity is not reserved until requested, and can be dynamically adjusted while maintaining QoS, thus enabling the optimal use of radio link capacity while offering an ideal consumer experience.

http://www.airspan.com
http://www.fujitsu.com

DecisionOne Tapped as TalkSwitch's Certified PBX Installer

By Johanne Torres

Voice systems designer TalkSwitch (news - alert) announced it tapped DecisionOne, (news - alert) a technology support services firm, as a certified installer of its TalkSwitch line of IP/PSTN hybrid PBXs. The deal calls for DecisionOne to work with authorized TalkSwitch resellers to provide installation and technical assistance to TalkSwitch customers in the U.S. and Canada.

The partnership will provide TalkSwitch customers with on-site assistance for equipment installation and technical troubleshooting in cases where the TalkSwitch reseller is unable to do so. These features will be available through the company's Desktop to Dialtone service.

"DecisionOne's North American coverage is a benefit to our rapidly growing reseller channel," said TalkSwitch's president and CEO Jan Scheeren, in a statement. "Resellers looking to roll out a large-scale deployment, or arrange an installation in an area where they may not have adequate coverage, can call on DecisionOne to provide expert installation and troubleshooting service at the customer's location. The company shares our dedication to providing quality service to small businesses, and we look forward to a successful relationship with them."

DecisionOne's Desktop to Dialtone offering is a set of nationwide technical support services for VoIP original equipment manufacturers (OEMs), service providers, and value-added resellers that provides immediate national support services infrastructure for those companies looking to up or enhance their service capabilities. Desktop to Dialtone service spans the VoIP lifecycle to include network assessment, logistics, deployment, technical service desk and maintenance.

http://www.talkswitch.com
http://www.decisionone.com

Decisions, Decisions…

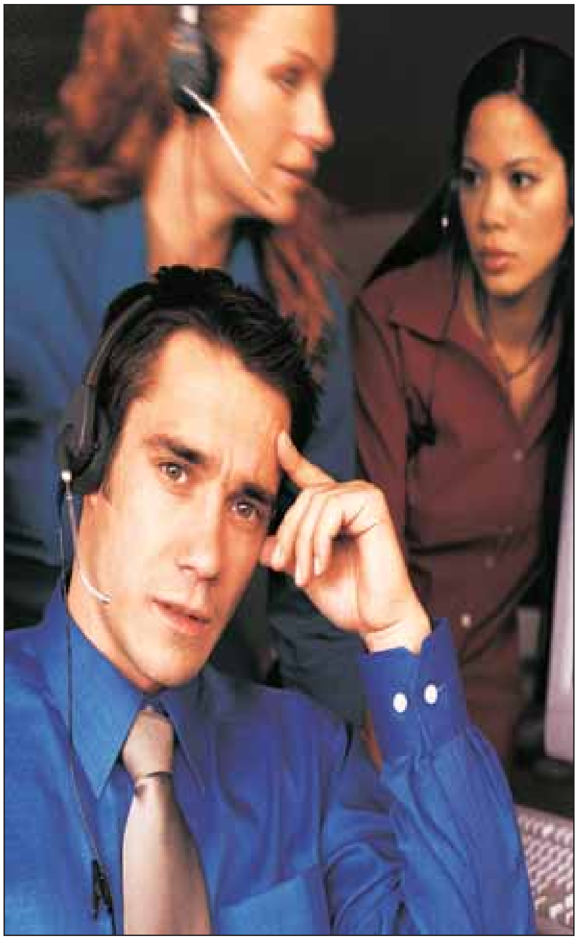
By Greg Galitzine
Internet Telephony

In our last installment, we discussed reports from the analyst community pointing to the growth of the hosted VoIP market, particularly in regard to the small to medium-sized business (SMB) segment.

This month we will take a look at a choice faced by SMBs as they consider the move away from legacy TDM-based infrastructure toward an IP-based communications solution. Essentially, the choice boils down to embracing a full-on hosted VoIP solution right from the beginning or migrating to IP in a more measured, piecemeal fashion by way of a VoIP trunking solution, which would allow an SMB to keep its existing equipment and protect its investment.

Hosted VoIP

Frost & Sullivan, a global growth consulting firm, believes that the hosted VoIP market is set to grow over the next four to five years at a compound annual growth rate (CAGR) of 84%. The company's research points at hosted VoIP services (in North America) growing from roughly 200,000 IP Centrex/hosted IP PBX lines to over four million lines in service by 2010.



The benefits of hosted VoIP are many.

To begin with, users pay on a periodic basis. There is no steep upfront cost; rather the cost is borne on an ongoing basis, per month/per station or per user. There is no need to pay a dedicated technical staff to manage the communications system as the solution is hosted off site by the service provider. The provider can enable the SMB to manage the regular moves, adds, and changes via a Web-based interface, which the company's administrative staff can easily manage. As the company grows, it's a simple matter to add more users to the system, each of whom can manage their own preferences and access the solution via a personal interface, often referred to as a "dashboard."

If a company has multiple locations, the benefits of a hosted VoIP solution become more evident. Since all the benefits of the solution are available wherever the network reaches, companies with disparate offices can all share the same features and functionality of the system, regardless of geography. This also enables companies to take advantage of things like extension dialing to colleagues, who may be half a world away. Other benefits include the ability to more easily transfer messages between users. A single receptionist can serve as the entry point for a company with distributed executives, with a full view into their availability to take a call.

Many hosted providers offer applications such as unified messaging, the ability to engage in ad hoc conferencing, and more. All this serves to make geographically dispersed employees feel as if they are more connected with their colleagues, which leads to better decision making.

Hosted VoIP solutions typically use a consultative installation process to help avoid installation problems too. The service provider usually sends an advance team to do a pre-deployment assessment of an SMB's office location(s). Based on that assessment, the proper steps are then taken to assure a smooth installation of whatever network gear is required as well as the phones. Beyond the installation, the service provider performs all the network monitoring and management remotely from their network operations center, providing a quality, hassle-free experience for the end customer.

Perhaps most importantly, hosted VoIP services are usually offered with 24 x 7 x 365 support and service level agreements. The service provider has a vested interested in not only keeping its customers up and running, but also in expanding and constantly upgrading its networks in order to remain on the cutting edge of new features and services.

For users concerned with disaster recovery and survivability, a hosted VoIP solution offers still more benefits. In the wake of numerous high-profile disasters, users are increasingly aware of the need to factor disaster recovery plans into their decision making process when it comes to purchasing a communications solution. Since hosted VoIP networks are based on IP and are generally built for redundancy, if a disaster, such as a flood, were to strike an enterprise location, users would be able to take their computers and their IP phones, head for higher ground, and plug in to the network wherever they could find broadband connectivity, and they would soon be up and running. If the users are not able to carry their phones with them, they can forward their phones calls from anywhere they have Internet access, through their "dashboard," to their home phone or a phone at a temporary workplace without their customers knowing the difference. Since there is no expensive on-premises PBX hardware and all voice mail is provided from the network, not only is important information preserved, but the loss related to equipment is limited. It would appear to the outside world as if business is operating as normal.

VoIP Trunking

VoIP (define - news - alert) trunking or VoIP access services are a good way for SMBs to add IP functionality without making the wholesale shift to hosted VoIP services. VoIP trunks enable a company to maintain its existing key system, PBX or IP PBX and — more importantly — allow a company to maintain the functionality of that on-premises system further into the service provider's network, as described below.

Todd Landry, Senior Vice President of IP PBX provider Sphere Communications, said in a TMCnet.com interview earlier this year that prior to such trunking solutions, using VoIP involved an expensive and complicated system of gateways to convert voice signal from digital to analog and back again — often involving several such "jumps" during the call's journey from sender to receiver.

"The jumps cost money, and they degrade the quality of the voice call," he noted.

VoIP trunking eliminates the need for additional investment in hardware by the customer, as this is usually bundled as part of the service by the carriers. The fact that the call can travel into the service provider's network as a digital signal allows the carrier to offer more features and functionality delivering the benefits of VoIP to the SMB.

According to the aforementioned Frost & Sullivan research, VoIP access and IP trunking services are on the rise as well. In fact, the number of deployed VoIP access/IP trunking lines is expected to grow from a scant 300,000 lines in 2005 to over five million lines in 2010, a CAGR of 70%. VoIP access lines are the perfect intermediate step for companies who believe in the future dominance of IP-based communications. It allows them to extend the useful life of their existing infrastructure or simply take a more conservative approach to embracing next generation technology.

Covad's Solutions

Covad (news - alert) provides its customers the best of both worlds by offering them a choice between hosted VoIP and a VoIP trunking solution. Covad's vPBX service is the company's fully managed hosted VoIP solution, which the company bills as being particularly suitable for sites with roughly 20 to 250 stations.

Covad also offers its PBXi Integrated Access Voice Service, which is a business-class VoIP service that allows SMBs to keep its existing on-premises phone equipment while typically providing lower operational costs. Once the customer's equipment reaches the end of its useful life and the investment has been paid down, Covad can easily migrate that customer to its hosted VoIP service, vPBX.

At the end of the day, it all boils down to customer preference. If an SMB is ready to embrace a fully hosted VoIP solution with all the benefits promised by such a system, then that's the right decision for it. If an SMB is not yet ready to take the plunge, for any number of reasons, then a slow and steady migration, by way of VoIP trunking, is probably the best way to go. IT

For More Information on Covad's solutions

Contact John Grady, Director of Product Management, Covad Communications:
jgrady@covad.com

Architecture Firm Speaks Up on VoIP

I recently had the chance to interview Hicham Benselloum, Technical Operations Manager at MKA Associates, Inc., an architecture design and planning firm based in Baltimore, Maryland. The company has been a Covad T1 customer for years and recently decided to sign up for Covad's Hosted IP Telephony solution. Here's what he had to say:

GG: Please describe your company.

HB: MKA Associates Inc. is an architecture design and planning firm with a long background in healthcare. MKA is composed of 14 professionals (five architects, two interior designers, two construction/project management professionals, four CADD operators, and an executive assistant). MKA has been providing planning and design services to the healthcare industry since 1993, and due to its success and new additions, MKA has been expanding its services to the corporate world as of 2005.

The president and founder, Mitchell Alguadich, is a registered architect and medical planner specializing in healthcare since 1983. As a medical designer and planning specialist, Mr. Alguadich started his career with Ellerbe Becket, moved to a national healthcare consulting firm, and then worked with Medifac Architects as a facilities planner and programmer. Prior to starting MKA, Mr. Alguadich was the vice president for planning with RTKL's Health Sciences Group in Baltimore.

GG: What challenges were you facing that made you decide it was time to upgrade or replace your existing phone system?

HB: Our move was the main trigger for our decision to replace our phone system. In addition, we were looking into our high monthly bills, and were faced with even bigger costs if we were to keep the same phone system. Furthermore, our system was just a very plain, featureless system and was not helping our productivity or satisfying our growing telecommunications needs.

GG: Why did you decide to go with a hosted IP telephony solution?

HB: The decision was not hard at all; not only was Covad's IP Telephony service very cost effective, it promised to provide us with "gazillions" of features and possibilities.

GG: Were there any specific benefits or features that you were hoping to take advantage of with the move to IP?

HB: The list of features is long: an auto attendant to serve as a backup whenever there is a need and to provide a professional image to our clients; call forwarding features at no extra cost; the possibility of monitoring the whole phone system and having access to almost all features from a very simple Mobile Smart Phone; access to voice mail from any Internet connection; adding/relocating extensions on our own... and lots more.

GG: Please describe the benefits you have experienced since switching to IP Telephony.

HB: The benefits are well worth the switch. When a new employee or intern joins the team, no problem! We can create an extension and assign a phone number in a matter of minutes. If someone is out, there are no worries about missing an important phone call — the forwarding feature works like a charm. Let's say you are on a worksite and need access to a phone/address book or to check on missed calls, or to review voice mails, or to stop or change the forwarding feature... well, it is all very easy and it can all be done from any PDA, smart phone, laptop — in fact, from any Internet connection.

GG: Please describe any negative experiences you have had since the switch.

HB: So far so good. Lots of people worry about losing connectivity and, therefore, phone service. We shared this concern as well. I am glad to report it hasn't happened yet. If it does, we have a couple of land lines and several mobile phones to transfer all the calls.

GG: Why did you choose Covad's solution?

HB: We have been a Covad T1 customer for years and have no complaints about the service, the knowledge, the responsiveness, or the friendliness of the staff. After we talked about the VoIP solution, checked the abundance of the features, and checked the prices in comparison with other providers, we determined Covad was, by far, the best suited to our needs, and we were right on the money. We couldn't have gotten a better package anywhere else.

GG: What advice do you have for other companies that are debating the switch to IP Telephony?

HB: You are probably afraid that you are going to be disappointed and lose all your business because your whole phone system is going to go "KABOOM." Well, that is not happening. Instead, here is what will happen when you switch to Covad:

- First, the quality of your communications is going to be better.
- Second, you are going to gain tons of features, and you can manipulate your system with a few mouse clicks.
- Third, you are going to gain a lot of efficiency. No one will lose a call — whether they are in the office, on a site, or working from home.
- Fourth, no need to call the phone company to add another line or do any maintenance for you.
- Fifth, you are going to save a load of money.
- Sixth, Covad is going to be happy and invest more on developing other features and be even more competitive.
- Seventh, you can schedule all your conference calls in advance via Outlook.
- Eighth…

The list goes on.

By Tony Rybczynski

Three Steps For Business Continuity if a Pandemic Hits

The avian flu virus has been in the news as it spreads from Asia to Europe and Africa. To date, it has resulted in the deaths of some 200 million birds and over 100 humans who have come in contact with infected birds. The concern is that the virus will spread globally and that genetic mutation will result in an ability to be transmitted efficiently among humans.

Timing is impossible to predict. However, governments around the world are taking the threat seriously.

The U.S. National Strategy for Pandemic Influenza Preparedness and Response, announced last November, is intended to limit the spread of a pandemic; mitigate disease, suffering, and death; and minimize impact to the economy and the functioning of society. The Implementation Plan of the Federal Government recommends that "government entities and the private sector plan with the assumption that up to 40 percent of their staff may be absent for periods of about two weeks at the height of a pandemic wave, with lower levels of staff absent for a few weeks on either side of the peak."

Business Continuity Planning

Pandemics can have different effects on an organization, this being highly dependent on industry. Certain demands may drop (e.g., in line with lower consumer spending), while others may increase (e.g., healthcare services). At the same time, supply chains and product distribution may also be impacted. These external factors may be difficult to control. On the other hand, enterprises should have a business continuity strategy that includes pandemic risks and balances business impact and cost. The first-order risks associated with a pandemic are primarily employee absenteeism due to illness, care-giving or fear of contamination. While employee education is outside of the scope of IT, an important focus for IT should be on developing an on-demand teleworking strategy for key employees.

This has three distinct components: remote user access, VPN connectivity, and IP Telephony and multimedia.

While everyone has a home phone (though they may not want to use it for work-related activities), not everyone has a PC. In addition, there is a proliferation of mobile devices, which can use WiFi or public broadband wireless technologies (e.g., EV-DO or EDGE). This creates new opportunities for employees needing to remain connected during emergencies. The enterprise or government agency should have a policy and strategy to leverage portable and mobile computing platforms in emergencies.

Many enterprises and government departments have IPSec-based Virtual Private Network (VPN) deployments for their existing teleworkers and mobile users, using broadband access, such as cable modems and DSL. Increasingly, SSL VPNs are also being used, a solution that avoids the need for management of VPN clients. Business continuity planning must ensure that adequate capacity is available to handle increased VPN loads during emergencies and opens the door for restricted SSL application access from home PCs.

The third component relates to using IP Telephony and multimedia soft clients across these VPNs, allowing anytime, anywhere access to resources. IP telephony can be leveraged to allow knowledge workers and contact center agents to be quickly relocated in case of disasters. These clients allow teleworkers to free up their home phones and, importantly, allow clients to act as a logical extension to business desktop phones. This can eliminate the need for call-out lists, since the network has the intelligence to find the employee wherever they are connected — a key value for emergency response teams. Multimedia clients add the important notion of presence, secure instant messaging capabilities, and video, which can be critically important during emergencies.

Never the Last Word

The avian flu virus is just one type of disaster that can hit an organization. Natural disasters, such as hurricanes, earthquakes, tsunamis, flooding, and, of course, terrorism, can likewise severely impact an enterprise's ability to run business as normal. During the SARS outbreak, Hong Kong authorities put severe control on commuting to work. After the Northeast blackout of 2003, the Ontario Government asked employers to close down their facilities while the nuclear power plants were brought back up on line. During this one-week period, I and about 5,000 of my co-workers (many of them software developers) were able to continue to operate on average at 85 percent of in-office productivity using our laptops configured with IPSec VPN, IP Telephony, and multimedia clients.

When disaster hits on a local or global basis, it is imperative to allow critical resources to continue to perform their duties, whether they are on-site or not. Portable and mobile devices and the proliferation of wired and wireless access technologies, IPSec and SSL VPN technologies, and IP telephony and multimedia are critical enablers for businesses and government departments to continue to operate under these conditions. Fortunately, these solutions are being widely deployed today as a means of lowering the cost of operation, enhancing productivity, and improving customer services. What is required is to establish a plan to rapidly expand the use of these technologies in emergency situations through in-house capacity planning or hosted services. IT

Tony Rybczynski is Director of Strategic Enterprise Technologies at Nortel. (quote - news - alert) He has over 20 years experience in the application of packet network technology. For more information, please visit http://www.nortel.com.

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By Hunter Newby

Global Crossing Appears to Peer

VoIP Peering press releases come and go, but big carriers are now moving into this space and some of the announcements have real tangible elements behind them. The growing trend for global carrier network assets is towards an on-net, zero-mile package where the users pay for access to all other customer endpoints on the network and then certain features above that. From a revenue standpoint voice is being treated like e-mail — not distance or size sensitive. The psychology of this topology can lead to very useful and far-reaching service offerings requiring only minor engineering changes to the carrier's network.

In a similar way to everything else in life, this is a case of mind over matter. In other words, it is like looking at the same box, but from another angle. The requirements for success are an open mind and the ability to execute on a vision of what the future should be. That usually takes someone to step out from the darkness to lead the way. With that, a new definition of VoIP (define - news - alert) peering has just been introduced and it combines a bit of the old and new, showing the way for just about any carrier to follow.

Global Crossing recently announced its VoIP Community Peering service. It is basically an "in-network" for enterprises consisting of all of the Global Crossing VoIP local service endpoints. Perhaps it can be described as an "in-extranet." For those readers that are not familiar with an "in-network," it is a rate plan model where the users can call any other user of the same service without incurring a metered, per minute charge for the call. This has been around in the mobile phone industry for a while now. It has also been around in the IXC long distance business too, but now it seems to have taken on the "peering" moniker.

An "in-network" doesn't require anything fancy with the internal routing, but benefits the customers with a lower, more predicable bill if both the originating and terminating points are on-net to the carrier. The model presumably has a self-propelled viral marketing element to it. Once the users realize that more numbers that are "in" equates to a lower phone bill they should go out and urge their friends, family, and business partners to join, so that they can all benefit from the lower costs.

There are many dimensions to this new Global Crossing service that warrant deeper understanding, not the least of which is the fact that Global Crossing is offering this "on-net" service with the use of existing architecture, needing no outside help to accomplish this. The network and its voice/VoIP users were "peered" by simply looking at the business of voice in a totally different way. No more billed minutes between users and any-to-any, on-net, worldwide is a bold move in my book.

Yes, it does create a VoIP island, but this is a start and a big step in the right direction. In time, Global Crossing will probably move to peer its island with other VoIP islands, just as e-mail services like Sprint Mail and MCI Mail began as islands and eventually peered. The same can be said about SMS traffic. It is a matter of evolution. Things can only happen and survive when they are supposed to happen and no time sooner.

Now is the time for the rise of the private Voice Internets. Peering those private voice nets will happen after each has had ample time to try and become the 800-lb. gorilla. Once they reach maturity they concede to interconnect because it makes sense to at that point.

Now is the time for the rise of the private Voice Internets.

Beyond the monolithic aspect of this new service are the details of how they do it. In different instances there will invariably be different technical permutations at the access layer as geography typically dictates the limitations of abundant fiber from multiple sources. To solve that, Global Crossing employs multiple access methods and interfaces with IP and legacy TDM PBXs to various points in its own network.

For enterprises with a TDM PBX, there are two methods. The first is a CPE solution using a Cisco router on the customer premises. From the router Global Crossing (GC) can accept a Layer 2 circuit, or a public IP VPN that connects in to the GC Acme Packet session border controller. From the SBC, GC connects back into its SONUS PSX core VoIP switch. The second method is a TDM loop out from the PBX to a GC SONUS GSX gateway. From the gateway, GC connects back in the PSX.

For enterprises with an IP PBX, the options include a Layer 2 Ethernet access circuit in from the customer to a GC Juniper router that then connects through the Acme SBC to the SONUS core. Also, there is the option of public IP into the SBC and then the core.

In both instances of TDM and IP PBXs, what is not obvious, but important to point out, is that the enterprise buyers have helped GC craft its offering and public Internet access back to the VoIP core was not a desired method. As Pat Reilly, Senior Manager of Voice Product Management, Global Crossing states, "By far, the most frequent request from the Enterprise customer is that they want VoIP bundled with private MPLS interconnections as the access method, not the public Internet. Security and QoS were the main concerns." GC will accept public IP connections if it is required, just to keep every possible option open. Usually, public IP transit is a necessary access method for remote locations that have little to no choice for cost-effective transport.

What all of this means is that GC has a service that can interface with old TDM and new IP switches over old TDM, or new Ethernet transport, or public IP transit networks covering all bases, methods, and potential obstacles. What it means for enterprises is that they need not have IP anything in their network and they can still use VoIP on the trunk side and take advantage of the favorable economics immediately. This gives them the luxury of time in their migration plans and a source of capital from their existing telecom spending budget to fund the migration.

There are few particulars about the service features in the higher layers that are worth mentioning. There is no ENUM database lookup functionality within the GC VoIP Community. If the call is bound for another GC local endpoint, GC logically keeps that on its VoIP network and it never touches the PSTN. Since there is no ENUM, the enterprise user does not have the ability to "query" the GC pool of numbers, but rather just chooses GC as the first route for all outbound. GC takes the call and routes it appropriately. If the number dialed is one of its own, GC routes it and the calling party does not get billed per minute for it. If the number is not in the pool, GC routes it to one of its many bi-lateral voice carrier partners at the best possible rate and quality.

What this means is that all of the users benefit from being able to send calls to all of the numbers GC possesses. The enterprise doesn't have to identify a list of customers, suppliers etc., that they wish to call and then become limited to that. So, if GC wins a major deal with a large organization that your business calls a lot and you are already a GC customer, they just did you a big favor. Now, you can call that company on-net and not receive a metered bill. As more customers sign up, the probability of an on-net call increases and the phone bill decreases. Once the VoIP Community evolves and matures, GC will eventually want to do two things: trunk SIP traffic to all of its outbound carrier partners for off-net termination and peer its endpoint number pool using ENUM with other carrier ENUM (define - news - alert) pools and or private ENUM registries. (That is, if they have not begun to do so already.)

Interestingly, in a seemingly IXC way, all of these components add up to VoIP Peering. The "call" itself is on-net as IP between the endpoints; it's not touching the PSTN and it is "free," or without an associated settlement charge. This still leaves open the debate about VoIP network peering verses VoIP call peering and whether or not the true sense of peering is strictly a Layer 2 event. In this case, it fits the definition and should act as a guide for others to do the same. IT

Hunter Newby is chief strategy officer for telx. (news - alert) For more information, please visit the company online at http://www.telx.com.

By Tom Schram

Integrated Network Threat Management

This article introduces the new product category, Integrated Threat Management. It is making life simpler for small and medium-sized businesses — and more lucrative for the VARs that serve them.

Before voice services became 'just' another data packet on the network, companies managed voice and data traffic separately. Enterprises had separate voice and data computing/networking worlds with staffs to keep both universes humming and only occasional territory battles.

Today, although much voice traffic has become IP-based, enterprises still value their voice services enough to put in place dedicated network connections that serve sensitive voice and video applications only. These voice/video-centric communications environments deal with 'quality' of services — such as having calls go through quickly and video so fast there is no jitter or static. Other traditional data traffic, such as that associated with e-mail, file transfers, computing, and such, is managed with different expectations and different tools. For instance, a short hiccup in a conversation becomes distracting if it happens very much, while most users tend to accept and expect regular interruptions to data transmissions.

Network applications for voice/video and for simpler non-voice data face similar problems, such as security, virus attacks, bandwidth and performance management, privacy, and so forth. However, because of the different expectations for the two types of applications, the solutions to such problems are usually very different in their execution. For either type problem, the enterprise typically has network protection infrastructure and staff to ensure the successful transmission of all traffic types and the budget to support these investments.

Until now, small and medium-sized businesses (SMBs) have been at a distinct disadvantage behind large enterprises in regard to network management. Their VoIP (define - news- alert) services are probably delivered by a specialist service provider who manages the voice network quality and security issues and simply delivers each call to its SMB destination over a dedicated high-speed data connection. But all other network issues, such as protection of data against viruses, violations to firewalls, management of spam, and the set-up and management of virtual private networks (VPNs), are the responsibility of the small-business owner.

