TMC

INTERNET TELEPHONY

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SONY Launches IPELA: "Beautiful IP"
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The VoIP Authority Since 1998™

Communications AUTOBAHN

VoIP & Presence Enable Just in Time Communications

Porsche Carrera GT

Also In This Issue:

- Siemens' President on the Future of VoIP
- VoIP Supply CEO Sayers Speaks Out

Photo courtesy of: Porsche Cars North America, Inc.


You just bought your entire company
Siemens optiPoint IP phones.
Good call.
That's because its high-fidelity voice quality is not only better than competitors, it's better than traditional phones. In fact, independent tests by CT Labs and the Tolly Group give the phones top scores for “excellent speech quality” and optimal flexibility with “greater management functionality than rival products tested.”
The optiPoint family of IP phones also offers easy-to-use features that fit into a LAN environment just like a standard data device. Like a side-car module, support for the widest range of voice compression and crucial security enhancements. And with optiPoint IP phones, you could even see your infrastructure costs and connection charges go down. So not only will you be making a sound investment, you'll be protecting it. Because you'll always be able to stay up to date and compatible with other SIP platforms, simply by adding the newest features with software downloads.
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To download a free copy of the CT Labs and Tolly Group Test Reports, visit http://enterprise.usa.siemens.com/go/opti
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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

Fast Forward

This month's issue is all about speed. The speed with which we communicate in today's world; the speed with which decisions need to be made to stay competitive; and the speed with which technology is advancing to meet the needs of end users. It indeed seems that we are hurtling down the Communications Autobahn at ever-increasing velocity.

In his Publisher's Outlook this month, Rich Tehrani recounts his conversation with Nortel (quote - news - alert) CTO Phil Edholm regarding the ever shortening time-to-decision cycle being driven by today's communications tools. For example, Rich mentions how on-demand conferencing can greatly increase the speed with which corporations can come to mutual agreement on an idea and quickly move to implement it.

Xten's (news - alert) Erik Lagerway writes about the promise of presence technology and how presence is proving itself to be a facilitator of what we've taken to calling Just in Time Communications (JITC). Presence is fast becoming more natural to use, and as work in the standards bodies and in the development labs of the vendor community continues, watch for presence to become a key enabler of applications that will propel our industry forward.

In his Inside Networking column, Tony Rybczynski addresses the speed with which VoIP (define - news - alert) is entering the workplace and the rapid return on investment that VoIP makes possible.

Marc Robins dedicates this issue's MindShare 2.0 column to a discussion of the recent lawsuit filed by Texas' State Attorney General against Vonage (news - alert) over 911 emergency calling — truly an illustration of how fast some groups can move against VoIP when they fell compelled to.

And so, it made perfect sense to find an image that represents speed and efficiency. So our thoughts immediately turned to German engineering and the fast, efficient machines turned out by famed automaker Porsche. The speed demon you see on our cover, the Porsche Carrera GT, can accelerate from a standing start to 62 mph (100 km/h) in only 3.9 seconds, reaching 100 mph (160 km/h) in less than seven seconds

Powered by a 5.7-liter, 605-horsepower (SAE) V10 engine, and featuring such unique components as the first use of a ceramic composite clutch in a production car, the Carrera GT's aerodynamic and race-bred suspension package delivers safe and stable travel at speeds of up to 205 mph.

And all this can be yours for a mere $440,000. Or you can just turn the page and dive into this month's issue, and try to keep up with the fastest-moving segment of telecom.

Now, how do I go about getting one of these babies to test drive…? And I wonder if I could keep it when I'm done…? Maybe if I made them the official car of Internet Telephony magazine...? A yellow one, with red and black lettering… Now that would be cool.

Greg

-Greg Galitzine, ggalitzine@tmcnet.com

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5. Singapore	10. France

QUOTE OF THE MONTH:

"The promise of UNE-P lay with the notion that CLECs could bootstrap investment in their own facilities over time by using a non-facilities-based approach — discounted wholesale access — to immediately build customers and a revenue base, thereby raising the money to invest in switches and access technologies over time. However, so long as the number of UNE-P access lines continued to grow without parallel growth in direct sales, UNE-P looked more and more like perpetual resale, and less and less like a wholesale business."

– Scott Wharton

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 www.tmcnet.com for all the latest news and analysis. With over 3.9 million unique page visits per month, translating into nearly half a million unique visitors, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

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

Microsoft Messenger Now With VoIP And Video Phone
Microsoft made an official announcement today about its release of a new version of MSN Messenger, its instant messaging program now bundled with VoIP-based calling and Video over IP features.
http://tmcnet.com/111.1

America Online Intros Internet Phone Service
America Online released a new Internet telephony service, an enhanced VoIP-based calling service that combines voice, e-mail and instant messaging communications.
http://tmcnet.com/112.1

Tekelec Tapped For U.S. Navy Aircraft's VoIP Call Control
Tekelec was selected by Rockwell Collins to provide VoIP capabilities for the U.S. Navy's fleet of E-6B strategic air command aircraft.
http://tmcnet.com/113.1

Special Report: The Impact of Skype And Other Private VoIP Apps
Study titled 'Emerging Business Models in Voice: the Impact of Skype and Other Private VoIP Applications' explores impact of Skype and other private VoIP applications.
http://tmcnet.com/114.1

Triple Play As Springboard To Multi-Service
VoIP is now having its promised revolutionary impact on the telecommunications industry, as new entrants like Vonage and Skype rewrite the rule book on voice communications.
http://tmcnet.com/115.1

TMC's IP PBX Channel
The IP-PBX Channel on TMCnet.com features the latest news and original bylined articles on IP-PBX. To visit TMCnet.com's IP PBX channel, just point your browser to http://www.tmcnet.com/channels/ip-pbx/. Sponsored by Sphere Communications Inc.

TMC's IP Contact Center Channel
The IP Contact Center Channel on TMCnet.com features the latest news, articles, and case studies in the booming IP Contact Center space. To visit TMCnet.com's voice channel just point your browser to: http://www.tmcnet.com/channels/ip-contact-center. Sponsored by FrontRange Solutions.

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By Rich Tehrani

Nortel, JITC, And The Road Goes On

Recently I have been hearing rumors on the Internet about Nortel merging with Cisco. While I am not in a position to comment on whether this will happen, I think it is a bad idea for those people tasked with purchasing technology as it will result in less choices. Nortel seems to be doing well these days. I wrote about them a while back and that was a very positive article as well.

What caused my last article to be written was an analyst meeting hosted by Nortel's CEO Frank Dunn. Turns out, things at Nortel weren't as good as we were told and a financial scandal ensued and Dunn and others stepped down.

But recently I have been in meetings with Nortel (quote - news - alert) executives from the service provider, cable, and enterprise divisions and this time it seems good things are indeed happening. Throughout the financial scandal, Nortel has been selling product and some of the independent research I have seen shows Nortel doing well in various market share reports.

At our last Internet Telephony Conference & EXPO in Miami I was able to sit down over breakfast with Nortel CTO Phil Edholm and Tony Rybczynski, director of strategic enterprise technologies and TMC columnist for over five years. They tell me that VoIP (define - news - alert) deployments are really taking off. They say that companies putting telecom in new buildings are going straight to VoIP and some of their installments are in the 20,000 to 30,000 port range.

VoIP deployments are three to four times what they were in 2003 and Phil explains that there is a SIP explosion taking place. I am sure he meant that in a good way. Nortel is integrating SIP (define - news - alert) devices and in the process, transforming how we communicate. They are focusing on mobility as well and they have worked with RIM to develop an in-building device that will use WiFi telephony and SIP to communicate with Nortel PBXs (define - news - alert) .

Speed and efficiency are just starting to come to corporations and consumers.

Nortel has a wireless SIP client as well. They see the coming of device proliferation where there will be myriad devices with different functions. In Europe, Edholm points out they have more device freedom that we have in the U.S. and in Asia there is even more. There are numerous other differences between the various cell phone markets... For example, we expense cell phones in the U.S. while in Europe they do not.

Behind all these devices you need intelligence and that's where Nortel's MCS or Multimedia Communications Server comes in. This product helps you decide where a call goes — cell phone, PDA, PC… and depending on what device is available and your personal availability, it routes the call. MCS can do many things such as be a personal agent, handle calls, integrate with Outlook, and provide collaboration services.

Our conversation went into security and we discussed SIP authentication. Edholm thinks data stream security isn't that important as it is tough to see traffic on the Internet. He says regardless of the need, encryption is coming. He says that there is danger in what is happening with SIP and security. As Edholm points out, SIP is a peer to peer technology and as such corporations have no control over communications. This is bad for a number of reasons. Phil mentions that you can send files over SIP encrypted lines. Can you imagine a virus spreading over these lines? In my opinion, this is a corporation's worst nightmare — just after having to deal with Sarbanes Oxley of course. P2P technology totally bypasses servers that are set up to scan for viruses, worms, etc.