SMBs have the same network security needs as Enterprises, but not the same budgets or in-house technical management resources. Now, reseller-friendly ITM packages protect SMB networks (and their voice, e-mail and other applications), without breaking the bank to do it.

Thanks to 'Integrated Threat Management' (ITM) offerings, management of network performance and use problems has gotten much easier. ITMs help small businesses protect all the data traffic on their network, just like the big guys do.

By definition, ITM packages combine two or more functions relating to monitoring, management, and correction of network security issues, such as viruses, privacy, performance, and even spam. But not all ITM solutions are the same.

Sometimes an ITM offer is created by adding on new capabilities to a product's primary function. While the base element may be best-of-breed, the 'added on' elements do not measure up in quality. For instance, an anti-virus 'add-on' to a firewall product may screen only 2,000+ threats, while, in reality, over 50,000 known virus types are potentially dangerous.

Another approach is to bundle several functions together, creating each element from scratch with a fresh look at today's business threats. This is the challenge my company has taken on. It can be great if each category element is equal in quality to the single-function competitors and 100 percent reliable. But many 'start-up' bundled products have compromised each individual function, resulting in an overall mediocre solution.

Microsoft suggests that ITM will be included in its next operating system, not a stand-alone product at all, but a service within the machine itself. But that is not here today.

Why should a VAR consider carrying ITM products?

• ITM gives VARs reasons to talk with the customer about multiple problems and long-term solutions, extending the VARs' value beyond a simple voice or data responsibility.

• ITM products that are truly channel-friendly do not require in-depth data network knowledge and can open up network solutions as a new business category.

• ITM packages are typically profitable, especially those with remote monitoring and management capabilities. The after-sale technical support issues are, in some cases, addressed within the product itself, reduc-

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ing the VARs' long-term support exposure.

What should an SMB consider when reviewing an ITM package?

• Match the solution power and scale to the needs of the company — either SMB or enterprise. Don't over-buy or under-buy, but make sure what you do purchase can be easily scaled up in the future.

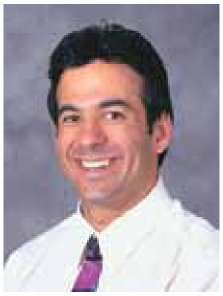
• Expect compatibility and flexibility. Don't throw out a good and working application you trust. Your ITM choice should supplement, not replace, your existing investments.

• E-mail is a critical consideration. If you use both POP3 and SMTP services, be sure your ITM product supports these equally well.

• Expect 100 percent accuracy and performance as promised. Look for certifications, lab test results, vendor-neutral recommendations. IT

ITMs help small businesses protect all the data traffic on their network, just like the big guys do.

Thomas Schram is the President and CEO of Wiresoft.Net, Inc., (news - alert) a member of the Enterprise Communications Association. For more information, please visit the company online at http://www.wiresoft.net.




By Rich Tehrani & Max Schroeder

Continuity Planning 101

Reality Strikes

The United Telecom Council (UTC) is a global trade association of companies that own, manage, or provide critical telecommunications systems to better support their core business. These companies include electric, gas, and water utilities; natural gas pipelines; critical infrastructure companies; and other industry stakeholders. Due to the nature of their environment during disaster situations, these utilities generally operate their own private communications systems, since past experience has proved that commercial telecommunications networks may have high rates of failure.

The storms of the 2005 hurricane season — specifically Katrina, Rita, and Wilma — severely impacted the entire telecommunications industry in Florida and the Gulf Coast. One aspect of the experience is that the radio, fiber, and microwave internal networks of the electric, gas, and water utilities functioned quite well throughout the storm and immediately thereafter. The commercial wireless, landline telephone and other telecommunications networks did not fare nearly as well. Even the UTC companies that experienced damage generally maintained communications via redundancy and most were fully back on line within 24–48 hours. Many of the commercial communications networks were still recovering months later. Perhaps the UTC can provide all of us with some guidance on how to better prepare for disaster and what better advisor than William R. Moroney, President and Chief Executive Officer of the UTC.

Rich Tehrani (RT): *I believe the UTC ran a formal survey to determine just how well the industry performed and to assess why its performance as compared to its commercial counterparts was so dramatically diverse.*

William R. Moroney (WRM): Yes, and the data provided some very interesting facts. For example, 86 percent of the responding companies reported that their communications networks not only survived, but continued to operate well throughout the restoration efforts. In particular, the crews from the many responding companies provided critical communications using land mobile radio (LMR) networks.

RT: *How is it that the UTC companies so clearly outperform the commercial operators?*

WRM: The quick is answer is we build them to survive disaster so that we can use them to restore critical services. However, in fairness to the commercial operators, the critical infrastructure industries (CII) have a much narrower focus. The redundancies and robustness built into the CII systems are limited in size and scope since they are designed and constructed to meet specialized needs. Similar construction might be cost-prohibitive for a commercial system. However, the performance of the CII networks proved the fact that communications systems can withstand the flooding and intense winds of a hurricane if they are built extremely well.

Max Schroeder (MS): *Even though the UTC networks performed very well, what limitations were observed and what steps can be taken to eliminate them in the future?*

WRM: Residents of the affected areas in 2005 were served by a wide variety of utilities, both large and small, that pool their resources in times of disaster through a network of mutual assistance contracts. The most reliable forms of communication during the crisis were the land mobile radio (LMR) systems, primarily because these systems are specifically built to weather such disasters. One downside is that the responding utilities operated on several different frequency bands, which, in many cases, forced host utility personnel carrying local radios to act as guides and communications links. Although necessary, it would be more efficient to have them working directly on the recovery effort. A second shortcoming is that, without a communal communications network, CII companies cannot effectively communicate with public safety or Federal responders. The solution to both of these limitations would be for the FCC to issue a small allocation of dedicated spectrum that can them be built to CII standards. This network could provide a reliable communications solution for both CII personnel and other respon-

The performance of the CII networks proved the fact that communications systems can withstand the flooding and intense winds of a hurricane if they are built extremely well.

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ders. The UTC has lobbied for this allocation on behalf of its members for a number of years. Certainly, the events of 2005 demonstrated the need for such a solution.

MS: *What recommendations would you suggest for enterprises to prepare for disaster?*

WRM: The "Top Three" recommendations are redundancy, redundancy, and redundancy. This can actually prove much easier for a typical enterprise than the UTC member companies. Although redundancy comes at a premium, it is much less expensive for the typical enterprise as compared to the cost for UTC member companies. Most enterprises have a major advantage in that they can flee a disaster. In contrast, our members are committed to getting to the disaster area as quickly as possible and restoring services. This means that enterprises can have failover locations in several geographic areas to minimize the cost of redundancy. With the exception of LMR systems, most utilities use the same telecommunications technology (albeit more private than commercial) as the average enterprise but their personnel do not have the option of leaving the affected area and connecting to a remote failover site.

Again, we see that planning for a disaster or business interruption can be achieved successfully if done properly. With today's IP solutions, even the cost is within the scope of most enterprises. The UTC (http://www.utc.org) is an excellent resource for any company seeking to understand the role of critical communications requirements in a disaster scenario. IT

If your company is interested in business continuity planning please visit http://www.tmcnet.com/channels/disaster%2Dpreparedness/default2.htm to view additional information provided by DPCF members, TMC, and the ECA.

Max Schroeder is a board member of the ECA, media relations committee chairman, and liaison to TMC. He is also the Sr. Vice President of FaxCore, Inc. (news - alert)

Rich Tehrani is the President and Group Editor-in-Chief at TMC and is Conference Chairman of Internet Telephony Conference & EXPO.

If your organization has an interest in participating in the TMC/ECA Disaster Preparedness Communications Forum, please contact maxschroeder@tmcnet.com or rtehrani@tmcnet.com.

By Kelly Anderson

Personalization and Privacy: Can the Two Coexist?

In the past month, the media, both trade and mainstream, have been covering data privacy. I have read viewpoints ranging from never using any customer data to the point of not even recording or tracking data to the idea that all data is the sole product of the provider of the service. I can't imagine what the layperson thinks the communications providers know about them. I suspect the opinion in this country is as polarized as the recent immigration discussion. On one side, we have someone like my father. I haven't dared to discuss this with him, for fear that we would argue over the fact that Ma Bell is selling his social security number to people overseas. The other side probably pays no attention, figuring that the government knows every time we check out a library book. In either case, we are left with the consumers not really being informed about what purpose service usage data actually serves. I think education consisting of industry collaboration might be in order.

Usage is recorded for various purposes. The first and, probably, most obvious reason is billing. Collecting revenue for services rendered has been the mainstay of back office systems, accounting systems, revenue assurance, and the like. One way to surely justify a million dollar system and an industry-wide standard is the collection of funds in the quickest, most efficient way. In addition, consumers understand that; they know that their bill each month correlates directly to what services and products they consume. Where it gets tricky is when the business model no longer requires transactional accountability. Companies that need to fund industry endeavors don't get it and, even further, consumers don't get it.

One item that seems to have appeared on the radar of the privacy news page is the host of new video and IPTV services that will offer personalized aspects of service. This may include offering free videos each month of a like category or having advertising pushed to your set that you will be most likely to buy. While a great business concept, this idea is a little scary to the average consumer. Imagine a message flashing on your screen informing you that you have two free movies available to you this month that are of a slightly personal nature. How would you feel? How can a valued service like this exist while not feeling like a stuffy telecom executive is sitting in your living room? As it is with most things, it would depend on the application. Being able to offer quality programming and content that is appreciated by the audience will take careful application and execution.

Personally, I appreciated the call I got a year ago to inform me of a $24.95 flat rate calling plan instead of a $200 usage bill. I dare say it changed my life in that I no longer watch the clock or give a second thought to picking up the phone. I appreciate that this company saved me $175 per month for the past year. That is the kind of personalization I like. The bottom line on personalization is that if a company can save time or money for their customers and let them feel like they are not being violated, then personalization and privacy can coexist as equals.

But, personalization requires information, which is where it can get scary. In most cases, this data is a normal part of functioning for a communications provider. Most of the data required for personalization is already required for other service delivery mechanisms like traffic analysis, fraud control, billing, capacity management, and customer service. I believe the average consumer would expect most providers know that service data is used to enhance services. This data is mostly looked at in the aggregate and is individualized only by account number or other internal identification criteria. I think most consumers feel their provider should know how and when they use the service for support and dispute resolution.

Information requirements for the personalization of services are becoming complex as more advanced services, such as multimedia and video, are rolled out. Time and duration stamping on the record is no longer applicable for many of the newer services. The industry has taken slow steps to get a standard that will satisfy the service and now is the time to increase that initiative. Many industry standard organizations have beefed up their agendas to include data transfer requirements, settlement standards, and usage requirements. The time could not be better. I encourage organizations that are offering services that require extensive content negotiation and delivery to consider joining one of these efforts. The time and resources it requires will pay off in an efficient and substantive standard that can ease implementation and allow partnerships to happen with minimal setup time. A cohesive approach to usage data will serve the industry and keep consumers informed. Do I believe that personalization and privacy can coexist? The answer is a resounding "Yes." With the right application of data that comes from the industry working together to create a unified approach, personalization will be a win for everyone. IT

Kelly Anderson is President and COO of IPDR.org. For more information, please visit the consortium online at http://www.ipdr.org.

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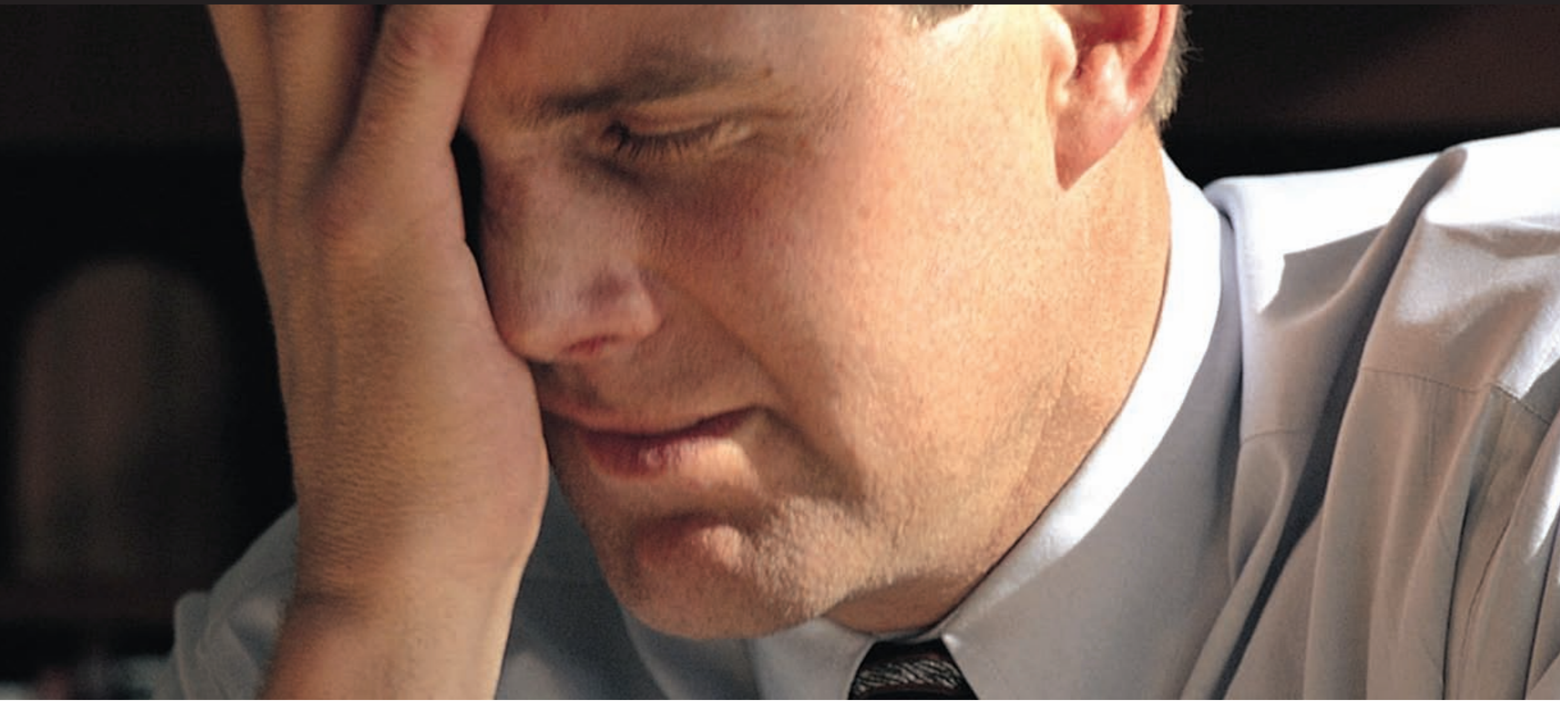
Don't Make a Million Dollar Voice Messaging Mistake



How much money can you expect to spend if you change to a new family of messaging systems? Our customers tell us that it's twice as expensive to purchase a new messaging system versus upgrading their current system!

Why?

- There is a huge time and expense commitment in training new users, re-recording greetings and auto attendants and rebuilding distribution lists.
- Users resist learning new user interfaces, and get little help from confused employees and overwhelmed help desks.
- Database conversions can be more complicated than anticipated.

At **CommuniTech Services**, we've specialized in providing and supporting voice messaging systems since 1983. We have the answers that solve your most critical concerns, including these:

Have you made sure that that the critical features you use now are available on your new system?
New "state-of-the-art" systems often lack the features you expect. Some don't even offer automated attendant.

Does your vendor know both the voice messaging applications and the capabilities of the new technology and can bridge the old with the new?
We know messaging intimately in both the TDM and IP worlds along with *Find Me, Follow Me, Unified Messaging, IVR,* and *Speech Recognition* and other productivity enhancers. Don't pay for your vendor's learning curve.

Does your vendor know how to proactively monitor and optimize systems that have heavy demand, high up-time requirements?
This is part of our standard support service. Don't wait for your system to malfunction...we'll make sure it never does.

If you're considering a messaging product from your PBX vendor, is their experience limited to just integrating their own messaging system to their own phone system? What if you need to integrate with other PBX's, e-mail or Presence servers, Speech Recognition, IVR or other Voice Messaging platforms?
As a messaging focused systems integrator, we can handle all of your future needs.

Does your system have the latest design considerations for up-time and high security requirements?
Our systems are built for maximum reliability and up-time. From the operating system to the CPU and power supplies, we are focused on insuring your up-time.

Can your vendor help devise marketing programs to help you get your user communities to use your services and to help you sell your services?
We'll help you sell your users on the services you provide

What other hidden costs of ownership are there?
Many vendors are inexperienced in sizing systems which leads to spending more than is necessary. We'll help you make the right decision the first time. Needs change – an expert in messaging can ensure that your messaging platform can adapt and thrive in an ever changing, complex environment.

CommuniTech Services can help you avoid potentially costly mistakes with your messaging systems needs. For more information, call us at 847.981.1200 extension 480.

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COMMUNITECH SERVICES, INC.

www.communitechservices.com

Three Ps to Finding the Best IP PBX for Your Enterprise…

…and to Steering Clear of IP PBX "Wannabes"

Industry trends can be presented to put most any new technology in a favorable light. But the movement to enterprise IP telephony and the Session Initiation Protocol (SIP) open communications standard is picking up steam for reasons that simply make good business sense.

On the SIP (define - news - alert) side, the protocol's interoperability enables networks, business systems, and communications devices to play nicely — seamlessly — and allows an enterprise to streamline voice and data and do things like ensure business continuity and support branch offices while protecting its technology investment over time.

Enterprises implementing IP telephony and migrating to voice over IP are realizing the efficiency and lowered operational costs of a single communications network. For business processes and customer service, they're also seeing that tools such as desktop softphones and IMS have made employees more productive, and that Find me/Follow me, presence management, and other "agility-oriented" capabilities have made workforce accessibility and collaboration more convenient.

In fact, in a recent North American survey on IP telephony by Frost & Sullivan Research Analyst Krithi Rao and Industry Manager of Enterprise Communication Applications Elka Popova (*IP Telephony: Premises-Based or Hosted... or P2P?*, June 22, 2006), decision makers in medium- and large-sized enterprises alike cited productivity/efficiency and convenience as primary drivers in planning a migration to VoIP. Yet, with 74% of the survey's medium-sized enterprises and 59% of large-sized ones saying they plan to implement VoIP by the year 2012 or sometime thereafter, the issue isn't when... but how.

Your First Decision

With IP telephony and SIP continuing to hit the mainstream, enterprises are looking at IP PBX systems with a more focused microscope to anchor their VoIP initiatives. Many, however, are also weighing hosted IP telephony options, especially when business process integration and investment protection down the road don't weigh as heavily as a limited budget and short-term survival.

Of course, business principals know what's best for their enterprise and its future, but there's still one very viable rule of thumb to deciding between a hosted solution and a premises-based IP PBX: Whereas VoIP can enhance the value proposition of hosted services with certain levels of flexibility, an IP PBX at your own site can increase communications capabilities, security and control — and offer a far higher return on your enterprise's technology investment over the long term.

The Three Ps Shopping Guide

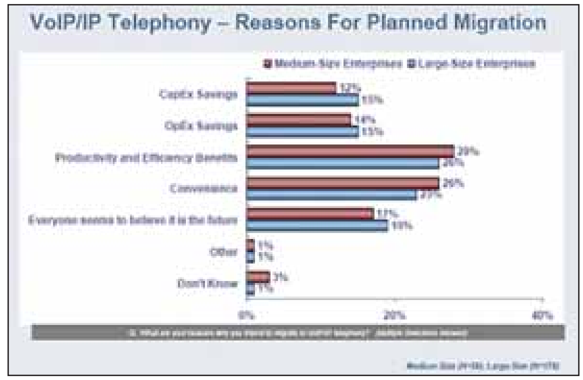
According to the results of Frost & Sullivan's survey, the enterprises they polled favored a premises-based IP telephony system over a hosted solution by a 2 to 1 margin. The reasons? Though hosted services rated higher in the flexibility category, survey respondents gave premises-based solutions a higher ranking in the categories for features and capabilities, control, and lower TCO. But beyond those factors should you choose to purchase an enterprise IP PBX, you should also consider the Three Ps — *Practical, Powerful, and Proven* — to find the best system.

Here's a look at practicality first, how the power of IP telephony plays a role in business communications today and what best defines a proven IP PBX system and the vendor who sells it.

Practical

"The modern IP PBX or VoIP phone system is not your mother's Centrex anymore." Credit Owen Linderholm of voip-news.com for that observation in his online article *Ten Features That Matter for Your PBX or Phone System–No Matter Your Business Size* (June 15, 2006). And he's right, especially in this age of information workers and the enterprise as an interaction center.

As business rules go these days, enterprises and their employees require a system like an IP PBX that can leverage open standards like SIP to integrate networked communications and business applications on the PC. After all, workers rely on computers and phone devices and an abundance of information every day to serve customers, process orders, track supply chains, collaborate, message, and



generally keep a business running. But, with one system for telephone calls — a Centrex, for example — a separate one for messaging, and others for data, it's nearly impossible to make business processes more efficient and put employees in control to effectively do their jobs.

This, too, with practicality in mind: An IP PBX should give departments and workgroups all the tools they'd expect from a phone system, yet should also enable an enterprise to utilize VoIP for broadband services and features, such as conferencing, and provide convenient, anywhere access to corporate voice and data.

Most of all, on the practical front, an IP PBX system should come complete, as in "all-in-one," including pre-integrated applications, IP phones and desktop softphones, and ACD and automated attendant for call routing — along with the flexibility to integrate third-party applications, such as CRM. Add everything up and it equates to usability, which is what Linderholm cited as the #1 "feature" on his Top 10 list. Besides, gains in efficiency and productivity have to start somewhere, and the most practical place to start is with a premises-based IP PBX phone and communications system.

Powerful

It used to be that more headcount manning the phones gave an enterprise all the power it needed to increase revenue. However, multimedia communications have now made adaptability the most powerful way to conduct business, and offering customers multi-channel avenues for the phone, e-mail, fax, Web services, and data access is crucial. Also, extend those touch points from the main office to every department, branch office, and mobile employee in your organization, and revenue streams multiply accordingly. The trouble is, market and customer requirements change constantly, and adapting to them — rapidly, before your competitors do — is nearly impossible with a bunch of non-integrated communications systems and technology silos.

Getting back to the new breed of all-in-one IP PBX systems, your enterprise should look for a system that provides a single platform and adaptable, pre-integrated multimedia applications to support changing business processes. Conversely, look out for "integrated" proprietary IP telephony systems that still come down to one piece of hardware after another, say for a PBX, ACD, automated attendant, chat server, and IVR system — with the promise of VoIP capability mixed in after adding another hardware box or two.

Beyond their higher potential for business continuity breakdowns, multi-box systems can actually rob power by putting barriers between multi-channel touchpoints and causing inconsistent service levels across media channels.

Proven

In our previous article for this series, we talked about an old rule of IP telephony and VoIP that says "IP-enabled" is close enough and that the New Rule is to buy an IP *system*, not "close enough." We also said to run away as fast as you can when a big-name proprietary vendor or some unheard-of new one says its PBX/VoIP phone system is IP-enabled to accommodate SIP and voice over IP.

No mentioning names, but to press the point about offering a proven IP PBX solution, we found some telling comments in another article by Owen Linderholm at voip-news.com, this one titled the *Top 5 Upstart IP PBX Vendors* (February 7, 2006).

"On its own, [a certain system] isn't simple to set up and isn't completely standards-based..." And, "You won't get Fortune 100 level support..."

If we say anything about a vendor and its solutions being proven, it's this: Do your homework. As we pointed out in this space before, if your enterprise is committed to finding a true IP PBX system, look closely at its back-end architecture with regards to a SIP carrier environment for IP telephony. More importantly, research the vendors you're targeting to see what their track record is by way of industry experience, implementation practices, support, and so on.

Unfortunately, many vendors now in the IP telephony arena remain unproven, primarily because they've only recently jumped on the VoIP bandwagon and have yet to work the bugs out of their IP PBX products. On the good side, though, a number of vendors are, indeed, reputable and offer significant industry knowledge along with the solutions they sell.

Proudly, Vonexus is one such vendor, bringing more than a decade of innovation and experience from its parent company, Interactive Intelligence, which has provided a complete SIP architecture and VoIP capability in its product suite since 2002. Moreover, in offering the pre-integrated all-in-one *Enterprise Interaction Center* (EIC) IP PBX, Vonexus has built EIC using the same multimedia software platform that Interactive Intelligence has implemented in thousands of enterprises and contact centers around the world since 1994.