Another issue worth considering is the tapping/recording of calls. Corporations need to address these issues as do governments. Encrypted SIP lines present sticky challenges. If you look at what Microsoft espouses, they say we need encrypted IPSEC (define - news - alert) tunnels. Edholm reiterates that in this environment as above you can use SIP to send encrypted distributed viruses.

My breakfast companions went on to tell me that we do need encryption for privacy but at the same time the encryption needs to be under corporate control... The enterprise must be able to see this traffic.

For example, the FTC says that all calls, instant messages, and e-mails in financial markets need to be recorded. But Edholm asks, what happens when you have users making Skype calls? Skype bypasses corporate control completely. Sarb-Ox demands transparency while P2P clients present unique challenges.

You have heard me talk about Just in Time Communications or JITC before and I believe the concept will change the world. It will make us all more productive and give us back control of our communications. Phil has been an

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incredible sounding board and has given me lots of good ideas regarding JITC. Phil mentions how the time-to-decision is crucial in today's organizations. We need to be able to have on-demand conferencing in our organizations. In business areas such as healthcare or supply chain management speeding up decision-making is critical.

Companies are using time-to-decision as a marketing differentiator. For example financial institutions are differentiating themselves by telling their customers they will approve loans in record time. In my opinion, when the telecom market can be leveraged by corporations to increase sales and reduce cost, we have a bright future indeed.

JITC

Since my last column on Just In Time Communications, I have been deluged with people telling me that the concept hit home with them soon after reading. Examples from readers, such as being on the phone and getting an unimportant interruption, have been flooding my inbox. I think as an industry this concept really hit home. We are on the cusp of seeing great things in communications. Speed and efficiency are just starting to come to corporations and consumers. The latest generation of find-me/follow-me technology from today's VoIP service providers embodies the best of JITC, but I believe we're still seeing the earliest stages of their usefulness. Amazingly, most every consumer using VoIP has better productivity features than their corporate counterparts. You wonder why Skype (news - alert) is catching on in enterprises? It is because you get more JITC features with Skype than you do with your million-dollar PBX! To be more specific, you can quickly escalate a chat session to a call if boh parties have the time. This is JITC in action.

To reiterate… We are heading into a world where communications is getting more complex and we are communicating with more people than was possible just 10 years ago. The advent of IM, e-mail, cell phones, VoIP phones, office phones, home phones, etc. and an associated message store with all but IM (I am sure somewhere there is someone working on the killer IM storage application) is unmanageable. Checking all of these stores while simultaneously fending off the callers, e-mailers, IMers, while trying to have a career, much less a personal life is becoming impossible. I am fascinated and somewhat amazed at how many other people have to check e-mail on nights and weekends because they just can't keep up with it all.

This problem will get worse before it gets better. There is no doubt about that. What we need is better software that allows us to turbo-charge our communications. The reason there is a state-of-the-art blazing-fast sports car on the cover of this magazine is because that represents where communications is going. Just in Time Communications is the future. There is no way we can continue the way we are going and maintain our sanity. We need integration of VoIP, presence, browsers, preferences, service providers, and equipment providers to allow us to create the ultimate Just in Time Communications killer applications. I look forward to seeing you all out on the Just in Time Communications Autobahn in the future.

VoIP Service Marketing 101

If you are VoIP service provider and you don't read this entry you are in trouble. It is that simple. I spoke with Level 3 recently and they told me that they just completed a comprehensive study of VoIP and have some amazing findings to share.

The survey consisted of 1,400 people and has a confidence of plus or minus 4 percent. The research shows that many people will switch to VoIP. Let's look at the numbers:

- With an aided description of VoIP 71 percent are open to consider a switch to VoIP.
- With an unaided description of VoIP 60 percent are open to switch.

Here is probably the most important statistic you will ever see. Please commit this to memory. If you discount your VoIP service too much, people will associate you with cheap service. They say the sweet spot is $35 or 25% less than the incumbent price.

I need to thank Level 3 for validating what I have been saying all along. In my view purchasing cut-rate telephone service is akin to purchasing used Band-Aids. You just don't want to rely on something that gives you 911/life-and-death service and recall that you saved a bundle on it. Yes, savings are important but most of us would prefer an ambulance made by Honda than a cheap Chinese knock-off. Trouble is that many service providers are run by technologists who don't get marketing. If I were a VC, I would hesitate to fund a company whose marketing is run by or dictated by an engineer (No nasty letters please, I am an engineer myself and you know I am right ;-).

But I digress. Differentiate yourself on features and services. If you think lowering the price to gain share makes sense, it doesn't. You will never be able to charge a premium price. A look at Volkswagen's Phaeton is shining example of a car company with an economy image trying to sell a premium brand. Volkswagen made a great car but no one wants a Volkswagen that costs $70,000. IT

Service Provider

The service provider division of Nortel has been working hard to bring Just in Time Communications to consumers through willing service provider partners. They are helping these partners sell converged services such as IM, voice, and video in one bundle. The goal is to give end users control of communications. They are in talks with Microsoft about working together to make this happen.

Their technology allows consumers to see presence information and also lets them set routing details. Nortel Executives cited research to me saying that 34 percent of customers would switch service providers if they had a single-number locate-me function. Selling JITC can gain you market share it seems.

They are working with Microsoft to be the backend client to the Microsoft front end. Apparently they tell me the Microsoft Istanbul client will make an appearance on *The*

Apprentice at some point in the future. This is good news as many of us will finally be able to show our spouses what we do for a living.

Another area they think is ripe for growth is the conference space where their technology allows meet-me conference bridges to be instantaneously created. This can save corporations millions on conference calling costs. Furthermore they think video calling is hot as they cite research that says over 50 percent of consumers would pay for video calls. While I think this number is high, I believe that Nortel is onto something here. They are trying to help service providers make their customers more productive and efficient and reduce churn. These are all noble causes that will help the bottom line of the service provider and the customer.

The Cable Division

Just a quick note on Nortel's cable solutions division… Nortel has a number of products for cable providers that allow a cable modem to be built into a WiFi access point with extended range. Such a device allows providers to light up a strip mall or hotspot with WiFi. A variety of antenna configurations allow your traffic to cover a neighborhood in a figure-8 fashion, point-to-point or hotspot which requires more circular coverage.

Thanks Allott

Allott Communications bill themselves as the communication specialists. They monitor your network, the applications and inspect their behavior. They can decode applications to layer 7 allowing IT managers to see what is happening in network. They can help you see not only VoIP but Web traffic, e-mail, movies, Oracle traffic, Citrix traffic... You get the idea.

Their products can actually block applications, and in the case of a denial of service attack, they can see where the attack is coming from and block it. They can tell you how much bandwidth users and applications are taking up and they can also set minimum and maximum session limits. Their equipment sits between the company and the Internet or intranet. When it comes to analyzing all the streams, applications and different types of traffic flow, there are all sorts of graphs to look at from real-time to historical.

The company points out that monitoring networks for VoIP is an ongoing challenge as a virus could spring up or someone could decide to host music files. A single computer in fact can change the dynamic of the network.

The company has enterprise and service provide customers as they have similar needs. They can help employees become more productive and they can allow service providers to create tiered services that can be charged for. Moreover they can help service providers balance oversubscription such as a WiFi hotspot and also allow application level prioritization.

Today's service providers need the flexibility afforded by such products allowing them to sell various products and services to their customers while enterprises can also benefit from knowing how their network looks today and in the past.

FreeScale Semiconductor

FreeScale Semiconductor is making inroads in the VoIP space and they tell me their market share against market leader TI is going up. They claim around 10–13 percent, which would make them number two in fact. Their latest DSPs are the MSC7118 and the MSC7119: a pair of higher powered DSPs that are part of the Starcore family. Zultys, for example, is using FreeScale DSPs in their MX250Enterprise Media Exchange and I know the Zultys products well so I though it was worth learning more about FreeScale. BTW, I was in the Silicon Valley headquarters of Zultys recently and they have a slew of products that blew me away. I can't discuss them now as I promised secrecy so check out my blog at tehrani.com for details.

I asked why the FreeScale devices are suited for VoIP and they told me you get lot of on-chip memory for the money and you get superior performance as well. You also get fieldBIST, which allows you to monitor memory, temperature, voltage and operating frequencies. You can do all of this remotely which makes the chips perfect for installing into VoIP devices that are always connected to a network. IT

VoIP Developer, IP Contact Center Summit On The Horizon

VoIP is changing the way the world communicates, and the world is taking note. If you have any intention of seizing the most promising telecom opportunity in recent memory, then I encourage you to attend the VoIP Developer Conference, August 2–3, in San Francisco, California.

Building on the success of last year's inaugural event, the second VoIP Developer Conference is the only chance you'll have in the United States this year to come learn how to quickly develop new VoIP applications that are in high demand.