That's why we take the meaning behind *Practical, Powerful, and Proven* seriously. Hopefully you'll take the same Three Ps approach to finding the best IP PBX for your enterprise. IT

Joseph A. Staples is Senior Vice President of Worldwide Marketing for Interactive Intelligence, Inc., and the company's wholly-owned subsidiary, Vonexus, Inc. (news - alert) For more on their suite of IP PBX and IP telephony solutions for the enterprise and IP contact center, contact Vonexus at 888-817-5904 (http://www.vonexus.com) and Interactive Intelligence (news - alert) at 317.872.3000 (http://www.inin.com).

Packet8's Bryan Martin

Rich Tehrani's Executive Suite is a monthly feature in which leading executives in the VoIP/IP Communications industry discuss their company's latest developments with TMC president Rich Tehrani as well as providing analysis on industry news and trends

In this issue, Rich speaks with Bryan Martin, chief executive officer at 8x8.

In its early days, 8x8 developed technology and sold products that enabled service providers to enter the emerging VoIP (define - news - alert) space. When the telecom downturn occurred, the company chose to become a service provider instead, marketing its offerings under the Packet8 (news - alert) brand name.

8x8 has its roots in video and retains a slew of industry patents. In fact, before many of today's VoIP companies came into existence, Packet8 was already in the business. Having visited the company's offices over the years, I was consistently blown away at how many patents adorned the walls at corporate headquarters.

8x8 has straddled the line between being a very well-known company and an aggressive entrepreneurial entity doing things others haven't thought of — or at least executed on.

I recently spoke with the company's CEO Bryan Martin about what his company is doing now and what the future may bring. Here is that interview:

RT: Tell me about the Packet8 Virtual Office service and why you feel it is such an important component of 8x8's offerings?

BM: In 1998, 8x8 bought a company, Odesei, which was the genesis of our Virtual Office technology. Odesei was later spun out as an 8x8 subsidiary, IP communications platform provider, Centile.

Shortly after the Packet8 residential offering was introduced, we decided to launch a small business VoIP service called Virtual Office. We saw a huge market opportunity for 3–50 extension organizations that could benefit immensely from a business VoIP phone service that not only offered a very low cost unlimited calling plan, but also provided corporate-class PBX functionality companies this size would not normally be able to afford. Although this is the small end of the market — in terms of business size — 99 percent of all U.S. businesses are micro-sized or home-based.

Virtual Office gives small businesses a complete business VoIP phone solution with robust PBX features they can afford, and eliminates the need for costly premises-based equipment with high maintenance fees. Our cost is $40 per extension per month, including unlimited calling to the U.S. and Canada. Our average customer spends about $400 each month, whereas, prior to switching over to Packet8, they were paying at least a $1,000 a month for their phone bill, including 5–7 cents per minute for local calls.

The average Virtual Office customer has 10 extensions and is a distributed or multi-location business. Take a local retailer, for example, with several regional stores that needs an easy and reliable line of communication between locations. With Packet8, they simply dial three numbers to connect with one another. In addition, they can request a single phone bill they could easily pay with a credit card — just one of many advantages we provide our customers.

RT: What Virtual Office features do your subscribers value most?

BM: Virtual Office includes the same powerful business-class phone features you'd find in high-end PBX systems like auto attendant, dial-by-extension, conference bridge, voicemail, call transfer, music on hold, and many more. The auto attendant is very popular, as is the Web portal, allowing drag and drop greetings to be created. There is also a receptionist application, if needed, but most of our customers use the auto attendant, which allows for an answering tree.

Customers can record greetings and upload WAV files of their greetings. They can also set up the system to ring multiple phones. With these and other features, like music on hold, a small company can easily sound like a large enterprise. A company with locations around the world can have 24/7 coverage and seem even bigger. Our voicemail is state of the art and conferencing is included as well — either can be accessed by three-digit dialing.

But, one of the most important features for subscribers is the option to

transfer over their existing number or choose any direct dial number (DID) for each extension, even one with a different area code. For example, a business can have a NY phone number for their office in Florida. They can also get an 800 number, if they prefer.

RT: What is your biggest competitive threat?

BM: In the near term it is Vonage; it has a known brand and many customers. That said, analysts argue that cable companies actually pose the greatest threat. I think that as mobility takes hold, the cable companies will have issues as well. In the long term, I have a lot of respect for SBC and this is where I believe the battle lines will be drawn. We need to seize some of their business customers. The longer war will be VoIP vs. Ma Bell.

RT: What about your BellSouth partnership?

BM: The partnership is strong and we launched our residential service with them late last year, where we provide everything except the marketing. In fact, BellSouth is coming out with its own small business service based on the Packet8 platform.

RT: What is happening beyond our shores?

BM: Internationally, we are looking for existing brands to partner with. For example, we are partnering with Dixons in the UK and Europe. The company is like Best Buy in the U.S. and has 1,200 stores. Dixons had a service called Freeserv, which was akin to NetZero in the U.S. Dixons sold the service to France Telecom, which then converted it to Wannadoo. When Dixons launched our service, they called it Freetalk, making use of the brand equity. Because Dixons is the largest retailer in the region, we have effectively been able to lock Vonage out.

RT: What about MVNO relationships? Is this coming?

BM: I've looked at structuring MVNO relationships, but none of the U.S. carriers have opened their networks to third-party VoIP carriers yet. I see potential in video, as wireless carriers may be interested in allowing their customers to have high quality video conversations.

RT: What other interesting things are happening?

BM: It is exciting to see how Yahoo! and Google bring new ideas to the space. We are getting to "everything over IP," which means everything, everywhere, and all the time. We will have access from any location, from any device, and we will control how and when we want to receive our personalized content. We will see tremendous innovation as the puzzle gets put together.

RT: How has being a public company affected you?

BM: Being a publicly traded company has two disadvantages. We had to grow the service from nothing, though now the business model looks good. It also has been tough for investors, as the numbers have not grown as quickly as they would have liked.

In addition, Sarbanes-Oxley has been a challenge, in terms of both cost and time. Year one was a tremendous investment and, now, year two requires time, but is much more manageable. On the plus side, there has been access to capital — we took in $60 million in the last three years. Also, we haven't had been exposed to debt deals or exploding VC financing.

RT: What about the Vonage IPO?

BM: We were not surprised to see they were losing $60 million a quarter. Once our investors saw what Vonage was doing, they said Packet8's numbers are not as bad as they originally thought.

We are on equal footing now that we are both public companies. Our other direct competitors have not had to break out the numbers, as VoIP is only part of what they do.

RT: Are you surprised a foreign service provider hasn't bought an American VoIP company?

BM: I am. For years, foreign telecom companies have wanted to buy a wireless company, which, of course, is fraught with regulatory issues. VoIP is a simple way for them to get into market. Regulatory uncertainties in the United States have likely deterred them.

RT: What do you think of Kevin Martin?

BM: We have done everything we can to comply with recent 911 mandates. We and others were waiting to see what the response would be late last year. As yet, nothing has happened; we have no guidance. The silence has him and others confused and we aren't getting any feedback.

RT: Where will we be in five years?

BM: Two major trends will dominate as we go into the future. Video will become prolific as bandwidth gets increasingly cheaper and more ubiquitous. In addition, wireless will continue to gain traction, though I won't profess to know whether WiFi or WiMAX will win their war. I hope, in five years, the wireless issue gets worked out. We should all have gigabits of access and application providers will go nuts. Business productivity will improve and new entertainment applications will come onto the scene.

The key takeaways from this interview are that the regulatory environment is, indeed, clouding the VoIP investment market. In addition, it would seem that video and wireless are going to play an integral part in our future. One point on which I thought I stood alone is the potential WiFi/WiMAX conflict. I get the feeling that WiFi will morph into something else with a larger range but with low-cost access points, allowing very robust competition to WiMAX (define - news - alert) in certain areas. I haven't heard anyone other than Bryan think that WiMAX and WiFi will compete.

One thing is certain: Packet8 will be battling aggressively with the old new Ma' Bell, the cable companies, and Vonage for a number of years. That is, unless a foreign carrier decides it is irresistible. IT

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TMC Labs Internet Telephony Innovation Awards 2006: Part II

This is our seventh installment of the TMC Labs Innovation Awards, recognizing the truly unique and innovative products and services within the VoIP (define - news - alert) industry. Our task in picking the most innovative products and services is always challenging. However, this year, we were pleasantly surprised to see new and truly unique products, such as Alcatel leveraging IP networks to solve radio interoperability issues between police, fire, and other emergency personnel. Choosing this product as "innovative" was a no-brainer. As more telephony applications are bundled with other IP services, including IPTV, video-on-demand, etc., the need for more bandwidth as well as managing that bandwidth for QoS becomes more critical. Thus, this year's awards feature more high-bandwidth focused products, such as Allot Communications' traffic management device, called the NetEnforcer AC-2500, which supports a whopping 5Gbps. Similarly, Occam Networks' BLC 6314 Transport and Optical Line Termination (OLT) Blade supports 10GigE (10 Gigabit Ethernet) of bandwidth.

TMC Labs proudly bestows 25 companies with a TMC Labs Innovation Award, which will be published in two parts in order to accommodate our in-depth write-ups for the winners. The complete winners list will be published in both the July and August issues; Last month we ran the descriptions from Alcatel to Lucent. This month we start with Meru Networks, concluding with XConnect.

2006 TMC Labs Innovation Award Winners (Full List)

COMPANY	PRODUCT NAME
Alcatel	*Alcatel My Teamwork Land Mobile Radio Conferencing and Collaboration solution (LMRCC)*
Allot Communications	*AC-2500*
Atreus Systems	*Atreus IP Service Provisioning Software*
Avaya Inc.	*Avaya one-X Quick Edition*
EdenTree Technologies, Inc.	*EdenTree Lab Manager*
Eicon Networks	*Diva Server SIPcontrol*
Envox Worldwide	*Envox CT Connect*
Esna Technologies, Inc.	*Telephony Office-LinX*
Global IP Sound	*GIPS Border Interface Engine (BIE)*
Grandstream Networks, Inc.	*GXV3000 SIP Video Phone*
Interwise, Inc.	*Interwise Connect version 7*
Lucent Technologies	*Lucent's Hosted IP PBX Service from the VoIP for Enterprise portfolio*
Meru Networks	**Meru Wireless Backbone System**
NICE Systems Ltd.	**NICE Contact Center Interactions Solution, VoIP Enhancement**
Occam Networks	**BLC 6314 10GigE Transport and Optical Line Termination (OLT) Blade**
Paragon Wireless Inc.	**PWTW-1100**
RADCOM	**R70 Probe**
RingCentral	**RingCentral Online 3.0**
Sangoma Technologies	**A200 FXO/FXS Analog Telephone Support System**
ShoreTel, Inc.	**ShoreTel 6.1**
Sipera Systems	**Sipera IPCS 310**
SPIRIT DSP	**TeamSpirit Mobile**
UCN, Inc.	**inContact**
Verizon Business	**IP Web Center**
XConnect Global Networks, Ltd.	**XConnect Alliance**

**companies appearing in italics appeared in our July 2006 issue with a full description.*

Meru Networks

Meru Wireless Backbone System

http://www.merunetworks.com

The Meru (news - alert) Wireless Backbone System differs from most other wireless networking systems by using a "blanket," in which every access point uses the same 802.11a and b/g WiFi channels. It essentially aggregates all the GHz frequencies together to give you more bandwidth — an aggregate of 240 MHz of backbone spectrum. This product aims to replace the wire going to the access point (AP) and uses wireless instead, essentially making this "zero-wire" to the desktop and the backbone. This solution enables an enterprise to go all-wireless. The WLAN is the only network — from the core to the device. Meru Networks claims that this product is the only one in the market that can support both voice and data over multi-hop wireless backhaul, with a switch-like experience over an all-wireless enterprise backbone, essentially making them the only solution to "unwire" end-to-end for both voice and data.

Meru Networks told TMC Labs, "In the future, all networks will work this way. Wireless CATx will be the de facto connectivity mechanism. The industry segments that are first adopters of this product are: verticals such as universities, manufacturing, retail, hotspots (airports, convention centers, etc.) Horizontal targets are in geographies where old buildings are not pre-wired. This entails large markets in Asia including China, India, Japan, and the remaining Pacific Rim."

The market for this type of innovative product will be especially strong in the SMB, which will enjoy very low deployment cost and no radio frequency (RF) expertise requirements, with the added benefit of the elimination of wire for both voice and data, as well as easier moves/adds/changes.

The Meru Wireless Backbone System provides hierarchical aggregation of capacity over the air similar to EtherChannel, and the ability to provide multi-hop QoS and end-to-end security, and the ability to provide link redundancy and dynamic self-healing.

Meru's WLAN product claims to be the first to enable the "unwired office" — entirely cutting the access cable for voice and data. Osaka Gas, one of Japan's largest utility companies with 50 offices and 6,000 employees, went entirely wireless — no wire for voice or data in any office barring disaster recovery — last year.

Some other unique features include full-duplex wireless channels, a significant innovation based on patented antenna technology; Multi-hop QoS for commercial-grade voice+data support, extending Meru's Air Traffic Control access technology to the wireless backbone; Seamless handoff and Virtual Cell, extended over multi-hop wireless backbone. If the clients are on 802.11g, the system can use the 802.11a channels for the backbone. Instead you can place all the data clients on one 802.11g channel and use the other channels for voice, and the "a" channels for the backbone.

When 802.11n arrives, giving more capacity, the system will be compatible with it. Supporting both voice and data over a multi-hop wireless network is unique in the industry and, Meru believes, the pioneer of a trend that will eventually replace wire with wireless entirely. While we're not ready just yet to give up our super fast Gigabit-wired Ethernet connection, in favor of a 100 percent wireless solution that is slower, we admire Meru for their vision of an unwired world.

NICE Systems Ltd.

NICE Contact Center Interactions Solution, VoIP Enhancement

http://www.nice.com

NICE (news - alert) has been one of the leading call recording and monitoring companies in the world has even diversified their call center feature/function portfolio by adding workforce management with their Performix acquisition and recent IEX acquisition. NICE's next-generation VoIP active recording solutions, with specialized contact center functionality, enable efficient management and administration of branches and agents from home and enhance centralized recording capabilities for distributed environments. By eliminating the need for setup, administration, and management of recording hardware at branches or remote sites, IP-based active recording profoundly reduces the overhead and complexity associated with previous generation VoIP recording. NICE's VoIP active recording solutions enable efficient and centralized management and administration of branches and agents from home.

The new offering is the first to include solutions for active recording in Nortel's newly developed Duplicate Media Stream over IP (DMS-IP) architecture, active recording for Cisco's CallManager, as well as offer new redundancy options for Avaya's Communication Manager API, and IP-phone applications. Nice also supports passive monitoring and includes support for all the major IP telephony vendors, including Alcatel, Aspect, Avaya, Ericsson, IPC, Mitel, NEC, Rockwell, Siemens, and others.

The nice thing about NICE's VoIP recording solutions (pun intended) is that they are an integral part of the company's unified product architecture and suite of solutions. This means a smooth migration to VoIP that is transparent to the user, thus providing investment protection. NICE also offers hybrid solutions that cover the entire spectrum of customer interactions management, combining traditional and VoIP, liability, and quality monitoring for contact centers. NICE offers software-only, scalable VoIP solutions that are certified by the world's leading VoIP switch vendors, addressing small-scale to large, multi-site, high-end environments. Finally, NICE offers speech analytics that can perform word spotting and even determine customer's state of mind (i.e., angry, excited) to aid in agent training and improve the customer relationship.

Occam Networks

BLC 6314 10GigE Transport and Optical Line Termination (OLT) Blade

http://www.occamnetworks.com

Occam's (news - alert) razor has always been associated with the concept of simplicity. For instance, Occam's razor postulates that the answers to most scientific phenomenon are usually the simplest rather than the more complex. Well, Occam Networks aims to provide the simplest solution to the complexity of bandwidth hungry applications running on a service provider's network. Occam's newest blade for its flagship BLC 6000, the BLC 6314 10GigE Transport and Optical Line Termination (OLT) blade, adds a whopping dual 10 Gigabit Optical Ethernet and multiple 1 Gigabit Ethernet interfaces to the BLC 6000 in a single, environmentally hardened blade. The Occam BLC 6314 claims to be the first and only product that delivers this level of bandwidth with a sub-50-millisecond failover rate. This enables telcos to create highly resilient access networks that can deliver a TV signal with "five nines" of reliability.

The BLC provides an abundance of symmetrical bandwidth that enables telcos to easily create very high capacity access networks with 10 Gigabit Ethernet redundant transport rings at a fraction of the cost of traditional SONET-based OC 192. The BLC 6314 can be used for fiber to the home (FTTH) optical line termination. When combined with the Occam BLC 6312 blade, the BLC 6314 creates a high-capacity FTTP Optical Line Termination system with integrated 10 Gigabit Ethernet transport that can serve more than 250 subscribers with up to one Gigabit of full-duplex bandwidth each. Having this kind of mega-bandwidth makes broadband cable and DSL look like dial-up!

Also, the BLC 6314 can also be used for higher capacity access network aggregation using the company's Ethernet Protection Switching (EPS). It provides resilient 10 Gigabit Ethernet aggregation rings for BLC 6000-based copper and fiber access networks. As bandwidth-intensive applications, such as HDTV, IPTV, VoIP, VoD, P2P, etc., continue to consume more bandwidth, service providers are looking for solutions to their subscribers' seemingly insatiable bandwidth appetite. Fortunately, Occam Networks' BLC 6000 Broadband Loop Carrier (BLC) uses IP as a simple, common service delivery protocol for all services and Ethernet transport to provide the economical, highly scalable bandwidth needed for delivering today's and tomorrow's new services.

Paragon Wireless Inc.

PWTW-1100

http://www.parawireless.com

(news - alert) The ability to seamlessly make and receive phone calls on both cellular and WiFi networks is not a dream for tomorrow — it's happening today with companies like Paragon Wireless leading the way. The Paragon Wireless PWTW-1100 is the first commercialized SIP-based, voice optimized, dual mode (GSM/WiFi-VoIP) handset in the world. In fact, they started shipments in China in March 2006.

Paragon Wireless has developed a patent pending technology that allows an

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optimal integration of two wireless radios (cellular and WiFi) in such a way that problems like power management, roaming, FMC, Mobile VOIP and security (WPA supported) are solved. It can automatically roam between radio networks in less than 50ms.

Paragon has created the first commercialized product in its segment and the technology solution of integrating WiFi and VoIP with cellular radio in a way that also allows great performance. Paragon claims that independent tests have been conducted by China Telecom testing standby and talk times of their dual mode phone to confirm its good performance. The handset sports four hour talk and 72 hours standby time, even when both its WiFi and GSM capabilities are running. It has strong VoIP support with acoustic echo cancellation, and a suite of codecs that includes G.711, G.723, AMR, and LPC, as well as a dynamic jitter buffer and a voice activity detector.

The phone features the most common PDA applications, including calendar, schedule manager, alarm clock, voice recorder, and more. It also features IM, e-mail, an Internet browser, and a VPN client. With its built-in SD memory card slot and MP3 player you can play music as well. It utilizes Intel's PXA271 processor, a 2.4-inch TFT touch screen (320x240 QVGA, 260k colors, and backlit), a built-in speaker and microphone, and a 1.3 megapixel CMOS camera. The phone includes a simultaneously active 802.11b interface and a tri-band GSM/GPRS interface. The product is a normal cell phone size handset with large TFT display. It's worth mentioning that both radios use the same user interface. While in WiFi call the user can receive cellular call and switch between the calls if needed.

RADCOM

R70 Probe

http://www.radcomusa.com

(news - alert) As enhanced applications converge onto IP networks and as service providers look to deploy Triple and Quadruple Play services, the ability to diagnose and troubleshoot network issues becomes even more critical and more complex. Fortunately, RADCOM's R70 provides what you could consider a "triple play packet sniffer." The Linux-based, intrusive and non-intrusive R70 probes monitor and analyze network traffic. The R70 probes are a high-end, upgradeable component of RADCOM's Omni-Q, a network and service monitoring, analysis and troubleshooting system for public IP, cable, and advanced cellular networks. R70 probes give operators total visibility into the session and application levels, with full seven-layer analysis. They provide operators with a comprehensive and correlated view of all IP-based services, including VoIP and IPTV, complete with service integrity metrics from their subscriber's perspective.

The R70 is the first probe in the market to process data at 10 Gbps. The R70 leverages the company's existing, proprietary technology (the Gear Chip) to achieve its high level of performance that exceeds the processing limitations of most of not all other existing probes in the market. It is a unique monitoring and analysis tool that, in a single product, addresses the needs of ILECs, MSOs, and cellular operators.

Service providers moving to an all-IP core network architecture will find that RADCOM's R70 offers a comprehensive feature set to monitor and analyze their networks. Because it gives operators complete visibility in the IP-based service running over the network, it enables early stage fault detection, pre-emptive maintenance and optimization. Additionally, when operators seek to introduce new services to create new revenue streams, RADCOM's R70 probe-based monitoring system reveals subscriber behavior providing valuable insight into the new service adoption process.

RingCentral Online 3.0

RingCentral Online 3.0

http://www.ringcentral.com

(news - alert) The ability for customers to reach you is important for customer retention. Large corporations have implemented many technologies to that end such as ACDs, digital fax servers, and more. Unfortunately, small businesses or entrepreneurs often cannot afford these systems. RingCentral aims to address to provide advanced contact technology to small corporations with 10 or fewer employees, mobile workers employed by larger firms, and road warriors. Small business customers include owners of home-based businesses, consultants, non-profit organizations, and the like.

RingCentral provides advanced telecommunications services, unifying landline, wireless, VoIP and e-mail communications, and even exceeds services available to large enterprises with a fixed PBX. RingCentral provides dial tone independence, full fax integration, and does not require customers to purchase any new hardware or software — the service works seamlessly with existing landline and mobile phones. RingCentral gives you the ability to manage messages, faxes, and call records over the Web, on a PC, and over the phone

RingCentral Online combines a toll-free and/or local number with advanced call management, PBX, voicemail, and Internet fax, enabling customers to automatically screen, forward, and place calls, take voicemail, send and receive faxes, and receive message alerts.

RingCentral utilizes their Call Controller tool, a desktop pop-up window that allows customers to react to calls in real time without having to touch their phone. The Call Controller toll gives customers four clickable choices for handling the call: accept, reject, send to voicemail, or reply with a text-to-speech response. This text-to-speech technology is a very innovative feature. Essentially, while customers are on hold, you can type a message, which will be translated to speech and broadcast to the caller.

RingCentral has powerful answering rules including call scheduling, filtering, and routing technology, which allows customers to greet callers with various messages and route them according to the day, time, date range, and caller ID. Customers can route calls to voice mailboxes, extensions, and any phone in the world, as well as block calls with exceptions, allowing users to set specifications and overrides for VIP customers or family members.

Their latest feature, RingMe, let's customers embed a button into their Web sites or e-mail signatures so that online

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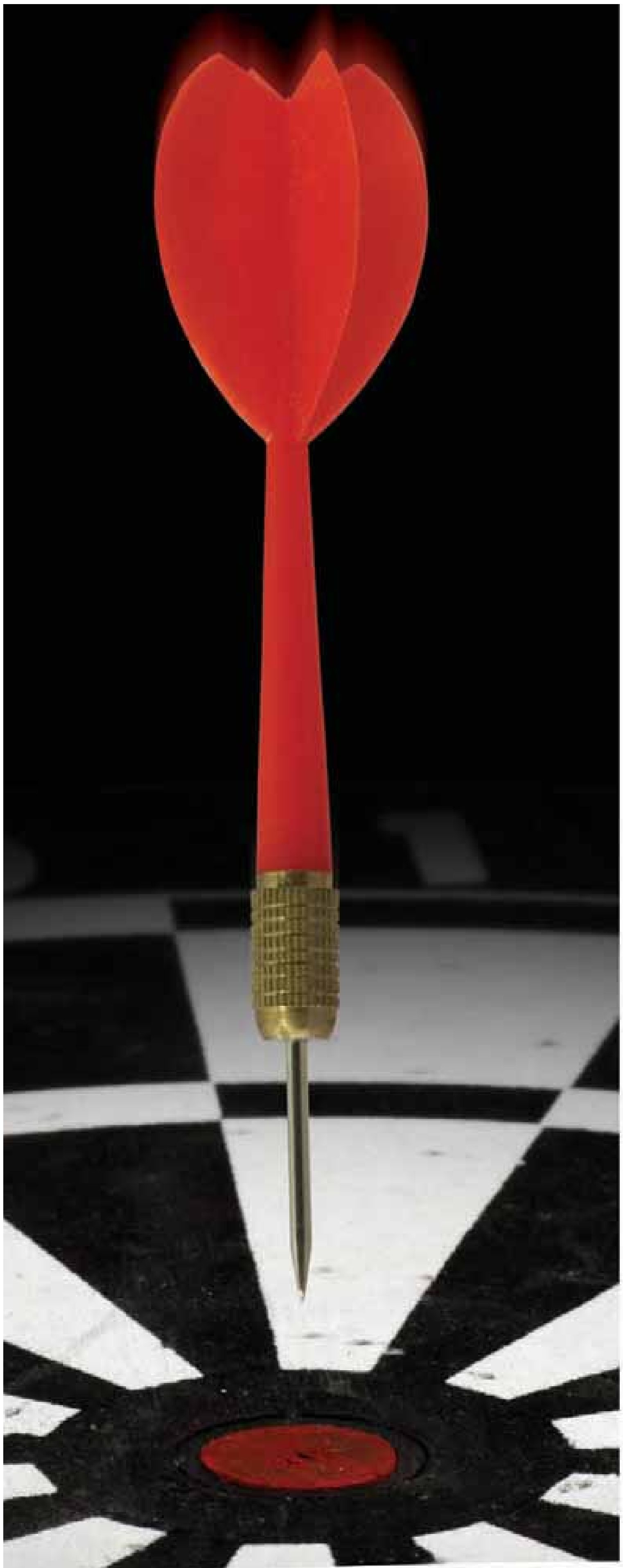

visitors can reach them by phone with click-to-call technology. Additional new and enhanced features include FaxOut for sending faxes from any computer, Fax From E-mail, and Caller Preview function, whereby customers can hear who's calling before deciding what to do with the call. RingCentral also features integration with Microsoft Outlook and Outlook Express.