You will learn what you need to know about today's hottest technologies: SIP, WiFi telephony, DSP development, Presence, Peer-to-Peer, Linux telephony, and more. And you'll be able to see hands-on demos and compare products from industry leaders like Intel, Aculab, SIP Foundry, Avaya, and more!

Registration is open at www.voipdeveloper.com. Don't delay! Register now. I look forward to seeing you in San Francisco.

I have been getting many requests for information on Speech-World collocated with IP Contact Center Summit in Dallas, TX May 24-26. You won't want to miss this inaugural event. Enterprises, contact centers, resellers, and OEMs need to stay on top of what is happing in the speech technologies market and of course IP contact centers are revolutionizing the way call centers work by saving money and increasing productivity. Check it out online at www.speech-world.com. IT

Letters

In response to Rich Tehrani's April 2005 Publisher's Outlook, "The Birth of a New Industry?"

Rich,

I'm glad to see you taking up the battle to change our woefully inadequate terminology to describe how people should communicate with others or be contacted by automated application services in a converging multi-modal communications environment.

You are on the right track by starting to focus on the relative priorities of the contact initiator vs. those of the recipient. These priorities will realistically start to come into alignment with the power of SIP which can enable end-to-end presence, availability, and last, but not least, modality management. It is the convergence of IP telephony and multi-modal messaging that will enable person -to-person communications to be both flexible and manageable. The "buddy list" for controlling contact relationships, as well as access "rules," are a starting point for dynamically controlling personal accessibility, but the dynamics of the real world will need more than such manual "programming" strategies.

Look at the complete failure of the voice mail "greeting" to satisfy the informational needs of a caller who wants to know how they can make immediate contact with recipient or at least know when they can expect a response if they leave a message. Most users record a standard, non-specific greeting. Whenever I ask an audience how many of them change their voice mail greeting at least once a day, I only see a handful of hands raised. Even then, the information is not accurate or adequate enough to be useful for a caller's immediate contact needs.

I have discussed the importance of the needs of contact initiators in articles of my own, pointing out that without the initiator's actions recipients would get nothing. With new multi-modal contact alternatives, however, the confusion that you complained about in your editorial is still a problem that I see being resolved through the still evolving facilities of SIP standards and what I have called "transmodal communication." That term describes the ability to quickly and easily shift from one modality of communication to another, as circumstances require.

Transmodal communications capability is not only useful when trying to establish contact with a person, but is also important even after one form of contact has already been made. Now that all forms of person-to-person communications are converging at both the IP network transport levels, as well as the communication devices level, people can initiate a contact at a message exchange level and escalate to a voice conversation (and vice versa) as the their needs and circumstances dictate. I proposed this perspective in a recent column that was published on your site (tmcnet.com/110.1) late last year.

The benefit of this kind of communications flexibility will be seen at the end user level (where it really counts), by enabling users to communicate more easily in whatever way works best for all parties, rather than guessing about what devices others have, what contact method to try first, and then being stuck with an initial communications modality that isn't effective. If the communication exchange suddenly requires a more efficient mode of communication, including conferencing in other people, then that should be a seamless and simple switch. The bottom line will be more, more successful contacts, as well as more "productive" use of time.

I don't know if my term, "transmodal communications" (it is kind of a mouthful!) will help fix the terminology problem you describe in your editorial, but the general reaction from experienced industry leaders has been favorable. Perhaps with a little more branding, it could catch up with the success of "VoIP!"

– Art Rosenberg

Correction

In our March 2005 issue, we inadvertently included some product specifications for the Toshiba Strata CTX28 in our write-up of the Strata CIX business communication system. For up-to-date information on the CIX, please visit http://www.toshiba.com/taistsd/.

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The 100% Microsoft®-based IP phone system designed *exclusively* for the Microsoft platform!

Both executives and IT management have come to rely on Microsoft solutions for managing their business applications and corporate data. Their objectives for a phone system, however, have historically been limited to a choice of proprietary systems, completely separated from their Microsoft business platform.

But now there's a 100% Microsoft-based IP Telephony solution that gives business executives what they want...and IT directors what they've been waiting for. Finally.

Enterprise Interaction Center® (EIC) is a fully pre-integrated, IP phone system that converges voice on the Microsoft Windows Server Platform, including Exchange Server for unified messaging, an Outlook® Telephony Console, and complete integrations with Microsoft® Business Solutions, such as Great Plains® and Microsoft CRM.

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- **Executives** get a solution that lowers costs, optimizes employee productivity, provides integrated communications for their mobile workforce, enhances their Microsoft investment, and helps them gain that all-important "competitive advantage."
- **IT Managers** simply get a solution that relieves headaches associated with separate proprietary voice and data networks. EIC is a complete Voice over IP communications solution based on Windows Server, using open SIP standards, with out-of-the-box business application integrations, and centrally administered with familiar Windows-based administration tools. *"Finally!"*

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Sony Delivers IPELA Product Lineup

Sony Electronics (quote - news - alert) recently announced the U.S. launch of its new IPELA line of IP-based communications products ranging from IP monitoring and recording devices to IP-ready videoconferencing systems.

Derived from the combination of IP and "bella" (Italian word for "beautiful"), IPELA signifies the arrival of business communications equipment that blends Sony's high-resolution imaging and audio technology with the universal reach of IP networks.

"IPELA represents our vision of expanded IP-based business communications — one that will make global visual communications as commonplace as a telephone call," said John Scarcella, president of Sony Electronics' Broadcast and Business Solutions Company.

IPELA products are designed to offer plug-and-play functionality for personal or small office use. They also come with options for supporting software and system integration capabilities that can create enterprise-level systems for large corporate environments. Sony has committed to intensive research and development for future IPELA technologies. Future roadmaps include the integration of high-definition technology into the IP communications experience.

According to Scarcella, the IPELA name is an example of Sony's commitment to positioning its products and solutions at the forefront of the growing IP communications market. This commitment has most recently been demonstrated by increased adherence to industry standards and collaboration with other industry leaders — one being Glowpoint, Inc. — to expand videoconferencing applications to virtually any location, and another illustrated by the integration of select conferencing systems with Cisco's CallManager 4.0 solution to provide simplified videoconference connections through the use of Cisco VoIP.

The first products to be labeled IPELA include the PCS-G70 large room videoconferencing system and the PCS-TL50 desktop videoconferencing system. Also showcased are network video monitoring cameras, including the SNC-RZ25N and P-Series MPEG-4 enabled models and the DF40 and DF70 mini-dome cameras.

Sony plans to introduce the new XIS Series of IP video monitoring cameras later this summer. This series includes the XIS-5100 wide-area monitoring camera that will cover a 160 degree x 15 degree area, and the XIS-10DC, a unique surveillance system for the industrial, government and aviation sectors with a 360-degree view and 100 megapixel resolution.

http://www.sony.com/ipela

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AudioCodes Announces Solutions For Carriers And Enterprises

AudioCodes (quote - news - alert) has announced its Mediant 1000 VoIP Media Gateway, a compact media gateway solution designed to interface between TDM and IP networks in enterprises or small-scale carrier locations. Incorporating AudioCodes' Voice over Packet technology, the Mediant 1000 is designed to enable rapid time-to-market and reliable, cost-effective deployment of next-generation networks. The Mediant 1000 is based on VoIPerfect, the architecture used in all AudioCodes' product lines ranging from voice over packet processors to high-density media gateway platforms.

The Mediant 1000 is designed for global deployment focusing on flexibility and cost optimized solutions for the SME market.

"Customers can scale up as their business grows, since the Mediant 1000 matches the density requirement for smaller locations while meeting service providers' demands for voice quality, product reliability, and scalability," said Lior Aldema, vice president of marketing at AudioCodes. "We have expanded our successful, existing line of Mediant gateways by developing a modular platform with both analog and digital interfaces that can be used as an integrated platform for OEM and partner applications. Deploying the Mediant 1000 allows our partners and OEMs to immediately address new opportunities in emerging markets and applications such as IP-PBX, fixed to mobile convergence, carrier hosted services and IP Centrex with a flexible, small-scale customizable solution."

The open platform on the Mediant 1000 offers partners the option to host their own applications (e.g., IP-PBX or call center application) using a powerful, low-power processor and hard disks to provide a complete solution within the Mediant 1000 chassis. The Mediant 1000 has enhanced hardware and software capabilities to ease its installation without requiring routing modifications in the PBX.

AudioCodes also announced new analog VoIP media gateways as additions to its MediaPack product line. The MediaPack 112, MediaPack 114, and MediaPack 118 are the latest members of AudioCodes analog media gateway product line addressing the requirements of service providers and enterprises seeking high-quality, cost-effective, analog media gateways.