Sangoma Technologies

A200 FXO/FXS Analog Telephone Support System

http://www.sangoma.com

TMC Labs has tested Sangoma's (news - alert) cards firsthand, so we know how versatile and innovative these cards are. Sangoma is well known for architecting their telephony hardware to support several softpbxs, including the popular Asterisk platform, Yate, OPAL PBX/IVR projects, as well as other Open Source and proprietary PBX/Switch/IVR/VoIP gateway applications.

Sangoma's A200 and Sangoma's REMORA system together comprise the FXO/FXS version of the company's Advanced Flexible Telecommunications (AFT) hardware designed for optimum support of analog voice traffic. The A200 series provides analog connections for FXO or FXS in expandable solutions from two ports to 24 ports.

One of the most unique aspects of all of Sangoma's products is that they are easily expandable to go from two ports to 24 ports utilizing the existing card and adding modules. This type of flexibility and field upgradeability has never been seen in this space before. Another innovative feature that Sangoma introduced in many of their products, including the A200, is a telco-grade hardware echo canceller of 128ms.

The A200 solution supports any combination of up to 24 FXO or FXS connections. A single PCI slot host connection for all ports ensures common synchronous clocking for all channels. The base AFT architecture is shared with Sangoma's A101, A201, A104, and soon to be released A108 cards, ensuring common 3.3V/5V, high performance PCI compatibility. Just as with all Sangoma's cards, the A200 has field upgradeable firmware to take advantage of hardware and software improvements as they become available.

The A200 consists of a REMORA daughterboard mounted on the AFT PCI card. The REMORA card has two sockets, each of which can accept a FXO-2 or FXS-2 module. Each FXO-2 or FXS-2 module supports two FXO or FXS ports respectively. Up to five additional REMORA daughterboards can be mounted in empty slot positions beside the A200 assembly connected to the A200 by a backplane bus connector. Importantly, the A200 fits into the 2U Form factor and short 2U compatible mounting clips are available for installation in 2U rackmount servers. It also features a 32-bit bus master DMA data exchanges across PCI interface at 132Mbytes/sec for minimum host processor intervention. Finally, it is fully PCI 2.2 compliant, making it compatible with all commercially available motherboards, and it performs proper interrupt sharing — something that often plagues other telephony hardware.

ShoreTel, Inc.

ShoreTel 6.1

http://www.shoretel.com

The ShoreTel (news - alert) system is a fully distributed IP phone system with no single point of failure, making it a reliable solution for multi-site enterprises. Created specifically to run enterprise-class telephony and voice services across multiple sites, ShoreTel's systems do not suffer from the complexities of traditional systems evolved out of legacy voice or data switches.

The ShoreTel system offers distinct benefits because of its unique, distributed architecture. For instance, call control is distributed to voice switches and voice applications, including voicemail and automated attendant that run on standard server hardware anywhere on an IP network. There is no single point of failure and there is a single system image across all geographies with complete feature transparency, making the system very easy to manage.

ShoreTel's switches and software are designed to provide easy deployment, and work well with their ergonomic and feature-rich ShoreTel IP phones. In fact, their latest phone is the ShorePhone IP 212k, an ergonomic IP key system telephone with 12 programmable buttons, exceptional audio quality, and big LCD display. Other features debuting in the latest ShoreTel release include a gigabit IP phone, the IP 560g, a new staff IP phone, the IP 230, Centrex flash capabilities, personal-assistant and Caller-ID enhancements. The ShoreTel system includes comprehensive key system functionality and it allows bridged call appearances to span locations, thus, virtualizing key system behavior across the enterprise.

One of the most unique aspects of ShoreTel's product line is that it's the same exact hardware, whether you are 10 employees or 10,000 employees — a simple software license increases the number. A single hardware platform means less training for the reseller and VAR channel as well as easy upgrades. Competitive solutions include distinct products for different sizes of companies, forcing folklift upgrades when customers move between platforms and impacting customer service levels by requiring partners and customers to support multiple, complex products.

Sipera Systems

Sipera IPCS 310

http://www.sipera.com

(news - alert) Edwin Andres Pena made headlines when he hacked into VoIP providers' networks, stole $1 million worth of VoIP minutes, and then resold them. This is even more evidence that intrusion and detection systems (IDS) specialized to handle VoIP are needed in our industry. The Sipera IPCS 310 is a comprehensive IP Communications Security appliance that includes a complete suite of security features for protecting session-based, real-time IP communications applications including VoIP, IM, multimedia, and other collaboration tools.

It can be deployed within the enterprise network in front of communications servers, between the voice and data VLAN, or along the SIP trunk, to prevent any potential attacks in real time. The Sipera IPCS 310 protects both the infrastructure and end users from a number of malicious, application-specific attacks and service abuse from DoS/DDoS flood and fuzzing attacks to


October 11-13, 2006
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more sophisticated reconnaissance, stealth, and VoIP spam attacks.

Sipera addresses all the required security functionality, such as firewall, IDS/IPS, DoS prevention, network level correlation, and spam filtering, while implementing sophisticated techniques to ensure unique VoIP threats are proactively recognized, detected, and eliminated. These techniques include anomaly detection and behavior learning. The Sipera IPCS 310 continuously learns call patterns and endpoint fingerprints, in addition to being able to constantly analyze raw event data and take automatic action, which gives the security solution the ability to evolve and adapt to effectively counter any threat. This level of sophistication is the only way to minimize false negatives (allowing bad calls) and positives (blocking good calls). This level of sophistication is the only way to identify both stealth attacks and VoIP spam, which are difficult to detect, as the real-time nature of VoIP does not allow the security system the luxury of storing the call while it's analyzed, before sending it on, as is the case with e-mail or other non-time sensitive IP applications.

Additionally, the Sipera VIPER lab, comprised of top wireless and VoIP vulnerability research experts who proactively identify SIP, UMA, and IMS vulnerabilities, has catalogued thousands of threats that can be launched against these networks to date. This expertise forms the foundation of the Sipera IPCS 310 that protects all VoIP elements from a variety of malicious attacks and service abuse.

SPIRIT DSP

TeamSpirit Mobile

http://www.spiritdsp.com

(news - alert) With dual mode phones and high-speed Internet connectivity (e.g., EV-DO, WiFi) on mobile phones, customers are looking for to leverage softphones on their mobile phones for inexpensive VoIP minutes and more "connectedness." TeamSpirit Mobile empowers OEMs with powerful VoIP functionality, bringing softphone applications to mass-market mobile devices. This product makes it easy for the equipment manufacturers and mobile software developers to build feature-rich, easy-to-use softphone applications. These innovations will allow the end users to quickly adopt wireless IP capabilities of their mobile phones and enjoy the benefits of VoIP communication. TeamSpirit Mobile also includes a Voice Engine Software Development Kit (SDK) for mobile devices to help equipment manufactures develop high performance softphone applications.

Spirit DSP claims that TeamSpirit Mobile is the fastest voice engine running on mobile devices today. They also claim that their mobile VoIP engine is the industry's first to run a VoIP application on devices with CPU clock rate of 200 MHz or less, for example, the Qtek

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Site Sponsors — May 2006

CYGCOM

www.cygcom.com

CYGCOM's Speed-To-Market Program for ISVs and OEMs facilitates delivery of certified voice systems through experienced voice reseller channels anywhere in North America. As an Intel Authorized Distributor and Microsoft Platinum OEM, CYGCOM combines 12 years of multi-vendor voice support with Arrow Electronic's design and manufacturing capabilities and ensures voice systems can be easily installed and supported.

805 SW Broadway, Suite 2440 | Portland, OR 97205
Phone: 1.800.407.7928 | Fax : 905.946.9612

envox

www.envox.com

Envox Worldwide is a leading global provider of voice solutions. The company's software and related services dramatically reduce the time, cost, and complexity of creating voice solutions enabling customers to significantly reduce operating costs, improve customer satisfaction and retention, and generate new revenue streams.

2000 West Park Drive | Westborough, MA 01581
Phone: 508.871.7604 | Cell: 617.792.3316 | Fax: 508.366.0009

INTER-TEL®

www.inter-tel.com

Inter-Tel offers voice and data communications solutions for customers of all sizes, whether you are a large enterprise, small to medium business, or have residential home-based business needs. Inter-Tel's diverse product line includes network communications solutions based on Voice over Internet Protocol (VoIP) technology, converged (IP PBX) platforms and IP-centric systems.

7300 West Boston Street | Chandler, AZ 85226
Phone: 480.961.9000

PRONEXUS

www.pronexus.com

VBVoice reduces your IVR development time through a drag-and-drop interface, industry-standard programming languages, integrated support for VoIP, speech recognition and Text-to-Speech engines. VBSALT simplifies and accelerates the development of speech-enabled IVRs on the Microsoft* Speech Server* platform through a graphical interface, encapsulated best-practices for VUI design, and generation of SALT tags and Jscript.

750 Palladium Drive, Suite 200 | Ottawa, Ontario | K2V 1C7
Phone: +1.87.PRONEXUS | Fax: +1.613.271.8388

european voiceway distribution

www.voiceway.fr

Value added distributor, Voiceway assists application providers, integrators and service providers customers in their telecom projects through technical assistance, engineering, installation and integration, trainings supports.

146, Boulevard Voltaire | 92600 Asnières-sur-Seine | France
Phone: +33 1.40.80.91.20 | Fax : +33.1.40.80.91.21

Alliance Systems®

www.alliancesystems.com

Alliance Systems, Ltd. designs, builds, ships, and supports communications and computing equipment providing the infrastructure for VoIP, wireless, security, and other applications. Through its engineering, manufacturing, and value-added services, Alliance helps customers optimize their businesses by enhancing profitability and reducing time to market.

Ludwigstrasse 8 | D-61348 Bad Homburg | Germany
Phone: +49.6172.2796.0 | Fax: +49.6172.2796.13

EDGEWATER NETWORKS, INC.

www.edgewaternetworks.com

Edgewater Networks develops networking and security products that ensure the secure and reliable delivery of IP based voice and video for service providers and enterprises. Edgewater Networks' award-winning solution simplifies setup and management of converged networking applications and provides proven operational savings from the remote office to the core of the network.

2730 San Tomas Expressway, Suite 200 | Santa Clara, California 95051
Phone: 408.351.7200 | Fax: 408.727.6430

intoto

www.intoto.com

Intoto is the leading software ODM of integrated security, wireless and voice software platforms for networking and communications equipment. Intoto's software solutions enable OEMs to reduce development costs and speed time-to-market.

3100 De La Cruz Blvd, Suite 300 | Santa Clara, CA 95054
Phone: 408.844.0840 | Fax: 408.844.0488

8310 and the iPAQ 6340.

TeamSpirit Mobile addresses the most challenging problems encountered in mobile IP applications, such as ensuring rich voice and quality video while dealing with limited system resources of a mobile device. This solution is optimized for appliances running under Windows Mobile 5.0, Windows Mobile 5.0 SmartPhone Edition, Windows PocketPC 2003, or Windows Smartphone 2003 operating systems. TeamSpirit voice engine has been specifically optimized for a dozen mobile platforms: OMAP 850, 1510, 1030 and others; Intel Xscale family; and any ARM9 processors starting from 168 MHz.

SPIRIT's proprietary technologies deliver exceptional voice quality and eliminate echo and surrounding noises while utilizing minimal resources. Their voice engine also includes optimized codecs, PLC, voice enhancement components (Full-duplex Acoustic Echo Cancellation, Automatic Gain Control, Noise Suppression) and adaptive jitter buffer. It also supports a built-in or third-party SIP stack, provides network support (RTP/RTCP) and call control API, and integrates network and voice processing subsystems. It can also be enhanced with video support using standard H.263 and H.264 video codecs. Finally, since it has low resource consumption, it prolongs battery life on mobile devices.

UCN, Inc.
inContact
http://www.ucn.net

(news - alert) InContact is a Customer Interaction Management service that combines multimedia customer contacts (voice, chat, e-mail, fax), skills-based ACD for both in-house and at-home agents, and a robust and completely customizable IVR. InContact is a redundant, hosted solution that leverages multiple core network carriers for maximum uptime. The product line addresses the needs of companies seeking to improve their customer contact experience by offering an integrated solution that addresses all customer interactions, including voice, e-mail, chat, and fax. Since the application is hosted, it grows as your organization grows and can be done very quickly. In fact, UCN was involved with quickly setting up the largest Spanish-speaking call centers spanning four countries to assist in Global Telesourcing's Red Cross/Katrina fund raising efforts.

UCN is unique in that they bundle two critical service packages together, namely the contact handling service application and phone service. By providing a single vendor solution, UCN customers get a single bill and single point of contact to resolve issues. Essentially, they are both the carrier and the enhanced software service provider. UCN incorporates a robust visual drag-and-drop programming capability, allowing the company to customize the IVR and skills-based ACD to meet their specific needs on their own, without the need for professional services.

InContact is highly customizable and can easily make moves, adds, and changes on the fly using a Web-based management tool. You can also customize the management dashboard (displays of reports and statistics in real time). InContact also offers database integration with the leading CRM database programs.

UCN is one of the true VoIP innovators building a nationwide IP network that was one of the first national, commercial VoIP networks in the country — going operational in 1997. The product was the first to deliver voice calls to the customer site via traditional TDM services, enabling the customer to keep their existing phone equipment yet take advantage of IP services — but hosted within the network.

Verizon Business
IP Web Center
http://www.verizon.com

Verizon, (quote - news - alert) a traditional carrier, expanding into providing advanced call center applications, is more proof that carriers are moving beyond simply providing "dumb voice" circuits. Verizon's IP Web Center is a network-based IP Contact Center offering that supports inbound/outbound telephony, Web call back, scheduled call back, Web chat, collaboration, e-mail, and fax.

Verizon Business married its Voice over IP service to its Contact Center service expertise to deliver one of the industry's first end-to-end contact center solutions. By combining the power of Verizon Web Center with the Verizon Voice over IP suite of services, businesses can reap the benefits of VoIP across their entire enterprise operations and achieve new capabilities and cost savings afforded by IP telephony. Some powerful features include unified messaging, ACD/IVR, quality monitoring, intelligence contact routing (across multiple media types), ability to create guest supervisors, pre-announced caller information (whisper announcements), enhanced statistical information, and remote agent capabilities.

Verizon Web Center and Verizon Voice over IP now share the same network infrastructure and customer premises equipment so Verizon Business can activate a wide range of IP telephony services, including IP Web Center, Hosted IP Centrex, IP Integrated Access, IP Flexible T-1 and IP Trunking, at a given company location.

IP Web Center is well suited for mid-sized businesses that want to expand the feature set of their existing contact centers. Since IP Web Center is a hosted solution that simply requires a phone and a broadband connection, large companies can also use the service for agents to make and receive calls anywhere in the United States, thus reducing the need to expand hardwired contact center facilities. It's worth mentioning that companies are increasingly moving to the IP contact center, taking advantage of an IP network environment to become more flexible and realize infrastructure cost savings.

Verizon Business's newly expanded pricing options let customers pay as they go for IP-enabled Web Center services. Businesses only pay a monthly per agent price, plus call transport fees and associated IP phone equipment costs. This works well for companies that need to increase agent levels during busy holiday times and then reduce them when normal call levels resume.

XConnect Global Networks, Ltd.
XConnect Alliance
http://www.xconnect.net

XConnect (news - alert) has a unique peering, settlement-free (zero-cost call),

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multilateral alliance. These benefits include more advanced call features, improved call quality and lower or zero call costs. These benefits are achieved by allowing members to deliver calls between their subscribers without involving unnecessary intermediaries and, where both call parties are VoIP subscribers, by delivering calls via pure end-to-end IP connectivity and completely bypassing the PSTN and its associated limitations and costs.

XConnect has created a globally distributed 'plug and peer' network that they tell us is the largest ENUM directory and the largest peering network in the world. XConnect provides what they claim is the first and only carrier-neutral network that resolves the challenges of VoIP interconnection, including numbering, settlement, interoperability, security, identity, and privacy across a globally distributed network of VoIP peering points.

The fast growing XConnect Alliance offers multilateral settlement-free exchange of traffic between VoIP service provider members. Alliance members peer on a settlement-free basis and pass on Zero-Cost or reduced cost calls to their customers. Calls are delivered via end-to-end IP connectivity, eliminating the need to traverse the PSTN, which would otherwise incur per minute costs, reduce call quality (because of limited bandwidth and repeated data-voice-data signal conversion), and preclude sophisticated features such as three-way calling, video calling etc.

XConnect offers a peering solution that combines ENUM Directory Management with multi-protocol signaling interoperability (i.e., H.323, SIP, IMS, IAX, PacketCable, etc.) as well as unique VoIP security features, such as avoiding misappropriation of misuse of the numbering database and sophisticated methodologies to identify, detect, and prevent Spam over Internet Telephony (SPIT) and CallerID spoofing.

XConnect has implemented genuine Caller ID authentication, validation, and normalization in a VoIP environment (based on a number of SIP messaging fields including FROM, CONTACT, remote_party_ID, and p_asserted_identity).

VoIP providers strongly prefer to have the ENUM numbering database available locally to facilitate speedy queries necessary to determine when calls can be routed to a peering partner. At the same time, they are concerned about their sensitive client numbering data being distributed around the world to competitors. XConnect has developed a Local Directory Server (LDS) deployed locally, while also preventing misappropriation, misuse, and data-mining of the ENUM database data. The LDS can perform thousands of queries per second while continuing to guarantee the confidentiality of each operator's own ENUM data. IT

Open Source Telephony Is Cheap to Buy, but Can Be Costly to Manage

By David Mandelstam

Anyone who has had the experience of installing a proprietary PBX or key system will know firsthand how the system itself — expensive as it is — can represent simply a down payment for a long and costly ownership experience. In the case of Open Source telephony, where initial costs are much lower, the ratio between first costs and total cost of ownership can be even more marked and can be the source of considerable frustration.

For any business, the telephone system still represents the most important contact point to the outside world. Notwithstanding the rise of e-mail, messenger services, even cell phones, the main contact portal for any company still remains the telephone system connected one way or the other to the PSTN. For some business, like call centers, the system being down results in immediate, quantifiable, and crippling losses, but even for less telephone-centric businesses, loss of PSTN (define - news - alert) access, even for a few minutes, is traumatic. So, it is important to realize that often the biggest hourly expense related to a PBX of any kind is the cost of down time. A billing rate of $300 per hour looks like a bargain when your company is entirely without phone service.

This is something to think about before deciding to build your company PBX using free, community supported beta software on a combination of the cheapest PC you can find and semi-experimental hardware of questionable quality.

The open source PBX implementations, such as Asterisk, support features that would be beyond the reach of small businesses and, in fact, beyond the reach of almost anyone. They provide the flexibility to integrate voice and voice processing into data and operational systems. They provide an easy way to transition into the world of VoIP. But, in many cases, the prime objective has been the saving of money, in particular, first cost.

The cost savings relative to proprietary equipment is considerable. But, don't be tempted to cut corners to extract maximum savings. Rather, treat the savings as a windfall and plan to invest a portion of the savings in system integrity.

For installation, set-up, and support, it is tempting to imagine that, because you have a Linux system administrator on hand, he can easily handle that PBX. After all, the soft PBX is only another Linux application among others and, once installed, the amount of work on a PBX should be fairly limited.

It's true that any competent Linux system administrator is perfectly capable of installing and maintaining one of the open source PBX implementations. There are no special arcane skills required and the configuration files and setup scripts are arranged logically and are fairly easy to understand. The problem, however, is that anyone in charge of a single PBX is going to be working on the system sporadically and, inevitably, under pressure. Skills picked up twelve months ago are going to be rusty when dealing with today's new crisis.

For businesses that are not primarily in the telephony business, it is probably good practice to make use of an outside consultant. A consultant is presumably looking after many systems and has sharp skills. She has also doubtless picked up some neat tricks in the field and has experience with which parts and versions of the open source telephony project work well and which are still experimental. The outside consultant is usually not very expensive because, unlike your old PBX consultant, she very seldom has to come on site. Almost everything she needs to do can be done through the magic of the Internet and SSH, from debugging driver problems, through setting up of dial plans and provisioning telephones.

What about hardware? Here again, a little investment goes a long way. Don't buy the cheapest motherboard/PC combination you can buy. Use a server supplied by one of the "big three." They are not that much more expensive, and at least you know that the different components have gone through a solid QC program and have been tested working together.

RAID is a pretty good idea. Disks are the least reliable component of any system and they do die. Backups are often not quite up to snuff when disaster strikes. RAID represents a small cost premium for considerable insurance.

If your PBX has a traditional PSTN access there will be additional hardware needed to provide the analog or digital interface. Choose this hardware with an eye to quality. If you are not careful, you can spend endless person-hours integrating PSTN hardware and getting it to work with your chosen system.

Remember that, as a general rule, hardware is less expensive than labor. If you can reduce person-hours by substituting hardware, it is almost always money well spent. For instance, hardware echo cancellation on a six-port FXO system at about $50 per port seems very expensive. But, if you compare it to the cost of several hours of system tuning needed to get software echo cancellation to work properly, it's a bargain. And if the system grows, it is nice to know that you have no further costs for echo cancellation.

A popular alternative is to install a pre-packaged, pre-configured system based on one of the Open Source projects. These can save enormous headaches because they have been through a testing program as a complete package. The packages include expert help in setting up the system and usually the user interfaces have been engineered for easy self-configuration with minimal risks. The costs are higher than the components of a homegrown solution, but the TCO is incomparably better. The only drawback to these PBX systems is that inevitably the customization and pre-packaging limit the flexibility. However, there is nothing to stop you configuring by hand to add features as you gain confidence. After making good backups, of course!

Open Source telephony projects can provide spectacular bang for the buck when compared to commercial closed systems. But, you should make sure that the goal of reducing costs does not compromise system reliability. It is certainly possible to achieve commercial levels of reliability and uptime for a system based on solid, reliable, hardware and software and with a competent, responsive support team. IT

David Mandelstam is President and CEO of Sangoma Technologies. (news - alert)

Keeping Tabs on Sangoma

By Greg Galitzine

Sangoma Technologies has lately been focusing on developing partnerships.

The Ontario-based company recently announced its partnership with Vyatta to advance the market for network routers and firewalls based on open source Linux technologies. Vyatta's Open Flexible Router (OFR) will provide the protocol software and Sangoma's WANPIPE PCI cards will supply cost-effective WAN connections. The goal of this partnership is to drive adoption of open source technology into the small and medium-sized business (SMB) community.

By combining these products, SMB customers will be able to build a WAN router using a standard PC platform and begin to save money when compared to solutions built on traditional closed source products. The companies have agreed to make their product lines technically compatible and are committed to making the solution easy to use and deploy. Vyatta and Sangoma also announced they plan to collaborate on marketing activities that promote growth of the open source router and firewall applications among user communities.

According to Sangoma Technologies president and CEO David Mandelstam, "It was natural for us to partner with Vyatta and support their innovative router implementation and open source model with our internal WAN connectivity solutions to achieve a truly disruptive technology in the router market space."

"It's time the data communications marketplace had an open source alternative to the closed source networking vendors," adds Vyatta chairman Allan Leinwand. "Sangoma shares our vision for the router market. Together, Vyatta's OFR and Sangoma's WANPIPE products allow small and medium businesses to realize these benefits in offices around the world."

In another recent announcement, Sangoma named a channel partner. Headquartered in Petach Tikva, Israel, Tikal Networks will serve as a distributor of Sangoma's products in Israel and the Middle East.

"By partnering with Tikal Networks, we now have the opportunity to extend our presence in Israel and other surrounding countries with a partner who understands the market and is well known to the local Soft PBX community," says Sangoma Technologies president and CEO David Mandelstam. "Israel is a center for telecommunications and is the base for more high tech startups than any other country in the world except the United States."

Tikal Networks has proven experience in setting up VoIP networks, with installations in Israel and abroad.

"As thoroughly skilled players in the VoIP space, we are looking forward to helping IT professionals understand the value and benefits of Sangoma's products in optimizing WAN and voice network infrastructures," said Tikal Network's chief operating officer Gil Farkash. "We will leverage our strategic relationships to strengthen Sangoma's market position in this region."