The MediaPacks are designed to enable a wide range of applications including converged access, IP Centrex, fixed-mobile convergence, and next generation PBXs. Simultaneous mixed interfaces of FXS and FXO, survivability, guaranteed QoS, security, and standard billing interfaces are examples of additional features in the new hardware and software of the MediaPack product line. Integration with AudioCodes' Element Management System allows fast deployment and maintenance of large and complex networks for large scale service provider deployments. The MediaPack 112, 114, and 118 provide support for two, four, and eight ports of FXS and FXO connectivity.

http://www.audiocodes.com

Alpha Telecom Unveils New VoIP Gateways

Alpha Telecom (news - alert) has announced a cost-effective, high-performance, plug-and-play VoIP gateway designed to save most of the capital equipment cost and all of the ancillary retraining expense and time associated with a new network infrastructure.

Alpha Telecom's Arion series VoIP gateways are designed to migrate an enterprise's legacy telephones into the service provider's system, enabling the carrier to offer Centrex features over its IP backbone and to seamlessly provision, configure, and manage VoIP service without user intervention. The service provider merely ships the Arion units to the customer for plug-and-play attachment to individual phones by the enterprise's IT department or by the users themselves. The IT department can also deploy an Alpha Telecom element management system (EMS) for provisioning, mass configuration, troubleshooting, and mass upgrades. This EMS does not require individual user involvement, nor does it require them to load software on a PC.

"With our VoIP gateways, an enterprise can protect their capital investment by extending the life of their legacy system on a per-user or per-workgroup basis," said Alpha Telecom President Sid Sung.

An added benefit of the Arion series gateways is that organizations can enjoy the advantages of IP telephony with their existing digital speaker phones, where the quality is far superior to that of a dedicated VoIP speaker phone Moreover, because the plug-and-play VoIP gateways reside at the customer site, they support portability. This allows an enterprise's IT department to avoid carrier service-order changes and gives it control to balance user requirements for high functionally with the benefits of streamlined management and reduced facilities costs.

http://www.alpha-tele.com

Centrepoint TalkSwitch Enables Emergency Services Access

Centrepoint Technologies (news - alert) recently announced and underscored that its TalkSwitch IP PBX customers are always connected to traditional local emergency services. Because the TalkSwitch provides connections to the traditional public switched telephone network (PSTN) in addition to the VoIP network, customers are not required to sign up or activate any additional services in order to reach emergency assistance.

"It is absolutely critical that businesses and homes retain some level of connection to the traditional telephone network," said Jan Scheeren, president and CEO, Centrepoint Technologies. "Issues like access to 911, the ability to communicate during a power failure, robustness, and quality of service are all vital to telephone users. As a company, Centrepoint has recognized that VoIP has not yet evolved to the point where it can completely replace all PSTN-based services. That's why we've designed the TalkSwitch system to deliver the best of both worlds. We believe a combined PSTN/VoIP access system brings the most choice, value and reliability to customers.

"For VoIP service providers, issues with 911 access have been a long-standing challenge. Being able to integrate a solution like TalkSwitch with their services is a potential solution to that problem," continued Scheeren. "Centrepoint is currently undergoing rigorous interoperability testing with a number of leading service providers, and we're confident those tests will be successful. We're also involved in a number of initiatives, like the recently announced SIPconnect specification, that aim to improve the way manufacturers and service providers interoperate to provide added value to customers."

http://www.talkswitch.com

Vonexus Selects Quintum For IP Telephony Solution

Vonexus Inc. (news - alert), a wholly-owned subsidiary of Interactive Intelligence Inc., has signed an agreement enabling it to package its Microsoft-based IP PBX software with Quintum Technologies' SIP Tenor VoIP MultiPath switches, a partnership geared towards giving organizations increased reliability, improved quality of service, and simplified deployment.

The Vonexus IP PBX, Enterprise Interaction Center (EIC), is designed exclusively for Microsoft small to medium-sized business customers. EIC applications include SIP-based switching, auto-attendant, presence management, call routing and queuing, and Exchange-based unified messaging. Out-of-the-box screen-pop and IVR integration will be available in the second quarter for Microsoft Great Plains and MS CRM.

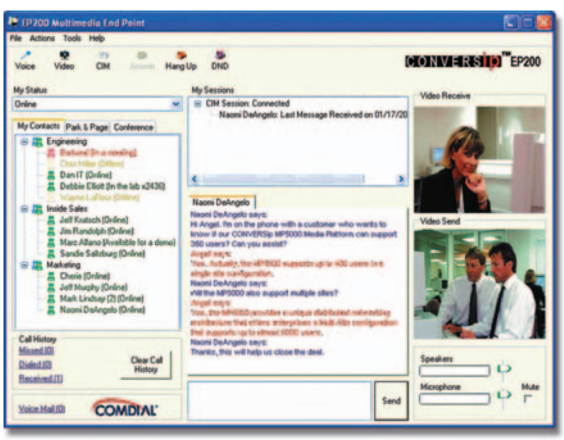
Quintum's SIP product, called the Tenor MultiPath Switch, is used to transfer calls between the public switched telephone network and IP-based local- or wide-area networks. It's designed to offer EIC customers maximum reliability and improved quality of service with patented technology that provides real-time backup integrated into the switch.

http://www.vonexus.com, **http://www.quintum.com**

Comdial's CONVERSip EP200 Streamlines Enterprise Communications

Comdial Corporation (quote - news - alert) recently announced its CONVERSip EP200 Multimedia Endpoint — a fully-integrated Windows XP-based solution designed to make business communication as simple as point-and-click.

The CONVERSip EP200 Multimedia Endpoint is a software application designed with the goal of adding advanced telephony functionality to every-day computers. The EP200, in combination with Comdial's CONVERSip MP5000 Media Platform, forms a complete enterprise communication solution that does not depend on complex integration of third-party hardware and software. Also, its built-in, secure corporate instant messaging feature does not rely on Internet-based consumer instant messaging services.



The CONVERSip EP200 offers traditional telephony features such as voice calling, park and retrieve, do not disturb, redial, conferencing, and paging, as well as advanced functionality such as video calling, presence management, and corporate single and multi-party instant messaging. In addition, the EP200 monitors incoming, dialed, and missed calls.

www.comdial.com

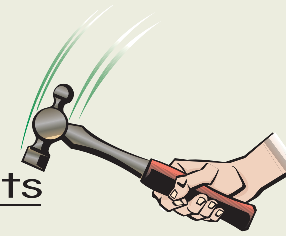
Orative Delivers Mobile Enterprise Applications

Orative (news - alert) has announced the commercial availability of Orative Enterprise Software, a client-server solution designed to improve the efficiency of voice communications by eliminating the delays caused by phone tag and allowing people to reach one another the first time.

Orative Enterprise Software is a client-server solution purchased and operated by enterprise customers. The software allows mobile phone users to easily and securely coordinate conversations, screen calls, collaborate with colleagues, and access personal and corporate contact information. The software allows users to gauge the importance of incoming calls by displaying details such as name, urgency and subject matter. It also delivers phone-based alerts like calendar reminders and conference notifications. A presence-aware phonebook makes it possible to determine at-a-glance if somebody is available to talk before actually placing a call.

Orative Enterprise Server and Orative Client Software are currently available. Server licensing starts at $4,999 per processor plus $2,495 for a Client Access License Starter Pack supporting 25 users.

http://www.orative.com



Quick Hits

Printing Company Selects FaxCore

FaxCore Corporation (news - alert), a provider of fax and document server automation solutions, announced that Modern Postcard, a provider of postcard, printing and mailing services, has selected FaxCore to automate fax processes in conjunction with Microsoft Exchange's Active Directory. The Carlsbad, CA-based printing company is also using FaxCore's fax tracking capabilities and reporting functionality to improve customer service.

http://www.faxcore.com

8x8 Enhances Virtual Office

8x8 (news - alert) announced it has added three new optional calling features to the Packet8 Virtual Office VoIP-hosted PBX service — Virtual Extensions, Metered Extensions and Toll Free Local Number Portability (LNP).

"8x8 is committed to making Virtual Office the most functional, flexible and cost-effective small business VoIP solution available," said 8x8 director of marketing Dave Immethun.

http://www.8x8.com

Customer Wins Demonstrate Security Of Protus Solution

Protus IP Solutions (news - alert) today announced that privacy firm, Nymity, Inc., and mortgage brokerage, Mortgage Intelligence, Inc., have selected the company's Internet-based fax solution to meet their needs for highly secure, efficient fax communications. Virtual Fax is an Internet-based fax solution designed to enable businesses to send and receive faxes using existing e-mail accounts or the Web.

http://www.protus.com

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Viseon Launches VisiFone With Multimedia

Viseon, Inc. (news - alert), has recently announced the multimedia version (MM) of its VisiFone Digital Home Telephone for VoIP. The VisiFone MM enables users to view personalized content and information automatically at any time they choose, on the large 10.4-inch color TFT/LCD screen in full motion video.

The VisiFone is a fully digital home telephone designed for VoIP services. It provides new features unique to VoIP, including digital wideband audio in the handset and speakerphone as well as TV-quality two-way video calling and on-screen VoIP feature management control. The VisiFone MM also empowers consumers to view new personalized content and information such as news, weather, sports and stock quotes. Consumers will be able to instantly view a variety of content from partners, including broadcast and cable TV networks and prominent web portals.

To aggregate content and sponsors for the VisiFone, Viseon previously announced the formation of ViseonMedia (VMN, LLC), a wholly owned subsidiary of Viseon, Inc. ViseonMedia has entered into agreements with various sponsors and marketing partners to deliver selected programming to VisiFone users throughout the globe. ViseonMedia will work directly with the VoIP carriers to develop content and information services for access by their VisiFone MM equipped subscribers.

http://www.viseon.com

XConnect Signs Leading VoIP Players To Global 'Plug & Peer' Network

XConnect (news - alert), which interconnects Voice over Broadband operators (VoBBs) to provide free calls and rich IP multi-media services to more end users, has launched in North America.

Its service is designed to make interconnection between VoBBs as straightforward, secure, and cost-effective as possible by overcoming the principal issues associated with VoIP peering, including numbering, interoperability and security.

As members of the XConnect Alliance, VoBBs can simply 'plug and peer' with other Alliance members over the XConnect Network to offer their customers free end-to-end VoIP calls to hundreds of thousands of end users. By keeping calls on the Internet and bypassing the public switched telephone network (PSTN), VoBBs can avoid termination charges and loss of VoIP call features.

XConnect has signed VoBB Yak Communications as a founder Alliance member and IP carrier iBasis as a founder Carrier member. International XConnect Alliance founder members include leading European VoBBs such as Telio, VozTelecom and Gossiptel. XConnect is also partnering with NexTone Communications, a provider of scalable session management of real-time IP services, to provide the first certified XConnect-Ready VoIP session border controller.

Eli Katz, founder and CEO at XConnect said: "XConnect's entry into one of the most dynamic markets for VoIP telephony will further speed the adoption of VoIP, by making its benefits available to more people. The XConnect Network brings the vision of an all-IP rival to the PSTN one step closer, where a rich array of multi-media IP services are delivered without legacy termination charges."

http://www.xconnect.net

VocalTec Adds Megaco Line Side Signaling Protocol

VocalTec (news - alert) has announced the expansion of its Essentra BAX to a wide variety of access interfaces, required for Class 5 migration to Next Generation Network (NGN) solutions. By incorporating Megaco line side signaling functionality to its Essentra BAX Broadband VoIP Access Platform, VocalTec provides carriers with the ability to continue supporting legacy telephones and fixed wireless networks, while seamlessly migrating to Next Generation Networks (NGN). The addition of the Megaco line side protocol facilitates a rapid, painless migration, offering carriers the option to keep existing consumer telephone connections and Digital Loop Carriers (DLC's) which concentrate these connections, unchanged.

"The addition of the Megaco line side protocol to our access solution reaffirms VocalTec's technological leadership in the Class 5 NGN market," commented Moti Suess, chief operating officer, VocalTec Communications Ltd. "This is further proof of our unwavering commitment to continue to provide the most complete and most scalable solutions in the market."

The Essentra BAX, currently being trialed by over 10 carriers, is part of VocalTec's expanded Essentra Product Suite, a set of open and highly focused VoIP products for next generation network (NGN) operators and service providers.
http://www.vocaltec.com

SOYO Introduces Z-Connect Gateways

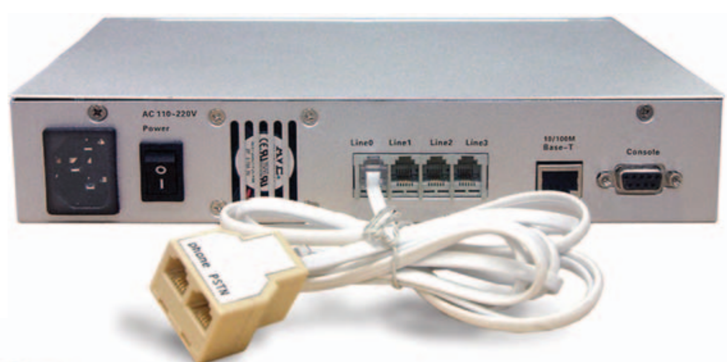
SOYO Group, Inc. (quote - news - alert), announced immediate availability of the Z-Connect Four-Port Single and Dual Mode Gateways, designed to deliver turnkey VoIP solutions to enable businesses to reduce telecommunications costs.

"Our Z-Connect VoIP Gateway solutions, designed for telecom carriers, VoIP wholesale dealers, and small and medium businesses, maximize the investments businesses have already made in their telecommunications infrastructure and data communications networks, as well as reduce ongoing long distance and international calling expenses. These new products are the perfect compliment to our existing line of Z-Connect consumer products and services," said Ming Chok, president and CEO of SOYO Group, Inc.

The Dual Mode Gateway (model N400DM) features an intelligent switching function that automatically determines which network — IP or PSTN — to use for making calls at the lowest rate, and also manages incoming calls from both networks via deployment of DID. The N400DM is compatible with a variety of networks, broadband access devices, and system configurations. Up to four analog phones or fax lines can be plugged into the Single Mode Gateway (Model N400S) for instant broadband capability.

"Users have the option of designating a separate DID telephone number for each line for receiving calls from any IP or PSTN telephone, and can also determine the order in which inbound calls are received," explained Mr. Chok.

The Z-Connect Four-Port Single Mode Gateway is available at $349. The Z-Connect 4-Port Dual Mode Gateway has a list price of $499.
http://www.soyogroup.com



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- 802.3AF POE Compatible

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Quick Hits

Intrado And TCI Form 9-1-1 Alliance

Intrado, Inc. (quote - news - alert), and Tel Control, Inc., (TCI) (news - alert), have announced plans to integrate TCI's advanced public safety answering point (PSAP) call center technology into the Intrado Intelligent Emergency Network (IEN), a robust emergency communications architecture designed to address the nation's expanding public safety needs.

The alliance brings together TCI's patented iPSAP solution, a fully-integrated, VoIP 9-1-1 call center technology, with the Intelligent Emergency Network, Intrado's next generation 9-1-1 infrastructure and services suite. The integrated solution should enable more effective communications between PSAPs, emergency management officials, and government agencies during times of crisis by facilitating interagency communication and providing emergency call takers and first responders with access to additional information sources to better respond to emergency 9-1-1 calls. TCI joins a growing list of companies who are providing new technology, services and content in support of the IEN architecture.

"TCI is committed to networking 9-1-1 centers to promote interoperability and facilitate greater communication between agencies. The combination of our innovative iPSAP solution and the Intrado Intelligent Emergency Network gives PSAPs new communications capabilities and services that will ultimately strengthen public safety for people across the nation," said Jeff Robertson, TCI CEO. "We look forward to working with Intrado to implement these new capabilities across our customer base."

The solution increases the 9-1-1 network's ability to respond to future communication technology changes without compromising the 9-1-1 network's critical requirements for security and redundancy. TCI will also work with Intrado to develop technology for public safety agencies that supports new communications and messaging capabilities enabled through the Intelligent Emergency Network architecture.

"The power of Intrado's network-based IEN solution combined with TCI's leading edge call handling solutions represents a significant step in the delivery of the next generation 9-1-1 network," said Stephen Meer, Intrado's chief technology officer. "Together, TCI and Intrado are working to evolve the emergency communications infrastructure to better support the country's changing public safety and homeland security requirements."

http://www.intrado.com
http://www.telcontrol.com

Voicenet Solutions Announces VoIP Services

Voicenet Solutions Limited (news - alert), a UK-based provider of premium Voice over IP and broadband business solutions to SMEs, today announced the launch of its VoIP services, VNconnect and VNgateway. Led by an experienced senior management team, which includes Non-Executive Chairman Godfrey Wilson, founder and Managing Director of MLL Telecom, and Chief Executive Officer David Crombie, a co-founder of Symphony Telecom Plc, Voicenet is dedicated to providing companies with next-generation telephony solutions that offer greater flexibility and improved communications, whilst significantly reducing costs.

Voicenet also announced an aggressive reseller recruitment program, whereby it intends to bring on board a number of strategic partners within the next year to resell its services.

Nconnect is Voicenet's IP Centrex managed solution for companies that want to benefit from the significant savings achievable by divesting themselves of PBXs and BT lines. IP handsets simply plug in to the LAN and calls are handled centrally via IP Centrex, providing advanced functionality without the requirement for a PBX. The service offers free on-net, between-branch calls, a unified dialing plan between remote locations, reduced calling rates to all destinations, and complete user online phone management tools.