Tikal Networks has already implemented Sangoma hardware into its Crystal Clear Open Source software based on Asterisk. Crystal Clear is a complete PBX in software that runs on Linux and provides many telecommunications features including VoIP. IT

Greg Galitzine is the editorial director of Internet Telephony.

WiFi Telephony: Back to the Future

It's time to take a look at how far we've come in the relatively young life of WiFi telephony and assess the progress made in terms of the technology and market adoption. Much of the development has been influenced by standards (or lack thereof), along with the typical growing pains associated with deploying any new technology. We've experienced hot enterprise markets, such as healthcare and retail, and we're now seeing the emergence of WiFi telephony applications targeted at the consumer. As we look toward the future, it's clear that both IP telephony and WiFi are here to stay. The synergies between them will have a significant impact on communications both inside and outside the workplace.

Looking back on the history of WiFi can tell us a lot about the future. With the ratification of the initial 802.11 wireless LAN standard in 1997, enterprise wireless data technology went from proprietary implementations serving niche applications, to widespread implementations by leading data network vendors. About the same time standards-based, interoperable wireless LANs hit the market, we began seeing viable and practical implementations of enterprise IP telephony. The initial challenges of running voice applications over wired IP networks — performance, security, and QoS — were amplified when dealing with wireless networks. And while most enterprise IP telephony solutions carried the PBX tradition of keeping telephone sets and protocols proprietary, interoperability and standards compliance were baseline requirements for wireless networks and client devices.

The initial driver for enterprise WiFi telephony was employee mobility. Most of the early deployments were in healthcare, big-box retail, and manufacturing applications. Return on investment was based on providing critical employees, such as nurses, department heads, and technicians, with telephone access while on the move. Although other wireless technologies, like pagers, walkie-talkies, and cellular phones, were viewed as alternatives, none offered the telephone system functionality and network reliability provided by WiFi telephony.

In many situations the business case for deploying a WiFi network was driven primarily by the requirement for mobile telephony, with mobile data support as a secondary benefit. Alternatively, some enterprises made the case for deploying WiFi based on data application requirements, and then leveraged the same infrastructure for voice. For example, in the hospitality market, many hotels deploy WiFi to provide guests with broadband Internet access for WiFi-enabled laptops. With the WiFi network in place, adding WiFi telephones for the hotel management and staff means lower cost, higher functionality wireless communications, enabling greater response times and higher productivity.

Standards to the Rescue

While concerns about WiFi security and voice/data convergence still linger, the clear trend is toward WiFi. WiFi infrastructure has come a long way in the last five years, with much of the innovation focused on providing higher quality voice applications. Security has been addressed with the IEEE 802.11i standard and the WiFi Alliance WPA and WPA2 specifications, which have all undergone extensive industry scrutiny to verify their suitability for enterprise applications. There are still some security improvements in the works, specifically the 802.11r standard for improving roaming without making compromises on security. Other security require-

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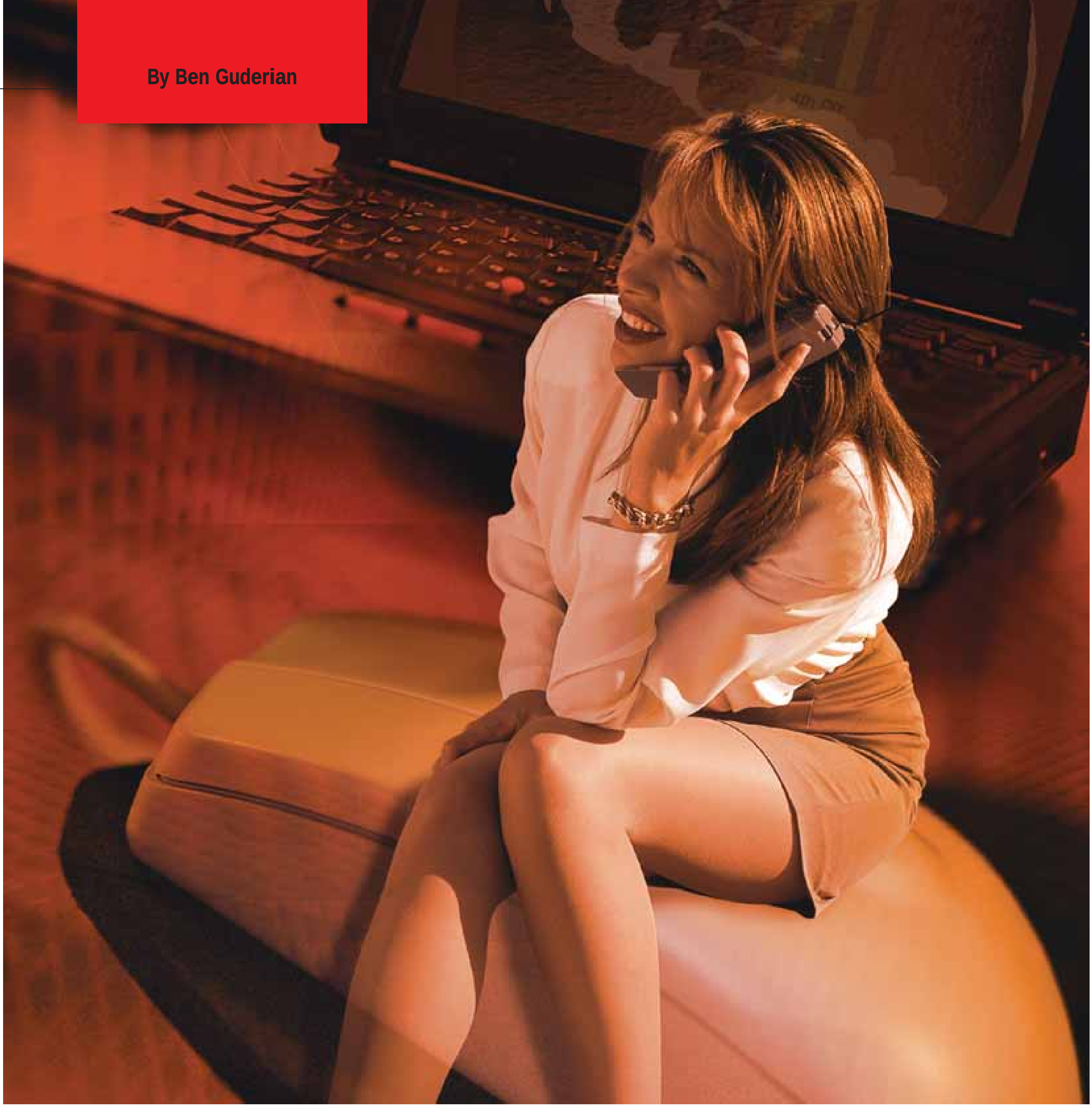
By Ben Guderian



ments have been met through implementing virtual private network (VPN) solutions.

From the very beginning, quality of service (QoS) for WiFi telephony has been a critical need. QoS is important for any IP telephony implementation, but even more critical with WiFi telephony because of the limited bandwidth. Shortly after the initial standard was ratified, the IEEE commissioned the 802.11e task group to develop a standard. Early entrants into the WiFi telephony market had to develop stopgap QoS mechanisms. Fortunately these solutions allowed the market to develop and validated the necessity for enterprise-grade QoS. The 802.11e standard was ratified in 2005 and the WiFi Alliance is now rolling out its WMM specifications for QoS components in enterprise WiFi telephony.

The most tangible and publicized WiFi innovations have been the evolving standards for higher data rates. The initial 802.11 standard had a maximum data rate of 2 Mb/s. Then came 802.11b, speeding things up to 11 Mb/s, shortly followed by 802.11a and 802.11g running up to 54 Mb/s. Now we're anticipating the 802.11n standard with potential for rates in excess of 300 Mb/s.

The missing link on the standards front has been the lack of enterprise VoIP protocol standardization.

Most WiFi telephony devices still use 802.11b and haven't kept up with the race for higher speeds for a couple of reasons. First, most enterprise WiFi implementations use 802.11b as a lowest common denominator, which means they may support 802.11g and/or 802.11a speeds, but they still need to provide backward compatibility for older client devices using 802.11b. The second reason is that voice applications don't benefit from faster data rates the way data applications do. Higher data rates translate into higher user density for voice applications, but the vast majority of enterprise applications can be served today by 802.11b. Over time, 802.11b will be phased out as WiFi devices migrate to support 802.11a and 802.11g and, eventually, 802.11n, depending on its adoption in enterprise applications.

The missing link on the standards front has been the lack of enterprise VoIP protocol standardization. While the WiFi industry has invested significant time and energy into standards and specifications to ensure interoperability and multi-vendor solutions, the enterprise IP telephony market has remained predominantly proprietary and closed. Even though it's clear that SIP has prevailed as the open standard of choice for VoIP, most enterprise IP telephony solutions continue to utilize proprietary protocols as a means to deliver advanced features and capabilities.

The Device Landscape

More devices are coming to market to support WiFi telephony applications, and in this area we see divergence between consumer-focused and enterprise-focused products and services. On the consumer side, we've seen WiFi handsets offered for accessing VoIP services such as Vonage and Skype, with some even bundled by VoIP service providers. These handsets are designed for use on home or hotspot WiFi networks, so they don't typically support enterprise-grade security, QoS, and roaming capabilities. Another category of devices are handhelds which incorporate both cellular and WiFi radios in a single device. These dual mode devices let the user take advantage of high-speed connectivity primarily for Web surfing and e-mail synchronization using WiFi networks, but some are also capable of running softphone applications to support telephony over WiFi networks.

Dual mode devices offer the advantage of having a single handset for ubiquitous use. Consumer-oriented dual mode services, such as British Telecom's Fusion, give customers a single handset to use as a cordless phone over WiFi at home, and a GSM cellular phone away from home. Similar applications are targeted at enterprise customers, although they carry the additional challenge of integrating with an enterprise PBX to deliver business telephone features in the office. Offering the ability to hand off calls between the WiFi and cellular networks is seen as a key benefit for dual mode applications, and various carrier-based and enterprise-based solutions are available today or coming soon.

However, dual mode handsets aren't for everyone. Just as laptop PCs haven't completely displaced desktop PCs, there is still a large number of employees that only need work-related telephone access at work. Employees who don't travel or need to be accessible during off-hours can be given WiFi devices that stay in the workplace and avoid the additional cost and management issues associated with cellular services. On the other hand, employees that are furnished with cellular phones for off-site calls are the most likely targets for dual mode handsets.

Looking Ahead

As we look to the future of WiFi telephony, there are some easy predictions to make. Penetration of enterprise WiFi technology will continue to grow based on lower cost of deployment, user demand, and new mobile applications. The latest generation enterprise WiFi solutions offer the performance to address most users' network access needs, plus administration and management capabilities that significantly lower the total cost of ownership. Meanwhile the cost and complexity of installing home WiFi networks have also dropped, giving many the firsthand experience of wireless connectivity in their homes. WiFi-enabled devices, such as PDAs and dual mode handsets, will help drive enterprise demand for WiFi access in the workplace, and users will want to have access to the business telephone system with these devices.

It is also evident that enterprise IP telephony is here to stay, as deployments of IP desksets have now surpassed traditional TDM (define - news - alert) and analog sets. IP telephony enables unified communication capabilities with other IP-based communication applications, such as e-mail and instant messaging. Delivering these integrated applications to handheld devices will further increase the value of WiFi telephony by making employees even more accessible.

WiFi telephony has come a long way in a short time thanks to rapid innovations in wireless networking and VoIP and it's making a positive business impact in many deployments today. As enterprises plan their IT strategies around IP networks, unified communications and wireless, it is critical they consider WiFi telephony to make employees more mobile, responsive and productive. IT

Ben Guderian is vice president of Market Strategy and Industry Relations at SpectraLink. (news - alert) For more information, please visit the company online at http://www.spectralink.com.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Real-Time Threat Mitigation for Service Providers

With a host of new challenges and vulnerabilities, the emerging VoIP (define - news - alert) network clearly requires a more sophisticated approach to security than those currently used to secure data networks. For service providers to successfully deploy VoIP in the mainstream market, it is important to understand that, while some VoIP security requirements are similar to those in data networks, several areas are specific to VoIP.

As a real-time service, VoIP requires infrastructure that provides automated, real-time response to security threats to preserve very high availability service. With availability requirements having been held to a standard of 99.999 percent reliability — which allows for less than five minutes of downtime per year — VoIP simply cannot tolerate any security techniques that introduce delays or impact availability.

Additionally, VoIP services can offer features such as caller ID, call forward, voice mail, and three-way calling, which open up service providers to a number of new service threats not seen in the data networking domain, such as toll fraud, service theft, voice spam (SPIT), and identity theft.

Another aspect of large-scale VoIP deployments is so called “monoculture” — all the VoIP endpoint devices, residential gateways, and access devices are the same from the hardware and software point of view making security attacks performed in the “broadcast” mode very dangerous, since they can disable the entire network by exploiting a single vulnerability present on particular types of devices or applications.

VoIP security solutions designed around network-based devices and signature-based applications are not able to address the real-time nature and complexity of VoIP networks. Only by combining network and host-based security devices and applications with sophisticated, systems-level threat mitigation platforms, can operators efficiently protect the entire VoIP infrastructure.

A Proactive Approach to VoIP Security

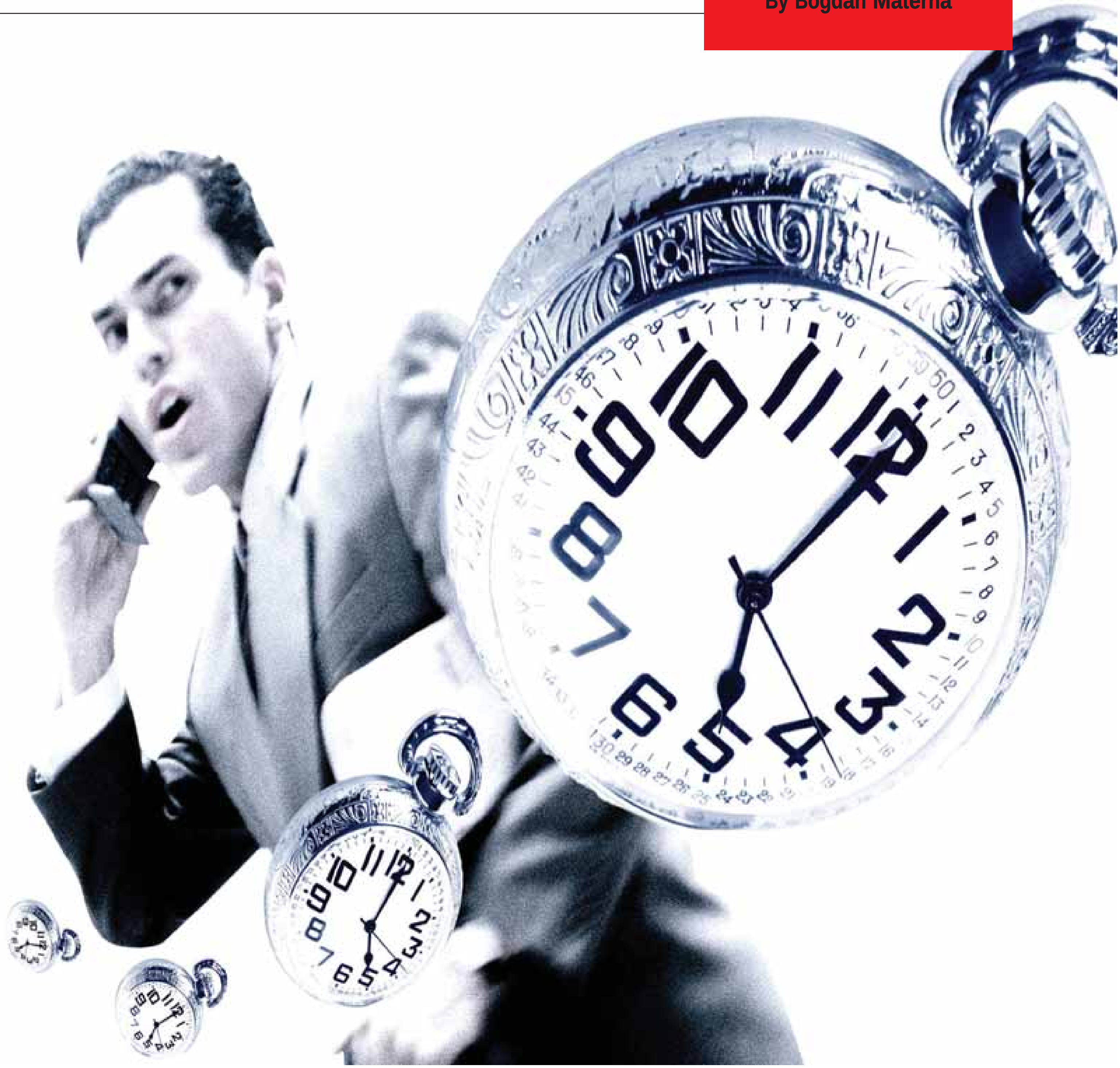
One of the greatest lessons learned from the data security world over the past decade is the importance of a proactive approach to security. The data security world evolved from a reactive to a proactive approach as new threats emerged, until finally, the risk of leaving networks vulnerable became all too clear. The very nature of telecommunications networks, and VoIP networks, for that matter, dictates that VoIP security doesn’t have the luxury of only taking a reactive approach until different types of attacks reach critical mass.

As service providers make the transition to VoIP, they need to take a proactive approach to security from day one, while ensuring they can secure all points in the VoIP network, whether it be local, national, or international. Building a proactive, systems-level approach to VoIP security must consist of three functional components: prevention, protection, and mitigation.

Prevention

Prevention enables organizations to proactively identify and fix VoIP-specific vulnerabilities before they impact end users. A commonly used approach from the data security world — vulnerability assessment (VA) — is particularly effective as a proactive strategy. By performing a VoIP VA in the lab, before any VoIP equipment or applications are deployed, organizations are able to verify vendor claims and identify security flaws early in the deployment cycle. Once VoIP is deployed, periodic or continuous vulnerability assessments should

By Bogdan Materna



become the base of an overall proactive VoIP security strategy. Once vulnerabilities are identified, they should be addressed by appropriate actions, such as patching, re-configuration, and network tuning.

Protection

Within the VoIP network, various security architectures and solutions should also be deployed to protect VoIP services from threats. Any security architectures and solutions must be "VoIP aware" so they do not impact VoIP service quality and reliability. Deploying a multi-layer infrastructure that provides both perimeter and internal network protection is ideal. In most cases, this will consist of numerous security devices and host-based applications, such as SBCs, VoIP Network Intrusion Prevention Systems (NIPS), VoIP DoS defenses, VoIP Network Intrusion Detection Systems (IDS), Host IPSs, Authentication, Authorization and Accounting (AAA) servers, encryption engines, and VoIP anti-virus software.

Mitigation

However, it is widely accepted that no matter how good the prevention and/or protection in place may be, eventually, an attacker or worm will successfully penetrate the defense architecture and impact VoIP infrastructure.

No matter how good the prevention and/or protection in place may be, eventually, an attacker or worm will successfully penetrate the defense architecture.

To date, there have been few publicized VoIP security attacks, but there is already an example of a financial institution that was subjected to a worm attack. In this case, the VoIP infrastructure was disabled and the company experienced loss of voice communication that translated into financial losses. However, as VoIP is adopted into the mainstream, it is a matter of when, not if, widespread attacks will occur. When that occurs, threat mitigation systems will take over, and must be able to respond autonomously to a detected security attack, while keeping their impact to a level where VoIP services can still function at a reasonable level of QoS and give the support staff enough time to restore normal services. It is important to distinguish between Intrusion Protection Systems (IPS) that will prevent an attack and, therefore, belongs to the protection domain from a Threat Mitigation System (TMS), the main task of which is to minimize the impact of the attack when it is already under way. TMS is the key for implementation of carrier grade VoIP services where even few minutes of downtime can have serious financial and legal implications.

An Effective Strategy for VoIP Threat Mitigation

Currently, a combination of human intervention and security management tools are being used to mitigate the impact of VoIP attacks. Take, for example, a "zero day" worm; this is a worm that is created almost at the same time as the vulnerability it exploits is discovered, hence the term "zero day." In other words, because the time between the vulnerability identification and an exploit could be measured in hours or minutes vs. days, it may pass through the protection infrastructure and cause the network and/or critical VoIP servers to go down, which may result in minutes, hours, or days of downtime as the issue is addressed by IT staff.

As the VoIP market matures and attacks become more prevalent, these methods will no longer be sufficient. VoIP networks — and VoIP consumers — simply cannot tolerate multi-hour or multi-day downtimes; they must have 99.999 percent availability.

Service providers will require a real-time, automated response to VoIP security threats. Otherwise, major VoIP security threats, such as SPIT, DoS, or fast spreading worms, may result in service disruption or service integrity degradation. While the need to secure IP-based voice communications is driving the need for threat mitigation, as new IP services such as IPTV become available, the same security requirements will apply. Just as users will not tolerate an outage of phone service, they will demand the same level of reliability and integrity for other paid services, such as IPTV.

The most effective approach to VoIP threat mitigation involves three core elements:

- Detection: A threat must be identified as soon as possible and needs to be a combination of signature-based and anomaly-based detection techniques — for example, to address zero-day exploits before the signature is created. Once the threat is recognized, the signature-based portion of the detector could be updated to detect future occurrences of this threat.
- Correlation: Once the threat is detected, it must then be correlated to known information on the device(s), such as known vulnerabilities, topology, and any additional information collected from the security infrastructure, such as IPS, firewall, SBC, etc.
- Response: The system must then respond in real time to ensure that the reliability or integrity of the network isn't impacted. Any response to attack must take place in the background, so it is seamless to end users and the service remains available.

This model of threat mitigation is delivered as a software solution. Because the software works at the systems level, it can deliver an end-to-end, layered view of VoIP networks, addressing the entire VoIP network including the OS, protocols, etc. Unlike human intervention, the goal with a software approach is to respond to an attack in a matter of seconds, and to be in-line and host-based to support real-time response and mobile users.

Conclusion

At the current time, VoIP threat mitigation systems are not yet available. However, they are the natural next steps for securing VoIP and they need to be actively planned for. As some equipment providers are beginning to talk about "self-defending" VoIP systems, which include elements of threat mitigation and the demand for IP-based services increases, threat mitigation will become a reality. In planning and deploying VoIP, a proactive, systems-level approach to security is required and threat mitigation should figure prominently in service providers' VoIP plans.

Service providers need to push for VoIP-specific threat mitigation to ensure their delivery of PSTN-level reliability and enable them to manage risks to the network. For VoIP to become a reality, all IP services need to be delivered with PSTN-level reliability and security, with industry-wide standards. Vendors must play an active role in pushing security issues such as threat mitigation to the forefront. IT

Bogdan Materna is the VP of Engineering and CTO at VoIPshield Systems. (news - alert) For more information, please visit the company online at http://www.voipshield.com.

Practice Makes Perfect:

Best Practices for VoIP Readiness Assessment

IP telephony resellers are doing it. VoIP service providers are doing it. Enterprise IT organizations are beginning to do it. And, increasingly, IP PBX equipment manufacturers are requiring it. We are talking about performing a VoIP readiness assessment — a relatively simple network planning process that has become a revenue generator for VoIP suppliers and a cost saver for their customers.

VoIP assessments take the guesswork out of preparing networks for voice/data convergence and help to assure the quality of VoIP (define - news - alert) services delivered to end users. The best practices for a VoIP readiness assessment are easy to follow and often produce revealing results that save time, money, and aggravation when deploying a VoIP service. However, in spite of its many benefits, the VoIP readiness assessment has yet to become standard practice for every VoIP project. Why? It's a puzzling question that is difficult to answer. Let's walk through a classic VoIP assessment process and see if you don't agree... it's an ounce of prevention that is worth more than a pound of cure.

VoIP Assessment Defined

First, let's review the elements of a typical VoIP assessment and define some terms. A VoIP assessment is an evaluation of the connectivity, capacity, and configuration of a production network that is destined for voice and data convergence. We'll assume that the goal of this convergence is to eliminate the cost and complexity of operating a separate TDM voice network by creating an IP-based communications system that mixes voice and data traffic over shared network facilities. As a result, an interesting chicken and egg problem emerges. How can you predict the performance of the new VoIP service before the IP PBX is installed and the flood gates are opened? And how can you avoid introducing new performance problems for existing mission-critical data applications when prioritized voice traffic is added to the network?

To resolve the dilemma, specialized VoIP assessment tools are used to simulate voice traffic before IP telephony systems are installed and services are deployed. A common approach used by assessment systems to simulate VoIP traffic is to deploy software agents at various network endpoints to generate synthetic calls. These agents are used to measure the quality of the VoIP service while the traffic load is progressively increased. Traffic agents can be very intelligent, performing a variety of end-to-end troubleshooting tests during the assessment process. Sophisticated VoIP diagnostic tests are designed to ferret out problems before they can manifest themselves as service degradation and user complaints later, after the VoIP service has been deployed.