For companies that want to continue to use their existing PBXs, but take advantage of the significant business benefits of VoIP, VNgateway seamlessly links existing PBXs to Voicenet's IP Centrex solution, offering free on-net calls, cost effective upgrades, improved scalability, and a unified dialing plan between remote locations.

http://www.voicenet-solutions.com

Labs2 Chooses Net Insight

Net Insight (news - alert) announced that Labs2 Group AB (Labs2) is using its Nimbra platform to build a Scandinavian IPTV triple-play network. The new network will be used to distribute TV, HDTV, Video-on-Demand, telephony and data services. Labs2, a Swedish broadband services developer and provider, will use the network to permit carriers to provide triple play services.

http://www.netinsight.net

PAETEC Selects Lucent VoIP Switching Solution

Lucent Technologies (quote - news - alert) and PAETEC Communications, Inc., have announced an agreement to deploy Lucent's VoIP solution for service providers, part of Lucent's Accelerate Next Generation Communications Solutions portfolio. PAETEC will use the solution to introduce VoIP services to enterprise and wholesale customers nationwide. Lucent Worldwide Services is providing engineering, installation, and network integration support to assist with the deployment.

http://www.paetec.com
http://www.lucent.com

Volo Expands Local Access Points To Over 5,000

Volo Communications (news - alert) announced that it has doubled the number of its local access rate centers in the past 120 days to over 5,000, now touching approximately 90 percent of the U.S. population. This move brings the company close to its goal of providing ubiquitous local access points into its nationwide VoiceOne network.

http://www.volocommunications.com

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Cedar Point, LongBoard In Cable VoIP Pact

LongBoard, Inc. (news -alert), and Cedar Point Communications, Inc. (news - alert), announced that they have demonstrated interoperability that will enable cable system operators to bring mobility to VoIP services over Hybrid-Fiber Coax (HFC) networks.

The joint solution is designed to allow operators to provide truly converged voice offerings — including a unified set of call features — that capitalize on the attributes of Cedar Point's SAFARI C3 Media Switching System and the LongBoard Mobility Application Platform (LMAP).

By integrating Fixed Mobile Convergence (FMC) with cable telephony solutions, cable operators can offer customers a new wireless service that seamlessly hands off calls across both local WiFi and public cellular networks, using one dual-mode phone and one phone number, as part of a quadruple play ("Home Run") of mobile voice, fixed voice, data, and video. The SAFARI C3/LMAP solution will enable cable operators offering mobility services to maintain control of their customers' calls and maximize their business opportunities with high-value applications and services. In addition, the solution capitalizes on SAFARI C3's existing integration with cable system operators' back office systems, accelerating time to market for converged services.

"The capabilities that were advance engineered into SAFARI C^3 will allow operators to simply and cost-effectively provide both WiFi and cellular mobility to their service bundles, using a single switch for both," said Dave Spear, executive vice president, strategy and market development for Cedar Point Communications. "A joint objective of our work with LongBoard is to enable our customers to move quickly from wireline voice delivery to a leadership position in the seamless mobility that is the future of telephony services."

"By working with Cedar Point, we can extend our expertise in FMC and address a large market of cable providers that want to benefit from the increasing use of wireless voice services," said Bill Leslie, founder and CTO of LongBoard. "Cable operators that can add wireless voice to their array of services will strengthen their ability to win new subscribers, and retain existing subscribers in an increasingly competitive communications market."

http://www.cedarpointcom.com

http://www.longboard.com

SpectraLink Wi-Fi Handsets Aid Community Hospital

SpectraLink Corp. (quote - news - alert), through the SBC family of companies, is helping Mission Community Hospital improve patient care with its NetLink i640 Wireless Telephones. The recently built 150-bed hospital in Panorama City, Calif., installed SpectraLink NetLink Wireless Telephones in its nursing and patient care areas to help nursing staff maintain immediate and direct contact with hospital physicians regardless of location.

Working through SBC companies, Mission Community Hospital installed 28 NetLink handsets in late 2004. The NetLink i640 Wireless Telephones offer nursing staff complete mobility while keeping them accessible to patients, physicians and others. Caregivers are now more responsive and efficient in meeting patient needs. The NetLink i640 Wireless Telephones also offer a unique push-to-talk (PTT) functionality that offers hospital staff an added sense of security by letting them broadcast to multiple handsets during emergency situations. Leveraging the hospital's existing Wi-Fi network, the NetLink i640 Wireless Telephones operate at low power levels minimizing risk of interference with sensitive medical equipment.

"The NetLink handsets are expected to help us save time and result in improved service to our patients," said Dr. Robert Thompson, the hospital's vice chief of staff and a pulmonary medicine physician. "With the new handsets, we expect much greater efficiencies in our process for the administration of both medications and necessary tests."

http://www.spectralink.com

Pronto Networks Demonstrates Premium SMS Capabilities

Pronto Networks (news - alert), has recently announced a new Premium SMS software module for its leading OSS platform for broadband wireless networks. The Premium SMS capabilities are enabled on Pronto's hosted Wi-Fi network for third-party service providers and allow both Cingular and Nextel mobile subscribers to log on at Pronto-managed Wi-Fi hotspots and metro-scale hot zones for 20 minutes by sending an SMS text message to the phone number 80211. A charge for $1.99 is billed directly on the mobile user's invoice for the service.

"The premium SMS service will provide mobile subscribers a new, convenient way to purchase Wi-Fi access for a short duration by using their cell phones rather than their credit cards," said Jasbir Singh, president and CEO of Pronto Networks. "The service will also provide additional revenue to carriers and hotspot operators offering this micropayment capability."

"Pronto's introduction of Premium SMS services demonstrates the company's continuous commitment to provide innovative, wireless applications and services," said Bruce Deer, President of SkyTel Networks, a subsidiary of MCI. "We look forward to working with them on future Wi-Fi projects."

Pronto Networks offers the Premium SMS software module as an extension to its core OSS platform for broadband wireless access. The module offers mobile operators considerable flexibility in defining key parameters of the service, such as how much time is permitted with each short code, how much to charge the mobile subscriber, and provisioning different short codes for the service. The module also supports revenue sharing between wholesale and retail operators. In addition to being offered as a software module, the Premium SMS capability is available to operators via a services agreement.

http://www.prontonetworks.com

Single-Chip Solutions From TI Drive VoWLAN Into Mainstream Mobile Phones

Texas Instruments (quote - news - alert) introduced the WiLink mobile Wireless LAN (mWLAN) platform, which includes single-chip solutions designed to drive Voice over WLAN (VoWLAN) into mainstream mobile phones. With single-chip solutions based on TI's DRP technology, the WiLink mWLAN platform enables seamless wireless connectivity to mobile devices. Comprised of hardware and software optimized for mobile phones, TI's WiLink solution is designed to provide consumers with on-the-go voice access over a WLAN or cellular network using their mobile phone.

"VoWLAN is becoming the emerging driver of WLAN technology integration into mobile phones and requires advanced technology for improved battery-life and talk-times," said Marc Cetto, general manager of TI's Mobile Connectivity Solutions Business. "As VoWLAN services become mainstream, TI's WiLink mobile WLAN platform will allow manufacturers to deliver a lower cost VoWLAN-enabled platform for consumer use. This becomes increasingly important to make the 'one phone' or 'universal phone' concept a reality. We expect consumers will soon be able to use only one phone for their mobile, office and home phones."

A complete hardware and software solution, the WiLink 4.0 mWLAN platform, TI's fourth-generation mWLAN solution, consists of two different options to meet a variety of marketplace needs. Manufacturers can choose between the TNETW1251 WiLink 4.0 802.11b/g single-chip or the TNETW1253 WiLink 4.0 802.11a/b/g single-chip depending on their product requirements. The platform includes a robust software package, the WiLink 4.X Software Development Kit (SDK) to deliver VoWLAN capabilities for mainstream mobile phones.

"VoWLAN penetration in homes and businesses is expected to propel the mobile WLAN market to new heights, offering consumers more choices in connectivity," said Allen Nogee, principal analyst, of In-Stat. "Texas Instruments has been a pioneer in the mobile WLAN space and is continuing that tradition by leveraging its single-chip expertise and DRP technology to provide their WiLink mobile WLAN solutions for cell phones."
http://www.ti.com

SerComm Selects Azimuth Test Platform

Azimuth Systems, Inc. (news - alert), announced that it has signed one of its largest international sales contracts to date with Taiwan-based SerComm Corporation, one of the leading wireless OEM vendors in Asia.

SerComm will test its family of wireless products, including access points and clients, with Azimuth's award-winning W-Series 802.11 WLAN test platform, a standardized platform for system level testing of 802.11 wireless access points, clients and other devices. The W-Series sets the standard for wireless data test solutions by allowing system vendors like SerComm to provide their customers with higher quality products in a shorter time-to-market.

"Asia is a tremendously significant market for Azimuth," said Ray Cronin, CEO of Azimuth Systems. "Because so much of the wireless component and system-level work is thriving in Asia, our relationship with SerComm validates our product's inherent capabilities and performance, and furthers our presence internationally."

Designed from the ground up as an off-the-shelf, wireless LAN test platform, Azimuth's W-Series systems are designed to provide the ability to configure an entire WLAN network in a bench top chassis designed for complete Radio Frequency (RF) isolation and control. The flexibility and programmability of the W-Series allows for the thorough evaluation of wireless LAN equipment under varying mobility conditions and traffic patterns, as well as precise analysis of the results. The system offers an ideal environment to perform software design validation and to test advanced wireless functionality and performance including the latest IEEE standards.