Detecting and correcting problems in a simulated VoIP environment as opposed to a production environment has obvious benefits. VoIP assessment traffic agents typically operate under the control of a centralized system that determines the frequency and duration of VoIP simulations and aggregates their measurement and diagnostic data onto a dashboard display or a database for data mining and reporting. There is a long list of features and functions available in VoIP assessment systems. The ability to replicate VoIP calls using a variety of voice compression codecs and signaling standards is one such important feature. Automated tests for diagnosing network pathways during the assessment to detect

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By Dave Zwicker

QoS problems or identify excessive packet impairments (delay, loss, and jitter) are also important features. If the assessment tool also has the ability to generate and diagnose data traffic, you can model a number of different scenarios to understand the impact of voice and data traffic peaks during busy hours.

Once you have acquired the right assessment tool for the job, there are six classic steps you will want to follow to perform a comprehensive network readiness assessment to insure success for your VoIP project in every way — from its quality of service to its cost effectiveness.

Step 1: Discover the network inventory

The first step in the process is to document the network devices (routers, switches, and traffic agents) that are participating in the assessment. You will want a record of equipment model numbers, software revision levels, and network addresses so you can understand the structure of the network and its components as they are organized into subnets to provide a context for your assessment results. If you conduct multiple assessments for many remote sites or external customers, creating a record of the network inventory in service at the time of assessment is essential. Before you run your simulations, make sure the infrastructure that is under evaluation is well documented.

Step 2: Verify the network configuration

The next step to be taken before you begin to generate traffic and take measurements is to identify any configuration problems that can skew your assessment results. There are some common network problems that impact VoIP calls differently from traditional data applications. One of these is the duplex mismatch, which is a problem that occurs when there are two consecutive devices along the network path that are configured to work in full-duplex and half-duplex modes respectively. This configuration will create problems for one direction of a VoIP call while having little or no effect on the other. Another common problem is a misconfiguration of QoS settings in network devices and their failure to properly prioritize voice packet delivery over data. There is an automated end-to-end diagnostic test in VoIP assessment systems that detects duplex mismatches and other tests that verify proper QoS operation. Be sure to correct these problems before you run your VoIP traffic simulations so that call quality measurements are based on a clean network environment.

Step 3: Determine the call quantity/quality threshold

Now you are ready to start generating some voice traffic in order to determine the maximum number of VoIP calls conveyed over the network that will meet a predetermined call quality standard. Your assessment system should be configured to generate synthetic VoIP calls using an appropriate compression codec (e.g., G.711, G.723.1, G726, G.729a, and G.729b) and signaling protocol (e.g., SIP or H.323). As the system gradually increases the number of simulated calls over time, it will measure the change MOS values. Recall that MOS is the five-level industry standard for call quality metrics. A MOS score of 4.0 or above is generally considered to be toll grade in the VoIP world. You can set any minimum standard (MOS value) for call quality and use it to gauge the quality of your future VoIP service when performing this preliminary assessment.

Step 4: Make necessary network adjustments

Very often a preliminary assessment will produce unexpected results. This is a beneficial aspect of the VoIP assessment process because you are detecting problems in the network that might otherwise go undetected. If capacity planning was a simple paper and pencil exercise, then life as a network planner would be much easier. In reality, networks are complex and dynamic. So, if the initial VoIP assessment indicates a lower call volume than expected, then it's time to troubleshoot the links and make the necessary network modifications. VoIP diagnostic tests are available to analyze the end-to-end route quality of a call path with poor results. A route quality test drills into every switch or router in the path to examine health and performance information. Excessive packet loss, delay, or jitter can be traced to individual switch/router metrics, such as the CPU, memory, or port utilization percentages, to quickly identify segments that are under-powered relative to the simulated load. There may also be configuration errors or intermittent equipment failures that are contributing factors. After the root cause is isolated and the necessary changes are made, repeat Step 3 and re-run the preliminary assessment until a satisfactory call volume is reached. Performing Steps 3 and 4 is often an iterative process that tunes the network for delivering maximum performance.

Step 5: Simulate a complete business cycle

After determining the maximum call volume and completing all network modifications, you are ready to perform a final comprehensive assessment. This step will serve as a dress rehearsal for your future VoIP service as it generates the maximum call volume over the pro-

VoIP Testing Top 10

By Robert Kinder

Voice over IP (VoIP) is an application deployed over converged networks where service impairments can be difficult to identify and resolve. Lessons learned from current VoIP deployments show that a gap in the service assurance process can result in excessive expenditure of time and resources involving many technical disciplines to resolve service issues. With the rapid growth of VoIP deployments, a resource-intensive approach can quickly overwhelm an organization. Implementing the following items will result in a scalable and cost-effective program for VoIP test and service assurance.

1. *Key Performance Indicators (KPIs)* — The VoIP product/service performance requirements should be mapped to network Key Performance Indicators. The KPIs will aid in identifying relevant network measurements and performance alarm thresholds. Common KPIs include time to dial tone, post-dial delay, call success rate, dropped calls, Mean Opinion Score (MOS), packet loss, latency, jitter, and echo.

2. *Pre-deployment Network Qualification* — Exercise the network in advance of service rollout with a combination of physical loop tests, call generators, probes, and analyzers to determine if the network can reliably deliver VoIP service with the required Quality of Service. DSL testing involves copper loop pre-qualification to detect load coils, bridged taps, wet sections, grounds, etc. Coaxial cable testing will ensure that the forward and return paths meet DOCSIS requirements. Automated call generators, simulating subscriber calling patterns, will indicate if the network is correctly configured (softswitch, dial plans, routes, policies, firewalls, PSTN gateways) and has bandwidth capacity for VoIP service.

3. *Installation Testing* — A large percentage of VoIP service impairments are caused by inside wiring, customer handsets, or customer premises equipment configuration. For professional installs by company technicians, service at the residential gateway RJ-11 port should be tested to validate dial tone with calls to a remote analyzer or handheld test set. Phone jacks in every room should be tested to fix inside wiring problems. Customer self-install business models can be validated with automated loopback testing triggered when the residential gateway registers with the network. The self-install manual might also instruct the customer to place a call to an automated response test system.

4. *Loopback Testing* — Automated testing of VoIP services through scheduled loopback test calls to the residential gateway is a proactive means for detecting signaling and media performance impairments. This has the advantage of finding faults before the customer goes off-hook and notices a problem. One strategy that balances test coverage without overloading the network is to test a small percentage of the subscriber population chosen at random every 24 hours. Additionally, some residential gateways allow loopback testing concurrent with a normal subscriber call; the loopback call is treated as a third session on the same line. This feature offers the possibility of interactive loopback testing by a customer service representative while speaking with the subscriber to troubleshoot a complaint.

5. *Signaling Analysis* — A signal analyzer provides call trace and protocol analysis to troubleshoot call setup and teardown issues for VoIP and PSTN calls. Historical call records enable investigation of customer complaints after-the-fact by Tier 2 or 3 support personnel. Newer initiatives seek to combine

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call flow traces with VoIP bearer channel (RTP media) Quality of Service measurements for a complete view of the customer experience; because signaling messages alone do not indicate speech quality. Call quality metrics can be collected from RTCP XR (RFC 3611) data in a softswitch Call Detail Record (CDR) database, passive monitoring probes, and active test systems. The greatest network coverage is found in the CDR database.

6. *Trouble Signatures and Corrective Procedures* — Troubleshooting a VoIP call can be a complex undertaking involving experts from many disciplines to solve the puzzle, even when the impairment is readily apparent and persistent. The development of trouble signatures, probable causes, diagnostics, and corrective procedures is needed for operational effectiveness. The result is a comprehensive troubleshooting guide for the organization and system requirements for service assurance and test systems.

7. *Event Correlation* — A major challenge is to integrate the telemetry from numerous network elements in the VoIP service delivery path — CPE, access, edge, and core networks, PSTN interface, service platforms, and test systems — into a meaningful view of service-affecting conditions to determine root cause. A layered architecture is necessary to filter information from the network elements and test systems before forwarding to higher level manager-of-managers for expert system identification of trouble signatures and root cause.

8. *Automation* — VoIP networks are highly dynamic due to rapid subscriber growth and the inherent flexibility of IP networks. Bottlenecks do occur and the network can be quickly overwhelmed with systemic failures, resulting in an avalanche effect when a capacity threshold is crossed. A high degree of automation is needed to identify trouble signatures, enact corrective measures and be proactive with customer notifications to reduce the volume of trouble reports. Test and monitoring systems play a key role by providing benchmark references for early detection of trends, threshold crossings, and alarm notifications.

9. *Training* — Training touches all VoIP service provider staff, from field technicians to network operations to customer service representatives. VoIP technologies require updating of knowledge and skills as network intelligence is increasingly pushed out to the residential gateways (e.g., per-subscriber VLANs). Training programs for field technicians and call center personnel, in conjunction with the tools recommended in item #6, can help optimize their ability to resolve VoIP issues. Call center personnel can be especially effective in the early detection of unusual conditions.

10. *Configuration Change Management* — Rigorous change management processes should be followed for the carefully controlled introduction of approved updates to the network. What worked for managing high-speed data (Internet) services can have unforeseen consequences for VoIP service. Lab networks do not fully duplicate the environmental conditions of large-scale production networks, making it difficult to fully validate a change before deployment. Risk can be mitigated by testing new configurations against in-network VoIP test probes that emulate subscribers making and receiving calls, as a confidence check. IT

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duction network during a complete business cycle. A business cycle can vary from company to company, but is typically three to five days in duration with eight hours of activity per day. The goal of this comprehensive assessment is to duplicate minimum and maximum utilization levels to ensure that consistent service quality can be delivered during the periods of peak demand, whenever that may be, throughout the course of a business day or week.

Step 6: Determine service level benchmarks

At the conclusion of your VoIP assessment, you will have made critical changes to the network, documented important facts about the network environment and gathered vital information about service levels for your VoIP project. This information provides a record of the assessment process and a deliverable for customers and business constituents. Equally important are the benchmarks that have now been established for monitoring the VoIP service going forward. Service level benchmarks can provide a reference point used for building service level agreements (SLA). They can also become the performance thresholds used for detecting service degradation when the VoIP service is fully operational.

These six basic steps represent the best practices for performing a VoIP assessment. It's a "measure twice and cut once" approach for outfitting your organization with integrated voice and data services. Remember, it's much easier to make changes to the network BEFORE your VoIP service is operational. Take it for granted that your end users will expect, if not demand, the quality and availability of a traditional telephone service. Follow these VoIP assessment best practices and you will be on track to meet those expectations while realizing the many business benefits of VoIP. IT

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Culture Clash: IMS vs. Peer to Peer for VoIP

After many years of promise, Voice over IP is now seeing strong growth in all sectors of the voice communications industry. Upstarts, incumbents, cable providers, and other telephony companies are all embracing VoIP. However, there are two very different ideas about how VoIP and its services will evolve. In the IP Multimedia Subsystem (IMS), all-IP services are built using a framework that borrows from both the PSTN and IP worlds. By contrast, there is a growing peer-to-peer (P2P) VoIP movement that sees voice as just another IP application, with most of the intelligence provided at the endpoints. This article will compare the two models and consider the resulting impact on the ecosystems and go-to-market strategies for equipment vendors, carriers, and customers.

Back to the Future: IMS

During the last 20 years, it has become apparent to most telephone companies that the path to improved profitability lies not just in providing voice phone calls to consumers, but also various other services. One of the key rationales for the Intelligent Network (IN) was to provide a framework that would enable new services to be rapidly developed and open up such service development to a wide array of developers. The IN uses circuit-switched technology and has had some successes, but never really unleashed the large array of profitable services its creators had envisioned. In addition, even as IN and its embedded SS7 signaling were getting off the ground, the success of applications such as the World Wide Web and e-mail over IP networks caused network architects to reconsider how they should be building future communications networks.

There were lots of arguments in the telephony standards community about which way to go, but eventually, a consensus began to emerge in the wireless community, which was attempting to figure out how to build its next generation (3G) networks. This consensus centered on a few basic ideas:

1. The network should move from circuit-switched to IP technology;
2. The focus of the next generation architecture should be on enabling a rich variety of multimedia services; and
3. The primary protocol that would enable these services should be SIP.

With these directions in mind, the 3GPP (Third Generation Partnership Project) developed a series of specifications that collectively became known as the IP Multimedia Subsystem (IMS). In addition, a related consortium (3GPP2) that has focused on building architectures for CDMA wireless systems, also endorsed the ideas of the IMS.

One of the useful ideas of IMS is its layered architecture (Figure 1).

There are three key layers within IMS, dealing with Service, Control, and Connectivity. A key part of the vision is that the core of the architecture is built solely on IP protocols. However, since there is a need to connect to other networks as well, the Connectivity layer was developed in order to enable users on other networks to communicate with the users of an IMS-enabled network and take advantage of common services.

IMS was developed by the wireless community, which already had an existing second generation network architecture based on the use of SS7 and other PSTN technologies. Hence, a connectivity layer was a must for the third generation wireless network in order to be able to communicate with users of the existing wireless and landline networks. However, other network flavors, such as

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By James Rafferty

WiFi, MPLS, and WiMAX, were emerging, so the connectivity layer concept was extended to support virtually any network fabric to which a 3G network might connect.

Once the problem of connecting to a variety of networks was resolved via the connectivity layer, IMS became interesting to service providers outside of the original wireless target market. In particular, landline providers were also looking for a blueprint as they began to move their networks to VoIP. (define - news - alert) Hence, IMS emerged as the candidate service architecture for both wireless and wireline networks.

A key element of IMS is that it has adopted SIP as its primary protocol. This enables IMS to build upon the already strong momentum of SIP in the application provider community. In fact, IMS encourages further decomposition of services into a combination of software and hardware components. For example, an application server vendor may develop a prepaid calling card application using SIP and then communicate with a Media Server that also uses SIP in order to handle media operations such as collection of digits and verification of pin numbers. With IMS, the application vendor no longer needs to worry about generalized service issues, such as authorizing the use of the network and the generation of related accounting records. All of this can be

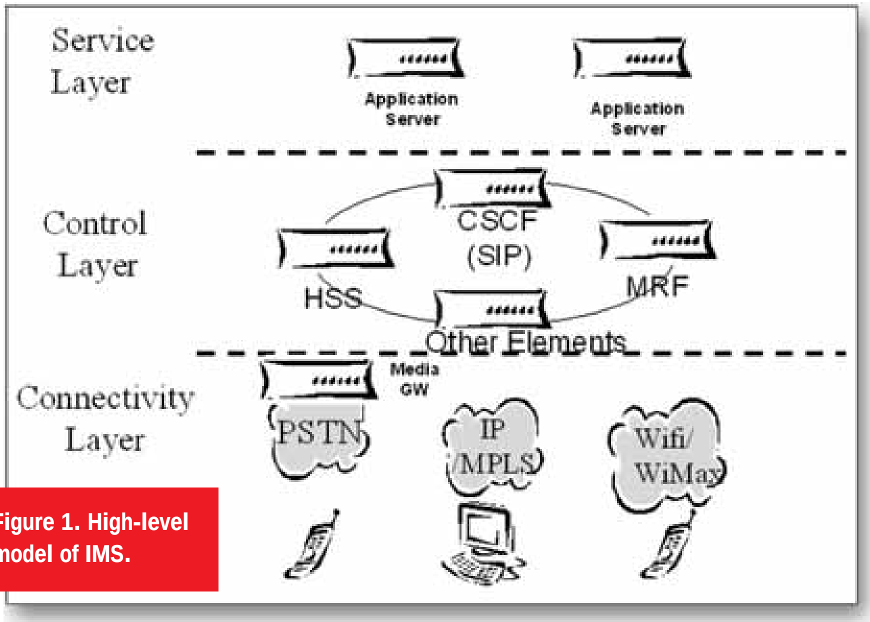

Figure 1. High-level model of IMS.

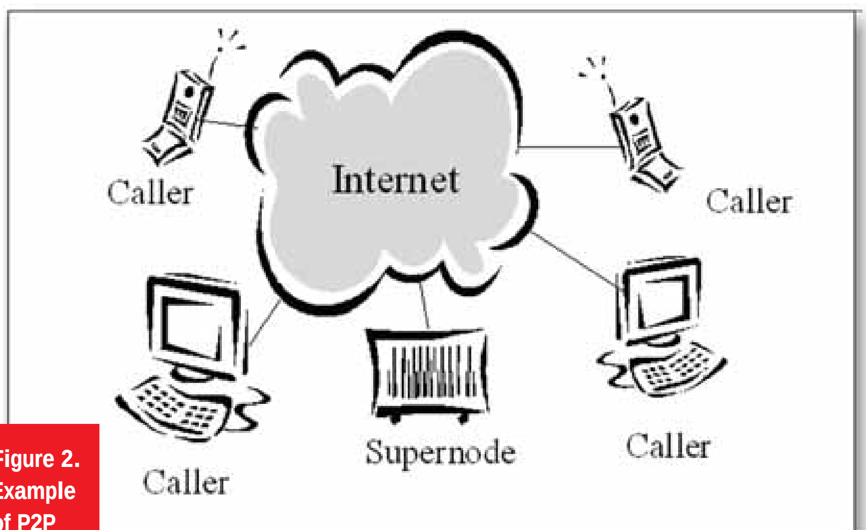

Figure 2. Example of P2P network for VoIP.

done using standard elements of the IMS control layer. In addition, existing SIP applications, such as prepaid and unified messaging, can be ported onto the IMS architecture.

For all of these reasons, IMS is quite popular among incumbent vendors in the wireless and wireline communities, and has also gotten the attention of more specialized providers, such as cable MSOs, that offer voice. In addition, IMS is not just about voice, but also is intended to enable a new generation of media-rich services that support other media, such as text, video, and instant messaging. IMS has gotten quite a bit of mind share among providers of hardware and application software. These vendors are rapidly developing their IMS strategies and recruiting a roster of IMS partners. The next few years should be very interesting as both traditional and next generation equipment providers devise their IMS strategies and build alliances with the many companies that are investing in IMS related technology and applications.

Revolution in Process — P2P for VoIP

Even as IMS was being addressed in the standards bodies, another set of companies took a completely different approach to rolling out multimedia communications services over IP. The best known of these companies is Skype. Over the last three years, Skype has created a mostly "free" IP voice service that now has over 100 million registered users.

Skype (news - alert) has used the concept of a P2P network as its starting point, where almost all of the sophistication of the application is built into the endpoints themselves (Figure 2). For example, a user can download the Skype communications application onto a PC and use a headset for speaking and listening. The Skype client also has built-in use of buddy lists and presence concepts, so that a Skype user can check to see if the person he wants to talk with is online before initiating a voice conversation. Skype also borrowed heavily from the original P2P services such as Kazaa and Napster in its marketing approaches, which are viral and rely more on social networking than on traditional advertising channels. On the technical side, these companies had also figured out how to cross the barriers erected by the NAT (Network Address Translation) devices used on most routers. This has been a crucial element in ensuring connectivity among P2P users, who use a wide range of commercial router and IP switching equipment.

Skype is not the only example of a P2P VoIP network, but it is the best known. Other advocates of P2P IP voice include the leading Instant Messaging (IM) companies: AOL, Yahoo!, and Microsoft. All of these companies have added voice capabilities into their IM clients, so users of these networks can communicate via text or voice messaging. One of the controversial aspects of Skype is that it makes use

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of proprietary communications protocols. By contrast, most of their competitors in the P2P space are using SIP as a standard protocol.

There is a standards movement afoot to further enhance SIP to make it more P2P friendly. A mailing list has been set up as a first step in creating a working group in the Internet Engineering Task Force to gain agreement on SIP specifications for enabling P2P network communication. In the meantime, Skype continues to move forward very aggressively with its own approach. In particular, Skype has courted the application developer community by creating an affiliate program and an application programming interface. This has already resulted in numerous software developers creating their own add-on services for Skype users and the development of a new generation of mobile phones that include support for Skype software clients.

Summary

There are two rather different movements emerging to shape the future of IP communications. Many players in the telecom industry have lined up in force behind IMS and are taking steps to develop this architecture over the next several years. There are already many related requests for proposals (RFPs) from operators and the early days of deployment have also begun, albeit sometimes referred to as "pre-IMS." Industry groups, such as the Multi-Service Forum, are hosting interoperability events that are designed to foster cooperation between the companies that are building IMS-compliant products and services. Consumers will not see IMS directly, but will be able to gain the benefit of new services, such as video mail and online mobile gaming, as they roll out using the common IMS infrastructure. For application developers, IMS holds the promise that an application can be developed once and then used on several different networks. In addition, its built-in security features and "blue chip" heritage should enable new services based on IMS to be effectively marketed to enterprises that want to take advantage of a hosted service business model.

By contrast, P2P IP communications is a classic disruptive technology, which has a much different starting point than IMS and is, therefore, attractive to a different business ecosystem. P2P is fertile ground for application developers, particularly if they want to leverage the large network of Skype users and become a Skype affiliate. There will also be opportunities to develop new applications based on P2P SIP, but the market for these is at an earlier stage of development. To date, P2P has been mostly limited to PC-to-PC communications and has been primarily used by consumers rather than businesses. Given its roots in file sharing systems, peer-to-peer is perceived to be more of an "outlaw" technology than IMS, which is likely to hamper adoption of P2P in the enterprise.

In summary, the IMS and P2P have excellent momentum as infrastructures upon which to build new IP communications applications and services. IMS is shaping up as the preferred approach among many telecom industry players for deploying an IP service architecture that will enable voice and other multimedia services such IPTV to run on multiple networks. Peer-to-peer technology for IP communications has shown tremendous growth to date, but has been

Peer-to-peer is perceived to be more of an "outlaw" technology than IMS, which is likely to hamper adoption of P2P in the enterprise.

primarily of interest to PC-based users. As a quickly growing, viral technology, it will continue to challenge the status quo in IP communications. It's likely to be on the forefront as real-time communications evolve and continue to take advantage of the ever increasing power of the CPU chips that are used in computers and mobile communication devices. IT

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Is Voice over IP Too Risky for My Company?

Looking at VoIP Risk from a Mathematical Perspective

Two years ago, Deloitte Consulting completed a research study predicting that by 2006, over 66% of companies would have deployed Voice over IP, a technology that allows phone calls to be placed over the Internet. They listed several of the common reasons for this migration — shared network infrastructure, reduced complexity and costs, technology trends, and added functionality benefits. Myriad organizations have also joined in. Boutique technology consulting firms have published white papers sharing the success stories of companies converting to VoIP. Cisco Systems provides detailed documentation explaining the protocols behind the technology, and even the business case for the technologies.

Yet, today's VoIP (define - news - alert) technology is not as widespread as many have predicted. The executive offices and IT departments of many companies are eager. But they remain on the sidelines, likely because of several high-profile implementation failures. For all of the advertised benefits of VoIP, it has significant downsides. User expectations, technical details, potential business interruptions, and financial payoffs all come into play. It's no wonder why many decision makers remain hesitant.

How to Analyze Risk?

Existing literature does an excellent job outlining and explaining the potential benefits of VoIP. Customers understand the advantages well, but it's the risks that they worry about.

Stable telephone communications is essential to many businesses. Companies could lose thousands, if not millions, in sales and revenues if the phone system went down for several hours. Executives taking on a VoIP project would also be concerned about potential telephone outages causing productivity loss, missed deadlines, and resentful employees and customers.

Even those who manage IP networks every day are hesitant to embrace VoIP systems. One network operations engineer who worked for a California energy company stated his concern this way: "When the computer network goes down, what's the first thing we do? Pick up the phone and begin a troubleshooting conference call. With Voice over IP, the telephone won't work!"

Upon hearing all of these worries, it became clear to me that those considering VoIP projects don't need stronger arguments for cost savings or a better ROI calculator. Instead, we need a quantitative analysis on understanding and managing the risks associated with VoIP.

One of the best quantitative tools for making sound decisions under uncertainty and risk is Decision Analysis, a mathematics-based decision methodology taught in business schools worldwide. In this paper, we will explore the risks and possible outcomes of a VoIP project using a Decision Tree model, analyze the likelihood of each outcome, and draw conclusions on when and how to implement a successful VoIP project.

A Decision Tree Model: Mathematically Analyzing a VoIP Project

The decision tree depicted in Figure 1, despite its daunting appearance, is actually quite simple. It has one decision event, designated by the square, and three uncertain events, designated by the circles. The story reads like this:

By Ryan Tang

#1 square: A company faces a decision on whether or not to implement VoIP
#2 circle: If the company chooses to pursue the project, they face uncertainty, because the implementation may succeed or it may fail.
#3 circle: If they are able to implement VoIP, they face more uncertainty. The new telephone system may be stable or it might be unstable and hurt business.
#4 circle: If the new telephone system is stable, the company may experience overall cost savings (or net benefits via new functionality), no cost savings, or increased cost (or net "loss" due to problems such as poor sound quality).

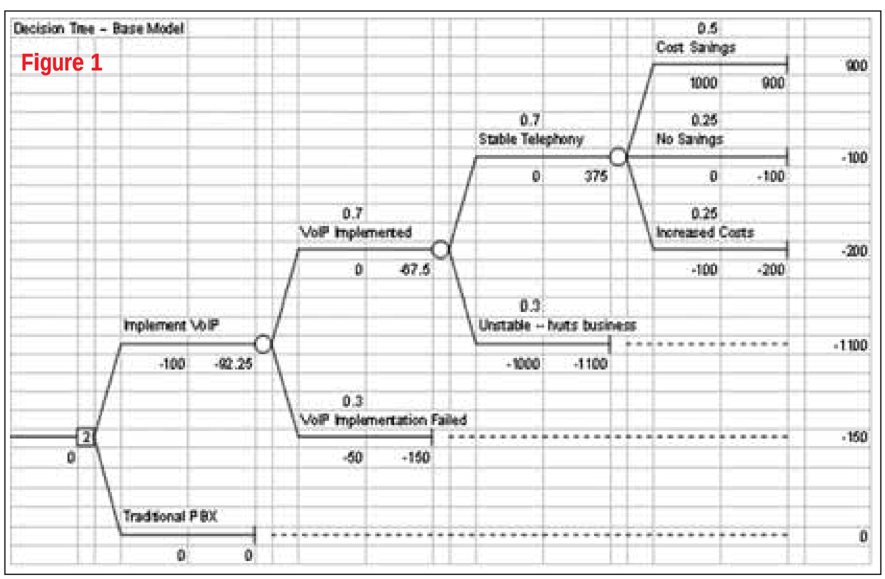


Figure 1

To build this model, we needed to make reasonable "best guess" assumptions:

- The cost of a VoIP project is normalized to 100 monetary units. While this cost may be $100,000 or $5.6 million, we use a project baseline of $100 for simplicity.
- The cost of "doing nothing" and staying with the traditional phone system is set at a baseline of $0.
- A failed VoIP implementation will incur an additional 50% cost — this cost includes reversing the telephone infrastructure changes, "managerial embarrassment," and costs of a damaged reputation.
- An unstable VoIP system will cost 10 times the original project ($1,000 in additional costs). This includes emergency troubleshooting, loss of productivity, loss of sales, and of damaged reputation.
- A stable VoIP system can leave the company "worse off," "about the same," or "better off"
    - ❒ If the company is "worse off", it will incur additional costs equivalent to the original project costs ($100 additional cost, net present value).
    - ❒ If the company is "about the same", the net cost and benefit will be about the same as traditional telephones ($0 more, net present value).
    - ❒ If the company is better off, it will save ten times the original cost of the project ($1,000 savings, net present value).
- This company has a 70% chance of successfully implementing VoIP.
- If the implementation is successful, the company has 70% chance of having VoIP run stably without affecting business.
- If VoIP is implemented and stable, the company has 50% chance of being "better off," 25% chance of being "about the same," and 25% chance of being "worse off."

The Results. What Do They Mean Anyway?

The decision tree recommends this company not to implement VoIP and to stick with the traditional telephone system. On "average," the VoIP project will result in a $92.25 loss and, therefore, the company should stick with the traditional PBX with its project cost baseline of $0.

The result, based upon our assumptions, makes sense. Conventional wisdom says if a company has a solid well-built telephone infrastructure and it depends on telephones heavily for its business, making a big investment and taking on significant risk for a VoIP project is likely a bad idea. Mathematically, the results are consistent. As the CEO of this company, if you are worried about heavy losses and expect a 24.5% chance of success (properly implemented *and* stable *and* better off: 0.7 x 0.7 x 0.5 = 0.245), you certainly wouldn't want to commission a VoIP project.

Keep in mind, however, that this recommendation and "average loss" is dependent on the numbers fed into the model. While the result is pretty good for a stable company with similar costs and a similar perspective on the risk, those with differing assumptions may end up with a different result. If our model was used for an actual company, we would take painstaking measures to ensure accurate input. We would calculate expected cost for the VoIP project, create a financial model to analyze the value of a "better off" telephony scenario and the "worse off" telephony scenario, and determine the realistic cost an unstable telephone system. All of these financial estimates would used as inputs to create a more precise model.

The Problem with Probabilities

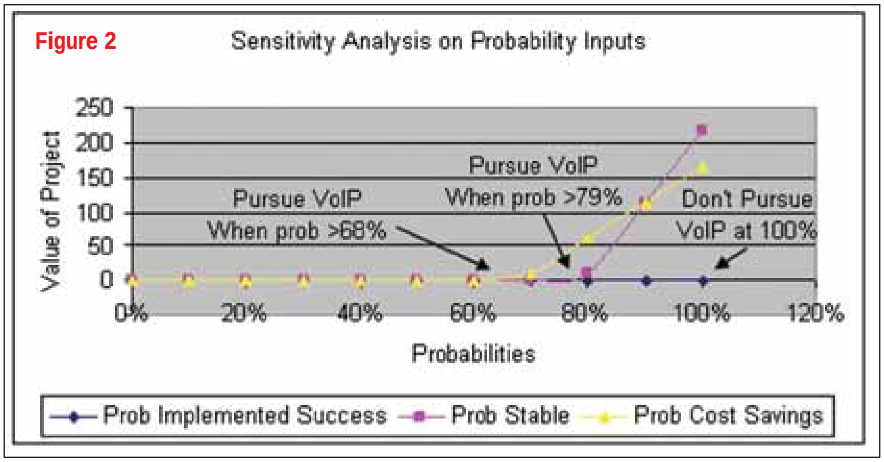
Take another look at the decision tree. Notice that probabilities are an important part of the model. Now the traditional notion of probabilities that we learned in school — the probability of flipping heads is 50%, the probability of getting a black jack in a fair, shuffled deck is 4.8% — doesn't apply here. We can flip a coin a 1,000 times in a controlled condition to calculate the true probability; we can't implement a VoIP project 1,000 times with the same company in the same way to calculate the true probability of success, nor would we want to.

Accurate probability inputs create a unique challenge. Financial estimates can be calculated fairly accurately, but probability estimates require a bit of guesswork. Consider the probability of a successful project. Even if a consultancy states that, historically, 90% of its VoIP projects were successfully implemented, an astute decision maker would make his own estimates. Many other factors, like current employee morale, the complexity of the current telephone infrastructure, the capability of the current IT support team, and relationships with the consultancy, could affect the likelihood of a successful project. In the end, these probability inputs will be subjective estimates.

We can examine how different subjective probabilities affect our recommendation by using sensitivity analysis — that is varying the model inputs and seeing how our recommendation changes. Consider a company with multiple decision makers, each with his own probability estimates. The CEO, confident in his employees and the VoIP consultants, estimates the VoIP project will be successfully implemented with a 90% probability. The CIO, concerned about unforeseen technical obstacles in implementing VoIP, gives VoIP implementation a 70% chance of success. The VP of sales, who believes that his customer support staff will try to cancel the project at the slightest sign of any glitches, places a 50% chance of success on the project. How would these different probabilities change the results of the model? The sensitivity graph shown in Figure 2 contains the answer.

The graph explains when our base recommendation of "do not pursue a VoIP project" changes to "pursue a VoIP project." The key is in the inflection points. When we believe that the proba-

Figure 2



bility of a stable VoIP (above in pink squares) system is above 68%, the VoIP project starts to become worthwhile. Likewise, when we believe that the probability of a successful VoIP implementation saving costs (above in yellow triangles) is above is above 79%, the VoIP project also begins to become worthwhile. The Y-axis shows the "average" savings of the project. For example, if the believe that we have a 90% probability of the VoIP system being stable, the "average" value of the VoIP project is $110; on average, the VoIP project will recoup its original cost plus 110%.

Interestingly, the sensitivity analysis graph above shows that the three executives' probabilities all end with the same result: adopting the VoIP project is not recommended. Even if the company is 100% sure that it can implement VoIP successfully, the base assumption's 30% risk of telephone instability and 50% chance of reducing cost makes VoIP a poor decision. These executives should, therefore, be more concerned about the stability and cost savings likelihood of a VoIP project than worry about the success of the actual VoIP implementation itself.

We can glean a few interesting points from the sensitivity analysis graph above.

- The probability of stability is the steepest once we reach the inflection point. This means that increasing the odds of a stable VoIP system adds the greatest value. Anything the company can do without significant expense can create value for the VoIP project. A few examples may include:
  - ❒ Focus more on technical planning to ensure reliability.
  - ❒ Create an efficient outage response procedure, so VoIP issues don't affect the business.
  - ❒ Provide sufficient training to those maintaining the VoIP systems.
  - ❒ Establish strict security controls so that unqualified individuals do not accidentally bring down the system.
- The probability of cost savings inflection point occurs the earliest, at 68%, but is not as steep as the probability of stability curve. This means that anyone implementing a VoIP project should be pretty confident (~70% sure) that it will reduce costs. It's not as big of a risk as an unreliable telephone system, but a good VoIP project requires a sound business case.
- The probability of a successful implementation never reaches its inflection point. This means that other concerns, such as stability and cost savings, should trump the worries of a failed implementation. A failed project is always embarrassing, but a project that disrupts overall business and hurts revenues would be much more damaging. Ensuring telephone reliability and building an effective, cost saving business case should take priority over creating a great project management team.

Two-variable Sensitivity Analysis: When Does VoIP Make Sense?

Simply stated, VoIP makes sense if a company can make it work. The telephone infrastructure must remain stable — or at least stable enough not to impact day-to-day business operations. Furthermore, the company needs to realize the proposed benefits. A fully stable new VoIP system would be pointless if it increased costs and didn't add any useful benefits. The combination of these two factors — reliability and cost savings — is key to successful VoIP projects.

Deciding to implement VoIP would be easy if we were 100% sure that the new VoIP system would be reliable and 100% sure that the company can realize the advertised benefits. No executive would be so naïve as to believe this to be the case. Figure 3 can help decision makers.

Let's use the previous example of a CEO and CIO as two decision makers. The CEO believes that the company has an 80% likelihood of realizing the promised cost savings and 90% chance of being able to establish a stable VoIP system given the right planning. Based upon these estimates, the table (created from the Decision Analysis tree) recommends pursuing the VoIP project. The CIO, however, estimates the probability of costs saving at 50% and the probability of a stable telephony at 70%. Given these estimates, the model suggests staying with the traditional PBX phone system.

Decision makers using this table as a tool should be careful at the border between PBX and VoIP. Research shows

								Probability of Cost Savings				
	PBX	0%	10%	20%	30%	40%	50%	60%	70%	80%	90%	100%
	0%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX
	10%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX
	20%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX
Probability	30%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX
of Stable	40%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX
Telephony	50%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX
	60%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	PBX	VoIP
	70%	PBX	PBX	PBX	PBX	PBX	PBX	PBX	VoIP	VoIP	VoIP	VoIP
	80%	PBX	PBX	PBX	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	90%	PBX	PBX	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	100%	PBX	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP

Figure 3

that most people are risk adverse, especially with the possibility of significant downsides. Therefore, in cultures that hesitate in risk, decision makers may be better off avoiding the risk of a VoIP project, even if the table makes a VoIP recommendation that lies on the border.

Decision Tree Variant: New Facility

Now suppose our company faces a different scenario. We are building a new facility, and the telephone network has not yet been built. In this new situation, does it make sense to install the new VoIP technology instead of the traditional PBX technology? A variant of our base decision tree can help us make this decision.

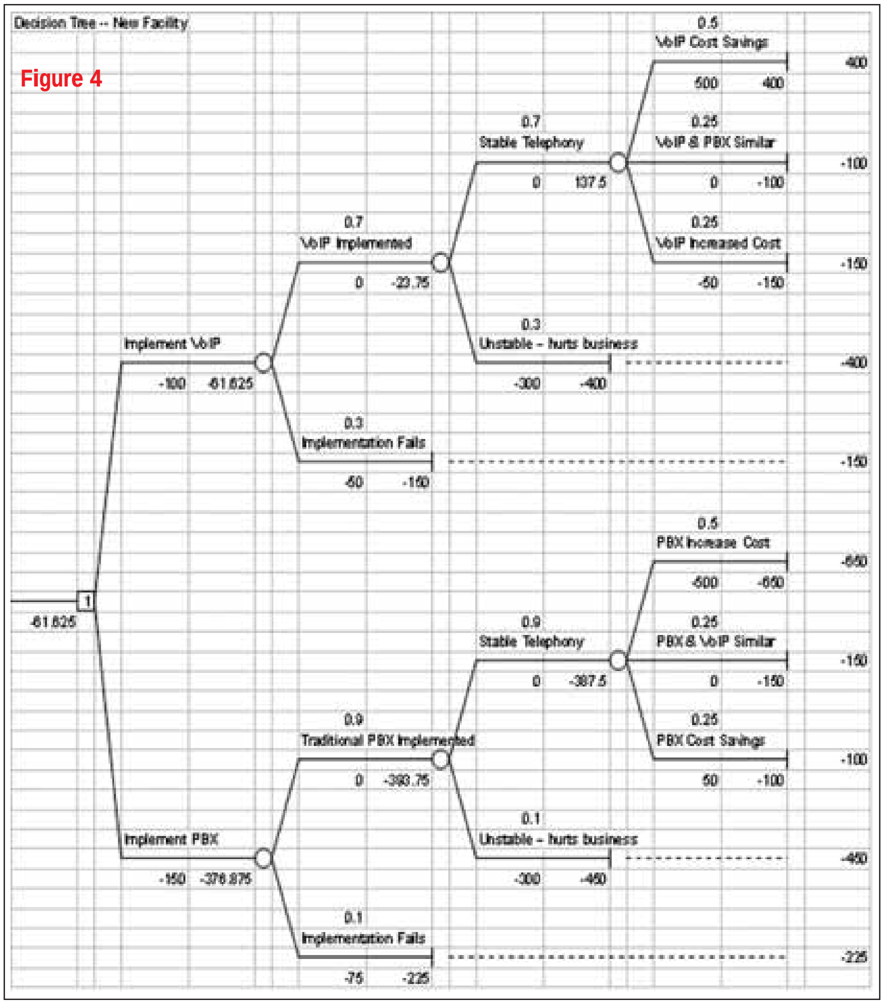
Our new decision tree is shown in Figure 4. This new model has three important changes.

• The PBX system has not been installed. The installation of a new PBX telephone system faces the same uncertainties of VoIP system. For example, the implementation of the new PBX system could fail; the new PBX system might be unstable; and the PBX system may be more expensive than the VoIP system. These three uncertainties are represented with the new uncertainty event nodes (shown as circles).

• A new VoIP system is less expensive to install than a new PBX system. The VoIP system will run on computer network infrastructure, thereby eliminating a second set of wires and cables.

• An unstable telephone system, while still costly, won't be as damaging as the base scenario. Employees aren't currently stationed in the new facility. Unreliable systems will mostly amount to more troubleshooting and delays, rather than interrupt crucial business processes and hurt sales.

This new decision tree recommends pursuing VoIP. This new VoIP system will have an "average" price of 61.6 monetary units — meaning that we expect the potential benefits and cost savings of this new VoIP system to pay off 38% of the upfront cost. The VoIP



system is much less expensive than a new PBX system, which the model expects to have a TCO of $376.9. Notice that both telephone systems will have a net cost, so we choose the less costly of the two. The VoIP system faces more uncertainty around the implementation and the reliability, but it is less expensive immediately and in the long term.

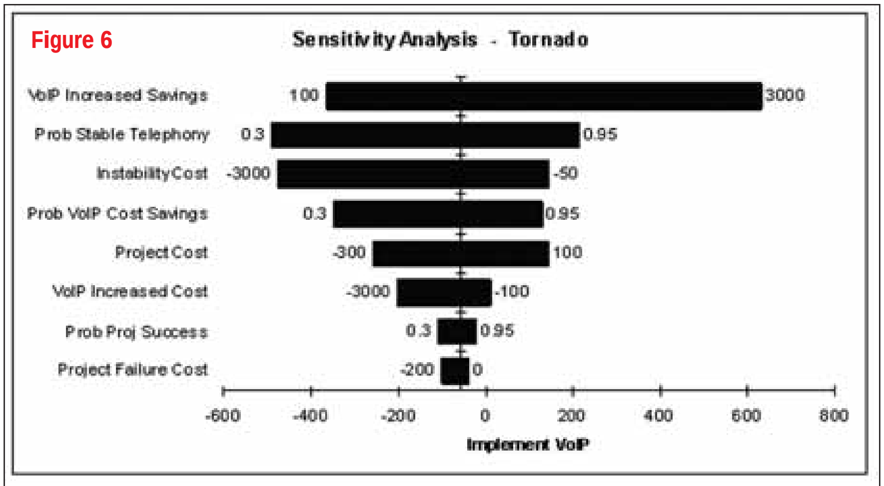
Let's revisit one of our earlier questions: When does it make sense to implement VoIP? In our base scenario, we had to be mostly sure (>80%) that the company could make it work. But, in our new facility scenario, we don't have an existing telephone network and the stakes are lower. A decision maker would only need to be fairly sure (>40%) certain that the company can make it work. A two-way sensitivity analysis is shown in Figure 5.

With our base case of 70% VoIP stability and 50% cost savings, the model makes a solid recommendation to adopt VoIP. This sensitivity analysis table recommends, in general, those who are fairly comfortable with the new VoIP technology should implement it in their new facility.

Sensitivity Analysis on VoIP Stability and Cost Savings

Figure 5

		Prob VoIP Cost Savings										
	VoIP	0%	10%	20%	30%	40%	50%	60%	70%	80%	90%	100%
	0%	PBX	PBX	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	10%	PBX	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	20%	PBX	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
Prob	30%	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
Stable	40%	PBX	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
VoIP	50%	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	60%	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	70%	PBX	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	80%	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	90%	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP
	100%	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP	VoIP



Top 3 Variables: What's Important in a VoIP Project?

The greatest value in this decision tree lies not in the final recommendation, but the analysis. Astute executives and decision makers rely on both numbers and intuition to make sound business judgments. A helpful intuitive benchmark for understanding this VoIP decision comes from the tornado graph below.

The tornado graph (Figure 6) provides a visual explanation for the most important variables in the decision making process. The most important variables bubble up to the top of the diagram. The less important variables fall to the bottom.

To read the tornado diagram consider this question: What would happen to the total cost of VoIP if one of our decision variables changed while we were implementing the project? For some variables, the swing can be significant. For others, the difference would be barely noticed. Consider a successful VoIP project that saves the company money. If it saves just a little bit of money — just enough to recoup the cost of the project — the overall value of this project is almost negative $400. However, if we see a $3,000 net present value savings, the project can pay for itself almost six times over ($600 normalized).

The top three variables in a successful project, as shown in the tornado diagram, are

1. The value of savings, or benefits, in a successful, stable VoIP implementation that saves the company money.
2. The likelihood of a stable telephone system.
3. The cost (including lost revenue and reputation damage) that will be incurred if a telephone system fails.

The most important factor in approaching a VoIP project is the ongoing savings and net benefits of a new VoIP system. In terms of hard dollars, this means reduced maintenance costs (e.g., payroll, tech support, technology replacement, and long distance fees). Soft dollars should also be included in this estimate. For example, the opportunity cost for missing a technology trend or upcoming technologies and benefits that may be supported on VoIP but not on traditional PBX systems. To increase the chance of the overall success of a VoIP project, decision makers should pay careful attention to what vendors call Total Cost of Ownership (TCO), because ongoing maintenance cost is the main business driver and the most important factor in a VoIP project.

The risk of an unstable telephone system makes up other most important factors in a VoIP project. In our model, we break this risk up into two separate components: the probability of experiencing telephone instability that would impact business and the cost of this instability, should the telephone system actually become unstable. Companies interested in VoIP projects can take many steps to mitigate this risk; many possibilities — such as better training for those maintaining the new VoIP system, thorough assessment of the technology, and strict security access — were suggested above. However, reducing the impact of an outage, should one occur, is very important. Back-up procedures and alternative technologies may significantly reduce the cost of telephony problems. For example, companies can leverage mobile phones or allow employees to work from home. Otherwise, the company may leave a few strategically placed traditional phone lines, in key areas (i.e., CEOs office or the network administrator's office). This backup system can help important business continue or allow troubleshooters to effectively do their jobs during an outage. Another way to minimize the damage caused by VoIP outages is to roll out this technology slowly. By implementing one location at a time in a tiered system, the company can allow certain offices to cover for others, should VoIP become unstable.

Conclusion: Manage Your Risk

VoIP projects, because of the risk involved should, indeed, be approached carefully. A successful project requires a very good ROI and carefully managed risk of telephone instability. Converting to VoIP from a stable traditional phone system only makes sense if the decision maker is confident (>80%) that the project will result in significant savings and that telephone instability won't hurt business, but those building new facilities only need to be somewhat confident (>40%). Through careful planning, such as detailed financial analysis, technology understanding, and well thought out measures to prevent telephone instability from affecting business, many companies should be confident enough to deploy VoIP — at least in their new facilities.

While VoIP may not be the solution for everyone, those who implement it properly — and in the right situations — can garner significant benefits and cost savings from it. Successful use of VoIP requires making the right decision initially and carefully navigating the VoIP implementation process. IT

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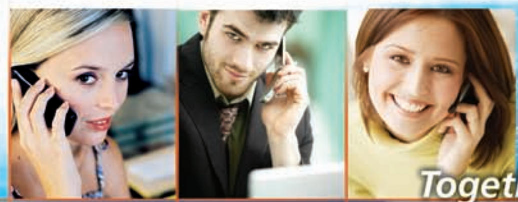

Together we make the call™

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Eicon Networks, founded in 1984, provides media processing hardware and software that enable developers and system integrators to deliver applications for voice, speech, conferencing, VoIP, fax and IP based solutions. Eicon's flagship Diva™ Server products share a powerful common architecture along with a wide range of tools for developers and world-class market leading support.

We are headquartered in Montreal, Canada with offices worldwide, distributing, marketing and supporting our products through an extensive network of channel partners, independent software vendors (ISVs), telecommunication companies (telcos), service providers and original equipment manufacturers (OEMs).

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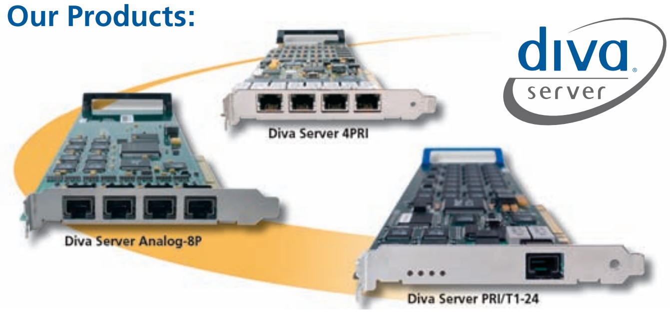
Our Products:



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- Roar Hagen, Ph.D., Chief Technology Officer and Co-founder
- Edward Abbati, Senior Vice President and Chief Financial Officer
- Jan Linden, Vice President Engineering
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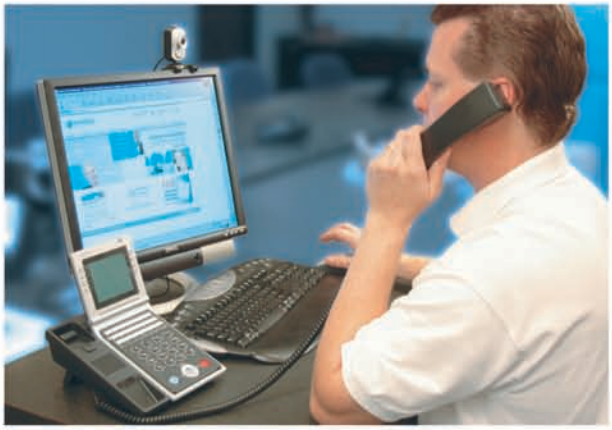

Three Decades of Experience

Multi-Tech Systems is an ISO 9001:2000-certified global manufacturer of award-winning telephony, Internet, remote access, and device networking products. Since 1970, Multi-Tech Systems has earned a worldwide reputation for delivering reliable products of innovative design. New technologies, same focus: connecting voice and data over IP networks.

"Multi-Tech Systems is unique in that it has approached the Internet telephony market with expertise in both data and telecom networks and we've been doing it for over three decades," explains Chip Harleman, Multi-Tech's Vice President of Sales and Marketing. "We developed the first modem, the acoustic coupler, in 1971. And, in 1992, introduced the world's first patented, simultaneous voice and data modem. Very few companies can match our data and voice expertise."

Our Approach to Internet Telephony

Multi-Tech's approach to the Internet telephony market is to target corporations that have invested heavily in their legacy phone systems and data network infrastructure, but are looking for an IP migration path. "We believe strongly in converged communications," explains Hari Arimilli, Multi-Tech's Engineering Manager for VON products. "We also believe we're in the early stages of this trend which means there are still a large number of installed, legacy systems out there. Multi-Tech's telephony solutions allow existing systems to merge effectively with, and migrate to, newer technologies. This allows companies the comfort of making a gradual transition to converged communications while utilizing the existing investment in their legacy systems."

Our Products

Multi-Tech **MultiVOIP Voice over IP gateways** integrate voice and fax communications into a data network providing distributed IP telephony, survivability and PSTN trunking, as well as, toll bypass savings to remote offices of multi-location businesses. So, whether you are looking for an IP migration path, or are ready to convert over to a pure distributed IP environment, Multi-Tech has a solution that is right for you.