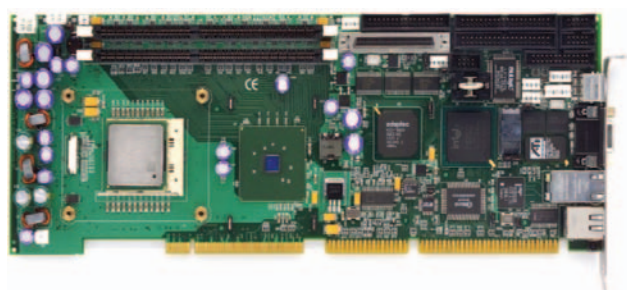
http://www.azimuthsystems.com

TRENTON Introduces T4R Single Board Computer

TRENTON Technology's (news - alert)T4R single board computer features a unique power control circuit that supports a +5V only operational mode and eliminates the need for an external +12V auxiliary power connection when using Intel Pentium 4 processors with speeds of 2.8GHz or less. This feature, combined with the T4R's ability to generate its own +3.3V power on board, makes the T4R an ideal choice as a drop-in SBC upgrade in PICMG 1.0 applications.

The Intel Pentium 4 processor and the Intel 82845-E chipset combination used on the T4R enables full PCI/ISA option card support. Trenton's T4R combines the Intel Pentium 4 processor's Intel NetBurst micro-architecture with 2GB of DDR memory, a 400/533MHz system bus, a 10/100Base-T Ethernet port and a 10/100/1000 Ethernet LAN port. The T4R merges the advanced feature set of the processor and chipset with long-life embedded product support in order to extend the computing system's value and performance.

Speeds for the T4R start at 2.0GHz using an Intel Celeron processor with a 400MHz system bus and continue through 2.4GHz and 2.8GHz using an Intel Pentium 4 processor with a 533MHz system bus. The Intel Pentium 4 processor has an L2 cache memory of 512K and the L2 cache of the Intel Celeron processor is 128K.



The T4R offers DDR200/266 memory, which is available in capacities up to 2GB. There are dual Ethernet LAN ports on the T4R. LAN Port 1 functions as a traditional 10/100Base-T port, while LAN Port 2 offers Gigabit (1Gb/s) connectivity in addition to the traditional 10/100Base-T interface. RJ-45 connectors, located on the I/O bracket, provide the mechanical interface to the Ethernet networks.

The T4R is available now with a list price of $1,299. Pricing and available processor speeds vary, so contact Trenton for the latest T4R pricing and CPU speed availability.

http://www.TrentonTechnology.com


"after 11 years, someone had to take a stand."
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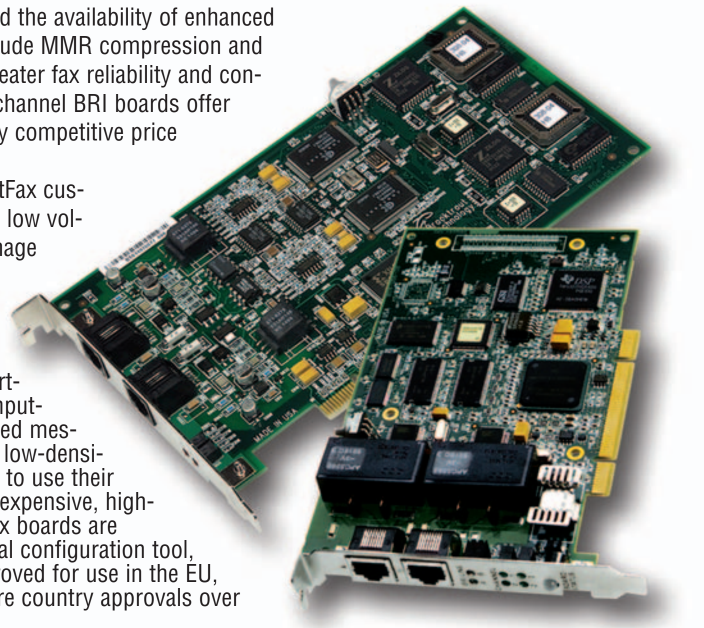
COVAD | voip
★★★★
"Outstanding performances!"
"TWO MOUSEPADS UP!"

Brooktrout Updates TruFax BRI Intelligent Fax Board



Brooktrout Technology, Inc., (quote - news -alert) has announced the availability of enhanced TruFax 250 and TruFax 450 BRI intelligent fax boards that now include MMR compression and Error Correction Mode (ECM), designed to give customers even greater fax reliability and connectivity. The new TruFax 250, two-channel, and TruFax 450, four-channel BRI boards offer small and medium business higher fax functionality at an extremely competitive price point.

"With the TruFax 250 and TruFax 450 BRI we can offer our RightFax customers an additional range of features for all small businesses and low volume fax customers who are looking for a dependable, easy-to-manage and cost-effective solution for their fax communications environment," said Matthew Brine, Vice President, International, Captaris.

The Brooktrout Technology TruFax Series is a line of intelligent fax boards that offers small to medium-sized businesses and departmental workgroups dependable fax capabilities for a variety of computer-based fax applications, such as network fax, fax broadcast, unified messaging, and business process automation. The TruFax BRI offers a low-density fax product at an economical price point. This allows customers to use their existing BRI connection, so that they don't have to pay for a more expensive, higher density PRI digital lines for their faxing requirements. The TruFax boards are Microsoft Windows plug and play compliant and contain a graphical configuration tool, which allows for easy installation. The TruFax BRI is currently approved for use in the EU, Iceland, Norway and Switzerland and Brooktrout plans to have more country approvals over the next year.

http://www.brooktrout.com

Kasenna Offers 64-Bit Video Server Platforms

Kasenna (news - alert) has announced a technology breakthrough that sets a new standard for next-generation VOD systems while lowering their cost of ownership. By developing its own high-performance I/O enhancements to the Linux kernel and capitalizing on advances in Intel's processor and system architecture, Kasenna has ported its MediaBase platforms to 64-bit processing architecture. The key here is that the solution is now able to deliver better space utilization, higher switch port densities and increased reliability for cable MSOs and Telcos deploying advanced on-demand services.

Using the 64-bit capability of Intel Xeon processors, Kasenna MediaBase achieved 3.2 Gbps of streaming throughput and (853 streams at 3.75 Mbps) from a dual-CPU, three rack-unit (3RU) disk-based server. The Kasenna system delivered 5.4 Gbps from a 1RU RAM-based server. At the CableLabs specification of 3.75 Mbps for streaming video, this amounts to 1,440 streams per 1RU server or 60,480 streams per 42RU rack.. Using a hybrid system of disk-based clusters and RAM-based clusters and its patent-pending Stream Clustering technology, Kasenna has achieved breakthrough densities in streaming performance.

http://www.kasenna.com

AudioCodes Licenses Telchemy's VQmon Technology

Telchemy announced that AudioCodes has licensed their VQmon/EP (End Point) for direct integration into AudioCodes' media gateway products. The manageability provided by Telchemy's VQmon/EP is designed to enable AudioCodes customers — service providers and enterprises — to monitor, diagnose, and troubleshoot complex problems in real- time, ensuring high service quality and availability for next-generation services.

Network operators and system administrators use Telchemy's VQmon technology to detect, monitor and resolve call quality and network related problems for networked multimedia services, including Voice and Video over IP, IP Centrex, 3G Cellular, Voice over WLAN, and streaming audio/video. VQmon provides listening and conversational call quality metrics in both R factor and MOS formats as well as detailed diagnostic information, giving network managers both high level metrics and the ability to drill down to identify specific problems.

http://www.telchemy.com

Elma Intros Portable PXI Chassis

Elma Electronic, Inc., (news - alert) has announced a new PXI chassis for instrumentation systems. The eight-slot Type 32 PXI chassis is designed to offer a portable and flexible format for easy prototyping and development.

The Type 32 PXI chassis is 17" high x 10.5" wide. It features 2 x 5.25" HH devices and a 350 W plug-in power supply with an option for dual redundancy. The unit is specifically designed for portable use and is geared toward protection from the wear-and-tear these units often face. The vinyl clad aluminum covers are scratch-resistant. An ergonomic handle allows for easy and comfortable portability. Also, rubber-soled feet prevent the chassis from marring or scratching desktops.