Multi-Tech's innovative **FaxFinder distributed faxing** solution, connects directly to your existing PBX, and allows users to receive faxes as e-mails and send them from any application that can print. It even delivers faxes over a WAN to the desktop of remote offices and field sales people.

Multi-Tech's **TalkAnytime web-based click-to-talk media server** provides direct, real-time interaction, over the Internet, between web site visitors and your sales and support staff. It is designed to help you close sales, enhance customer service, build customer loyalty, and ultimately generate increased revenue for your company.

Multi-Tech's **CallFinder DID-enabler** automates inbound call routing through non-DID phone systems making them smarter and more efficient.

Multi-Tech's **CallFinder GSM cellular gateway** connects to a PBX trunk line, PBX extension line, or a single PSTN line and routes incoming and outgoing calls through the GSM wireless network. It enables companies to take advantage of lower cost GSM networks to provide a substantial savings in their overall telephone bill.

Multi-Tech's telephony solutions **preserve** a customer's infrastructure **investment** while deploying **converged solutions** to **meet** their **future needs**.

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MultiTech® Systems

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NEC Corporation is one of the world's leading providers of Internet, broadband network and enterprise business solutions dedicated to meeting the specialized needs of its diverse and global base of customers. NEC delivers tailored solutions in the key fields of computing, networking and electronic devices.

Charged with providing integrated communications to the enterprise, NEC Unified Solutions delivers the industry's most innovative suite of products, applications and services to help customers achieve their business goals. NEC Unified Solutions, a subsidiary of NEC Corporation of America, offers the broadest range of communication choices, flexible platforms, and an open migration path to protect investments.

NEC Unified Solutions is a Cisco Systems Gold Certified Partner, an Advanced Technology Partner and an IP Telephony Specialization Partner.

Partners & Alliances

NEC Unified Solutions recognizes that key partnerships and alliances are critical to business success. Current strategic relationships include:

- Active Voice®
- Cisco Systems®
- Counterpane® Internet Security
- Dukane®
- F-Secure®
- Top Layer®
- Xtend®
- Polycom®
- Zeacom®
- Genesys®
- FrontBridge®

Customers

NEC Unified Solutions serves the Fortune 1000 and customers in vertical markets such as:

- Architecture
- Automotive
- Education (higher education and K-12)
- Entertainment
- Financial
- Government (federal, state and local)
- Retail
- Healthcare
- Hospitality
- Legal
- Manufacturing
- Pharmaceutical
- Utilities

Solutions

NEC Unified Solutions' comprehensive portfolio includes:

- Data Networks
- IP Communications
- Wireless Communications
- Broadband Solutions
- Video Solutions (Traditional and IP)
- Network Security (Consulting IDC, Firewall)
- Carrier Services
- Implementation Services
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The Patton brothers, Bobby and Burt, founded Patton Electronics in 1984, while students in college. Over the succeeding 20+ years, Patton has taken those simple beginnings and expanded into a multi-national manufacturing company that today employs more than 180 people and provides a product line in excess of 1000 items.

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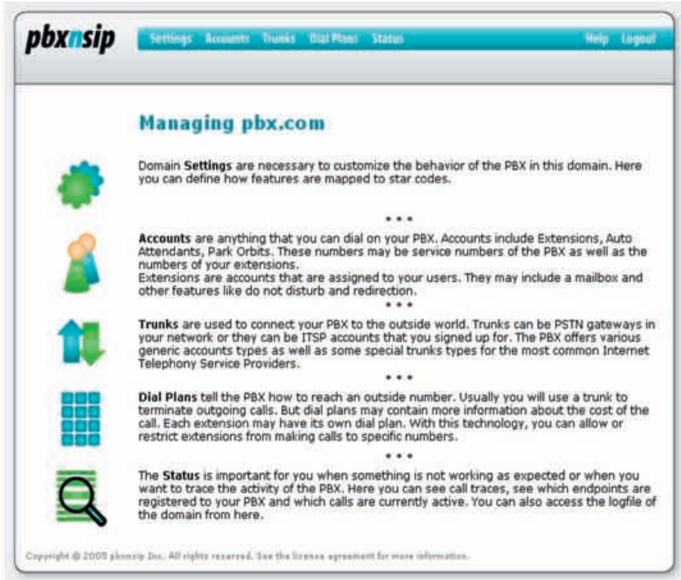

independent from a single vendor and maximizes the choice on devices. The pbxnsip PBX is interoperable with almost all available SIP devices on the market. Check out our web site for a listing of available devices.

The PBX was designed to minimize the total cost of ownership. By supplying plug-and-play support via TFTP, HTTP and HTTPS, instant configuration, built-in web server and SNMP protocol support it reduces the TCO compared to other solutions.

A special focus of the product is on security. Protection against eavesdropping is a must-have in corporate environment, especially with upcoming sniffing tools that make such attacks possible from every seat in the local area network. The PBX uses sips and SRTP to protect the traffic on application layer, using existing security infrastructure components like certificates.

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Tadiran Telecom

For over 40 years Tadiran has been satisfying real-world communication needs of organizations around the world.

A company rich with history

Tadiran Telecom products have long been the standard by which other communications solutions are measured. Forming strategic alliances over the last 40 years, our products have been represented by some of the world's largest telephone and communications companies.

Today, Tadiran Telecom solutions are sold in 41 countries with tens of millions of ports installed. Best known for our Coral line of converged IP-PBX systems, Tadiran maintains a commitment to never leave a customer behind by providing a clear migration path.

A commitment to R&D and tomorrow

Tadiran's commitment to R&D is evident with new products like the Sea Softswitch. While most manufactures ported their legacy PBX components, Tadiran built an entirely new softswitch. In 2006 we will also release a new series of IP telephones and a new desktop productivity tool - NAVIGATOR.

Tadiran America Business Partners

Tadiran products are sold exclusively through business partners. Tadiran America Business Partners provide sales and service to our customers in North and South America. These resellers must maintain technical certification. Tadiran America is based in Port Washington NY.

CORAL
SEA SOFTSWITCH
multimedia communications

Tadiran Government Systems

Tadiran Telecom products continue to be the choice of federal, state and local government agencies around the world. Located in Alexandria, VA, Tadiran Government Systems is experienced with purchasing procedures and provides complete project management.

Technology that speaks volumes

Tadiran designs and manufactures communications products for organizations of all sizes. Our Coral IPx systems provide complete hardware/software telephony solutions. The Sea Softswitch is a entirely new SIP softswitch that features a revolutionary "replication" architecture that provides a seamless distributed communication system for all size organizations. Our Composit Contact Center is a flexible call center application with database integration and agent toolbars. Tadiran produces both softphones and desktop telephones in a variety of models. We offer standard SIP telephones as well as MGCP.

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IP-6812

- Full duplex handset • Full duplex speakerphone
- 3 x 24 LCD display • Up to 10 feature buttons
- Up to 11 line appearances
- Hold & mute buttons • DSS/BLF
- Handset/speaker/ringer volume
- Headset jack • MGCP/SIP Protocol
- Power over Ethernet compatible

IP-6804

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VoIP ADMINISTRATORS

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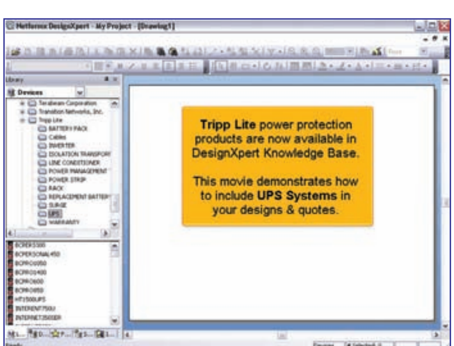
Since 1922, Tripp Lite has been recognized for innovative products and superior service. The company is a leader in the power protection industry, with 1,000+ products, including a range of uninterruptible power supply (UPS) systems optimized for the protection of VoIP telephony systems.

Now in DesignXpert: Tripp Lite UPS Systems

Tripp Lite has partnered with Netformx to include Tripp Lite UPS Systems in the DesignXpert knowledge base. The addition of Tripp Lite to DesignXpert enables users to incorporate UPS power protection/backup during the design process. For VoIP systems and other mission-critical applications, adequate power protection and power backup are key design issues. The Netformx/Tripp Lite partnership makes it easier to assure the safety and reliability of networks designed with DesignXpert.

NETFORMX



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All in one

Complete, single-box Business Gateways for cost-effective provisioning of hosted IP services to SME and enterprise customers

About U4EA Technologies, Inc.

U4EA Technologies provides telecom service providers and network equipment manufacturers with advanced single-box solutions for the customer premises.

These integrated Business Gateways are fully interoperable and provide a complete set of networking features for the cost-effective and secure delivery and management of converged carrier-class voice and data services to SMEs.

U4EA Technologies is based in Fremont, CA.

VoIP Session Controller
Traffic Shaper (QoS)
WAN Router
FXO/FXS
Firewall
Lifeline

ICAD40 - SME Business Gateway

ICAD80 - Enterprise Business Gateway

Features	Benefits
Advanced **session control** for VoIP	Allows service providers and users to set call admission policies suitable for their business
High performance **security** with wirespeed IDS, firewall, NAT traversal	Protects against DoS and malicious intent and ensures reliable delivery of VoIP and data traffic across public/private network domains
Multi-service **QoS** and real-time voice and data quality monitoring	Guarantees low latency of VoIP and zero dropped calls, allows voice quality monitoring against SLAs
Supports a mix of legacy and IP technologies, **IP connectivity for analog phones** and faxes	Protects existing investments and allows smooth migration to IP based services
Ethernet, T1/E1 and ADSL **WAN access** options	Greater flexibility at a lower price point
Built-in **VoIP survivability** and redundancy	Enables redundancy and survivability via lifelines
PoE LAN switch	IP phones and wireless access points powered (and rebooted) remotely

ICAD Business Gateways

U4EA's ICAD series of Business Gateways give service providers enormous potential for maximising revenues through cost-effective provisioning of IP services to SME customers.

Reliable delivery of managed communications services to SMEs

Cost savings through integrated functionality

Secure and reliable VoIP delivery

Service demarcation and ease of management

Interoperability

These integrated devices are single-box solutions for the customer premises, providing a complete set of networking features for converged voice and data services over IP. ICAD devices provide a convenient transition path from legacy services where customers can be smoothly migrated to all-IP. Both new and existing services are simple to manage and ensure reliable service delivery, providing new revenue streams, opportunities for Opex and Capex savings and reduced customer churn.

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USA DATANET[SM]

IP NETWORK SOLUTIONS GROUP

Changing the way the world communicates

USA Datanet is at the forefront of the development and commercial application of Voice over Internet Protocol (VoIP) and softswitch network technology, a powerful convergence of voice and data that is rapidly transforming the communications industry across the United States and around the world. The company has developed an advanced, next-generation VoIP network with the capacity, scalability and flexibility required to meet the rigorous demands of the highly competitive and rapidly evolving telecommunications market. Over the past eight years, USA Datanet has leveraged this network to build a successful and consistently profitable multi-million dollar telecommunications business serving hundreds of thousands of customers.

Understanding and serving the carrier market

Compelling market and economic factors are driving growth in carrier markets, as traditional service providers large and small are faced with increasingly difficult decisions concerning the inevitable migration of their own network infrastructures. Customer demand for enhanced services, increased competition from new VoIP telephony providers and the eroding economies of narrowband network delivery create opportunities for operators of advanced VoIP networks.

USA Datanet's IP Network Solutions (IPNS) Group provides VoIP telephony and customized IP network services to ILECs, CLECs, ISPs, value-added channel partners and other enterprises on a wholesale basis. IPNS offers carrier customers a number of tailored services including Nationwide Integrated VoIP PRI, Hosted IP Services, IP Termination/Origination, Dedicated Internet and a variety of Co-Location options. In addition, the Company's consumer broadband telephony service is also available as both a private label resale and a wholesale service.

USA Datanet's switching, network management systems and network operations center reside in the company's Data Center in Syracuse, NY. This efficient and scalable center provides support for the company's commercial and carrier customers from a team of service and technical support professionals, delivering service that includes proactive notification of service-affecting events prior to customer awareness.

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IP TERMINATION AND ORIGINATION SERVICES

Allows carriers and VoIP providers to connect to USA Datanet's network via direct IP, public Internet or circuit-based facilities. Our rates are competitive for domestic and international termination.

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VoIP MARKETPLACE

CEO Spotlight

Alan Pound
Founder and Managing Director
Aculab

In the CEO Spotlight section in *Internet Telephony®*, we recognize the outstanding work performed by exemplary companies. Each month, we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Alan Pound, Founder and Managing Director of Aculab. (news - alert)

GG: What is Aculab's mission?

AP: At Aculab, our mission is made up of three components — products, people, and profit. In respect of our products, we strive to be respected worldwide by customers and competitors alike as a leading provider of enabling technology for the communications market. This is achieved through providing excellent value, quality products, and support to our target markets. With regards to people, Aculab recognizes the importance and value of its staff and its mission is to provide reward and recognition to all Aculab employees and to maintain a work environment of satisfaction, pride, and fun. The profit element of Aculab's mission relates to achieving controlled financial growth to facilitate the objectives outlined above.

GG: What is your vision for Aculab and how is the company positioned in the next generation telecom market?

AP: Aculab intends to remain a provider of enabling technology to application developers in a converged communications market. We will expand our high-density IP platform, Prosody X, to include a range of codecs and protocols enabling mobility, security, and video. We also anticipate seeing an increase in take-up of our host media processing software, Prosody S, and plan to incorporate similar enhancements and features to Prosody X. Aculab believes that solution providers will increasingly demand enabling technology functionality in the form of 'plug and go' appliances, and will service this need through collaboration with our specialist partners.

GG: What are the latest product developments at Aculab and what new developments can customers hope to see over the coming months?

AP: Prosody X, Aculab's high-density, IP-based media processing card, was released to general availability in PCI format in January 2006. Aculab is making continued, ongoing enhancements in terms of new codecs for security and video. Prosody X cPCI has just been released and offers double the density of the PCI version — up to 1,200 channels per card. It also enables hot-swap and is a telco grade product. We plan to make a third form factor, Prosody X PCIe, available towards the end of 2006. We envisage making an ATCA environment offering available during 2007.

In terms of our host media processing product, Prosody S, there will be three further releases during the next phase of development, incorporating G.723, G.726, and G.729 codecs as well as iLBC, secure RTP, T.38 fax, RTCP, and live speaker detection.

Aculab's SS7 product offering took a major step forward earlier this year with the addition of dual, resilient MTP3 and distributed ISUP capabilities. We also added a 'flexible' ISUP feature and further progress is being made in the areas of TCAP and SS7 monitoring.

GG: What makes Aculab's product offerings unique and how do companies benefit from using them?

AP: Essentially, Aculab is an R&D company. Our focus is on rising to the engineering challenges presented to us through developing leading edge hardware and software technologies. We focus our resources on continually enhancing and developing our products to give customers who have chosen to develop to our API, a good return on investment. At Aculab, we can now provide our customers with a range of low- to high-density solutions, offering maximum flexibility. Developers choosing Aculab also benefit from a consistent API, making it easy to port applications across various product releases.

In addition, we support our customers through offering comprehensive pre-sales technical consultancy, customer training services, post-sales, and technical support. We also provide them with strong marketing support in taking their Aculab-based products to market. Our customers benefit from the fact that Aculab is a privately owned, established, debt free company, which is focused on establishing long term relationships.

Aculab continues to support integrations with third-party vendors' products

> Everyone is at a different stage of the adoption cycle, so we cannot provide businesses with a single set of messages and technologies, as each set of information needs to be appropriate to the individual company.

such as Asterisk and Microsoft Speech Server and for a number of speech technology vendors. Aculab is also making ongoing protocol developments, particularly with SIP and is continuing to work closely with partners.

GG: What are the most pressing issues facing our industry today, and what can be done can to alleviate these problems?

AP: The biggest issue facing our industry today is migration and how businesses will manage the transition from existing TDM technologies into IP technologies. Everyone is at a different stage of the adoption cycle, so we cannot provide businesses with a single set of messages and technologies, as each set of information needs to be appropriate to the individual company. To address this, we need to understand how businesses' evolve and the information required at different points of the learning curve. This can be achieved through putting in place a sound infrastructure of training, pre- and post-sales support, and effective marketing, which is what we have done at Aculab.

The industry also needs to meet the challenges of rapidly maturing VoIP — increasingly voice comes for free, so we need to decide how revenue can be generated, how we will deal with security issues, and how we can ensure a high quality of service is maintained. Voice is going through a rapid paradigm shift, moving from an environment that has been stable for many years. SIP must also undergo further development in order to fulfil what people expect in terms of voice communications.

GG: Describe your view of the future of the IP telephony industry; also what does the future hold for Aculab?

AP: I believe VoIP is here to stay. There are still issues that the industry needs to work through, such as how revenue will be generated and who will generate it. From Aculab's perspective, there are a number of engineering challenges to conquer, and we look forward to overcoming these. We believe that Aculab's sound financial footing, stability, and continuity in terms of ownership and strategic direction, will serve us well, enabling us to rise to those challenges and continue to support developers who choose Aculab. IT

CEO Spotlight

Tom Linhard
President
FaxCore

In the CEO Spotlight section in *Internet Telephony®*, we recognize the outstanding work performed by exemplary companies. Each month, we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with FaxCore (news - alert) President, Tom Linhard.

GG: Fax has been around for a long time. Why has fax continued to be a viable communications technology in today's sophisticated market where fax competes with e-mail, IM, and the Internet?

TL: A Scottish inventor, Alexander Bain, was granted a British patent for a facsimile device in 1843. Bain's idea was to create a transmitter that would pick up images that would then be transmitted using a telegraph wire and translated to paper images on the receiving side. Although the technology has changed, the fundamental idea is still intact. E-mail and IM are text-based and fax is an image. It is the key reason why fax continues to be an important technology for communicating information and, for non-technical people; it doesn't get much easier than putting a piece of paper into a fax machine and dialing a phone number.

GG: Good points, but that same image could also be scanned into a computer and sent as an e-mail attachment, so why hasn't e-mail replaced fax?

TL: Certainly, for many documents, e-mail can replace fax but there are several limitations with e-mail. First, you need a scanner and a PC with connections to the Internet for e-mail. In its simplest form, fax only requires a phone line and a fax machine. Also, as opposed to e-mail, a fax transaction provides the sender with a solid confirmation of a complete successful send or a failure. The second key reason is that a fax is considered a legal document in most countries around the globe. An e-mail has very limited legal status for banking, legal, medical, financial, and other major industry segments, so fax remains the technology of choice.

GG: So e-mail and the Internet are not threats to the success of fax and FaxCore?

TL: On the contrary, e-mail and the Internet plus IP communications have breathed additional life into fax.

Many information workers have become accustomed to sending and receiving documents by e-mail rather than fax. On the upside though, faster and more reliable computing platforms have enabled the processes around document delivery to be moved from manual processes to e-mail, the Internet, and fax servers. This is a situation where the phrase "a rising tide lifts all boats" truly applies. The explosion of messaging has renewed enterprise focus on which communications technique best fits specific types of documents. Add in increased regulatory compliance requirements, such as Sarbanes-Oxley and HIPPA, and you have legal requirements that are best satisfied by fax servers. Sending and receiving from a fax machine is easy but difficult to track. A fax server logs what was sent, who sent it, when they sent it, and where they sent it. At FaxCore, we realized that enterprises are also focusing on productivity and mobility, so we designed a product that placed top priority on these two key criteria. Our customers can have both a mobile workforce and the security of central control.

GG: A lot of vendors are looking for the ideal architecture/feature set to address mobility and productivity. What advice can you provide?

TL: The concept is actually quite simple: Build a product that integrates easily and seamlessly into the enterprise with maximum reliability. Plus, use components that today's enterprises demand and make all these features standard. FaxCore is built on .net and our SDK offers fast and easy application integration while our SQL-based database and workflow engines make it reliable. FaxCore also supports Active Directory, includes tight integration with Outlook/Exchange, incorporates a pure Web browser, and integrates with VoIP solutions, like Cisco's Call Manager, via SIP. Every day, more mobile workers are connecting their laptops from home or WiFi hot spots. Mobility is just as important for fax as it is for VoIP, so creating, viewing, sending, and receiving faxes in a mobile environment is a must.

GG: Okay, fax has evolved from a simple telegraphic transmission to today's IP world. We are in the midst of a colossal and accelerating shift from circuit-based to packet-based communications. How will fax communications be impacted by this? What does the fax the industry have to do to stay around for another 150 years?

TL: 150 years may be pushing it a little, but another 20 years is more than reasonable. VoIP (define - news - alert) is not yet the indispensable telecom solution, but it is well on it way. A recurrent theme expressed is that IP telephony plus converged IP communications are driving the market. Many companies that planned on a single connection for all their VoIP, data, and other communications quickly realized that they had to roll the fax machines or fax servers back to the traditional PSTN connections. Our work, in cooperation with leading vendors, like Cantata, allows companies to move all fax traffic onto their VoIP network and truly have a pure IP voice and converged communications system. As long as the fax industry vendors continue to keep pace with the needs of their customers, fax will remain a viable communications solution. IT

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CEO Spotlight

Richard Minervino, Sr.
Chairman and CEO
Profitec Inc.

In the CEO Spotlight section in *Internet Telephony*®, we recognize the outstanding work performed by exemplary companies. Each month, we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Richard Minervino, Sr., Chairman and CEO of Profitec Inc. (news - alert)

GG: What is Profitec's mission?

RM: Profitec's mission is to be the preferred provider of integrated billing and OSS solutions for communications related endeavors. To help us achieve that goal, we have created a large data processing center with in-house print facilities, we develop all of our own software, and we have a back-office services division, which is open 24x7x365, capable of handling everything from order entry to customer service and provisioning, to facilitate handling of end user requirements.

GG: What is your vision for Profitec and how is the company positioned in the next generation telecom market?

RM: Profitec will continue to grow and evolve its software applications in support of new product offerings while we maintain our commitment to legacy applications provided to current and future customers. Profitec will continue to maintain our commitment to be a single source vendor, regardless of our clients' underlying resale platforms, thereby allowing for fully convergent billing. Relative to IP-based customers, Profitec's products fully support this new technology including billing, customer service, activations, and fulfillment. Profitec can handle present and future taxation and USF requirements. We currently support dozens of VoIP (define - news - alert) providers processing data from a wide variety of underlying IP-based switching equipment.

GG: Now that it appears that growth and opportunity are the trends in the VoIP industry, what possible hurdles do you see that might upset this momentum?

RM: Currently, much of the country does not yet have access to high-speed broadband, which limits the available marketplace for IP-based services. Also, the government is increasingly extending taxes and surcharges, which formerly applied only to traditional telephony-based services. Limitations on available markets and taxes and surcharges, which diminish margins, will both serve to potentially force communications providers to maintain some traditional telephony-based services in their product portfolios. It is for these reasons, among others, that Profitec will continue to provide an integrated solution that spans underlying technology platforms.

GG: What are some of the technology areas where Profitec is increasingly focusing, and why are these areas important to the future of your company?

RM: Profitec is increasingly focusing on evolving our existing Web-based applications, which facilitate e-commerce activity and real-time rating and on-demand activations. Increasingly, communications providers are dealing with diminishing margins and wish to limit their revenue exposure against unsecured customer activity. Point of sale revenue generation, usage threshold monitoring, and expedited revenue generation from new business will help companies retain their competitive advantage. By extension, Profitec's support for these new perspectives will be instrumental to our long-term viability. Profitec has been thriving for over 20 years, providing support solutions to the communications industry because we have studied the marketplace, listened to our customers, and have a total commitment to the industry. Profitec is a privately held family owned business, which has been succeeding against much bigger competition because of our unique dedication to our chosen field of endeavor. IT

Limitations on available markets and taxes and surcharges, which diminish margins, will both serve to potentially force communications providers to maintain some traditional telephony-based services in their product portfolios.

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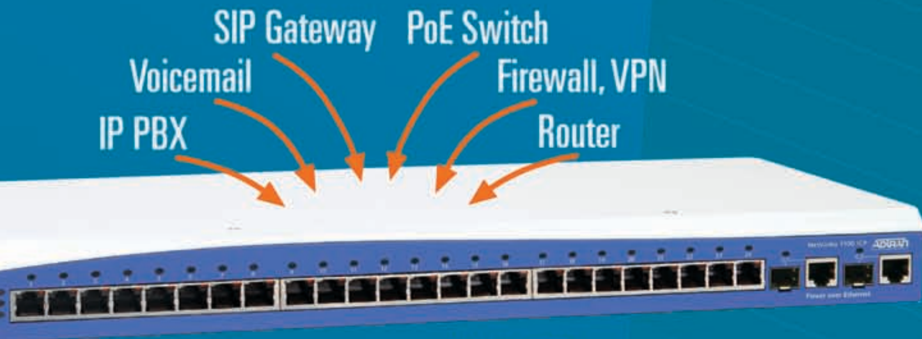

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