The PXI chassis is compliant to PXI Specification R2.0. The chassis also has rear I/O and system monitoring options.

http://www.elma.com

Quick Hits

Acterna Delivers Over 3,000 Testers

Acterna (quote - news - alert) announced that it delivered over 3,000 of its HST-3000 handheld service testers to more than six Tier 1 carriers in North America and Europe during the past six months. Widespread adoption of the HST includes the delivery of handheld field testers designed specifically to enable the rollout of IP Video service.

http://www.acterna.com

Texas Instruments Licenses Impulsesoft's Bluetooth

Impulsesoft (news - alert) announced that Texas Instruments (quote - news - alert) has licensed Impulsesoft's iWALTZ multimedia platform for Microsoft Windows Mobile. Targeted at handset manufacturers worldwide, the iWALTZ platform from Impulsesoft has been optimized on TI's Bluetooth single-chip technology enabling mobile phone manufacturers to rapidly bring to market a complete Bluetooth stereo multimedia solution.

http://www.impulsesoft.com
http://www.ti.com

RADCOM Enhances Network Consulting

RADCOM (quote -news - alert) announced an enhancement to its Network Consultant solution. The new feature enhancement is designed to provide the call-flow visibility carriers need to troubleshoot inter-technology handoffs between 2.5 and 3G network-roaming partners.

The Network Consultant is an umbrella application that provides an integrated, end-to-end view of the entire network.

http://www.radcom.com

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Your Network is Complicated Enough

Upgrading to Power over Ethernet Shouldn't Make you Lose Sleep

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Zultys Launches New Family Of IP Phones

Zultys Technologies (news - alert) announced immediate availability of its new family of ZIP phones. This range of phones is built entirely on open standards and runs on a highly stable real time Linux operating system. The phones are compatible with any IP telephony system using SIP (Session Initiation Protocol), providing customers with greater flexibility in their telephony system choices. Support for features such as power over Ethernet (PoE), line-rate Ethernet switching, voice encryption, and conferencing also make the phones ideal for a number of applications and environments.

"This new range of phones clearly sets us apart from our competitors," said Patrick Ferriter, VP of Product Marketing at Zultys. "We can supply phones separate from our range of IP PBXs or with our IP PBXs. We have received high praises from our customers for the look, feel, and sound quality of these new phones."

Available in black or white, the ZIP 2 family is priced in single quantities at between $150 and $230 depending on model.

http://www.zultys.com

SIP Industry Bolstered By IMS Developments

New research by Venture Development Corporation (VDC) (news - alert), estimates worldwide markets for SIP infrastructure and software will exceed $5.5 billion by 2007. This represents a compound annual growth rate of 36.1 percent between 2003 and 2007.

Factors contributing to SIP market growth include:

- Replacement of legacy infrastructure.
- Adoption of VoIP services.
- New applications.
- IP Multimedia Subsystem.

According to Chad Hard, VDC's Telecom/Datacom Practice Director, IMS adoption will drive SIP growth, with initial trials through 2006 and more significant deployments in 2007. "The acceptance of IMS by both wireline and wireless carriers solidifies SIP's role as the protocol of choice for all network operators. Most carrier-focused vendors will repackage their existing SIP offerings to cater to this rapidly emerging market."
http://www.vdc-corp.com

Mercom To Support SIP Call Recording

Mercom Systems, Inc. (news - alert), announced that its Audiolog call recording server now supports call recording for Session Initiation Protocol (SIP) environments. Mercom's SIP-interface VoIP recorder provides cost-effective software-only and turnkey recording solutions for SIP-enabled IP telephony contact centers. SIP enables contact centers to operate as a network-based service, feed calls to remote agents located anywhere, unify communication channels, and achieve substantial cost savings.

"Mercom is moving quickly to meet the recording and quality monitoring needs of a changing IP telephony and contact center market," said Bob Jagendorf, director of marketing and sales at Mercom. "By offering SIP recording, we are demonstrating how Mercom continues to be on the cutting-edge, keeping up with the newest and best technologies that are available to call centers."

Mercom's Audiolog systems are installed in nearly 6,000 of the world's leading call centers, mission-critical public safety and government agencies, utilities, and financial institutions.

http://www.mercom.com

Veraz Launches Call Center Network Compression Solution

Veraz Networks (news - alert) announced its new Call Center Network Compression solution for multi-city and multi-country call centers. When operating point-to-point communications between onshore and offshore contact centers, the high cost of international bandwidth reduces the advantage of the global call center and Business Process Outsourcing (BPO) market even with recently proposed International Private Line Circuit (IPLC) rate reductions. Veraz's compression solution eliminates this obstacle by reducing the bandwidth requirements by more than 90 percent thus increasing voice capacity on IPLCs by over 1,000 percent. This is designed to provide service providers and enterprise customers with an expected Return on Investment (ROI) of less than six months.

This latest Call Center Network Compression solution is based on Veraz's I-Gate 4000 family of media gateways that deliver 12:1 voice compression, enabling major savings in bandwidth costs while maintaining toll-quality voice. The I-Gate 4000 family of media gateways incorporates Veraz DSP technology based upon more than 12-plus years of in-house DSP experience and code development. The I-Gate 4000 was specifically designed to minimize bandwidth requirements by implementing toll quality compression algorithms such as G.729, while maintaining the level of voice quality that carriers demand.

"Our customers with multi-city and multi-country call centers should benefit greatly from the significant costs savings that this Call Center Compression solution provides," said Amit Chawla, executive vice president, global marketing at Veraz Networks. "We are continuing our trend of offering a consistent string of cutting-edge solutions for our international customers that leapfrog the competition."

http://www.veraznetworks.com

MCI And Tellme To Deliver Internet-Based Contact Center Solutions

MCI, Inc. (quote - news - alert), announced an agreement with Tellme Networks, Inc., to offer Internet-based contact center solutions to large enterprise customers in financial, insurance and healthcare industries, and the government sector.

The two companies are now jointly selling customer self-service solutions running on Tellme's network and integrated with MCI's routing platform for live agent support. This combination means MCI customers can take advantage of Tellme's Internet-based and carrier-grade network to power their customer service phone numbers.

"Businesses are embracing Internet-based contact center solutions to enhance their customer service operations," said Nancy Gofus, senior vice president of MCI Product Management. "Our relationship with Tellme complements more than 10 years of MCI contact center experience by expanding our portfolio to deliver powerful applications that address the requirements of some of our largest customers."

By adding Tellme's capabilities to its suite of hosted voice services, MCI is able to offer large enterprise customers in select markets an end-to-end contact center solution and expedite the roll-out of more sophisticated Internet-based services. Customers can achieve efficiencies by leveraging their existing Web infrastructure to answer calls, increase automation using speech recognition technology as the primary, self service interface and provide callers with a personalized customer experience that also reflects a company's distinct brand.

"We are at the very beginning of what is possible on the phone when you power the conversation with Internet data. We are pleased to be working with MCI to offer these applications to a broad base of enterprise clients and to support MCI transport on our network," said David Weiden, senior vice president of marketing at Tellme.

http://www.mci.com
http://www.tellme.com

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By Marc Robins

VoIP's 911 Dilemma

The current suit filed by the Texas State Attorney General against Vonage over 911 emergency calling services has cast a pall over the broadband telephony marketplace. And even though the suit appears to be without merit — given the fact that Vonage strongly advises customers to opt-in to its 911 service and clearly discloses its inherent shortcomings, and because of the lack of any regulatory mandate to provide emergency services — it also has the potential to not only keep prospective broadband telephony customers on the sidelines but to also usher in a new wave of regulatory fervor.

For those of you unfamiliar with the case, here's what started it all. In early February, The John family in Texas suffered a home break-in and assault by two armed robbers. Joyce John was upstairs when she suddenly heard gunshots and her parents screaming for the teenager to call 911. When she dialed from the family Vonage line, she heard the message: "Stop. You must dial 911 from another telephone. 911 is not available from this telephone line. No emergency personnel will be dispatched."

Unfortunately, Joyce's parents had not registered for the optional 911 service on the line, and Mr. John (who along with his wife fortunately survived the incident) claims he had no idea that he had to sign up for emergency service in order to activate a 911 call.

In swoops Texas Attorney General Greg Abbott, announcing, "This family's moment of crisis signals a dire need for Vonage to clearly communicate to its Internet telephone customers that 911 access may not be available to them. This is not just about bad customer service; it's a matter of life and death." The Attorney General then went about filing a lawsuit under the Texas Deceptive Trade Practices Act, asking a state court to order Vonage (news - alert) to stop saying it offers "911 calling," and to change its marketing to highlight the steps a customer needs to take for emergency service. The suit also asks for Vonage to pay $20,000 per violation.

Although there is no mention of emergency calling issues or procedures on the Vonage Web site home page (an omission that can and should be corrected), there is a complete description of the available 911 emergency service on the site associated with each calling plan Vonage offers. In addition, Vonage alerts new customers twice during signup that they should register for 911 service, and sends e-mail reminders and pop up alerts during account login for customers that sign up for a calling plan but neglect to register with the emergency service.

Clearly, in my opinion, the problem here is not due to a lack of disclosure — but is due instead to the many differences between traditional 911 service and the type of emergency services currently being offered via VoIP (define - news - alert)— and the ingrained expectations of the calling populace raised on circuit-switched PSTN (define - news - alert).

Rather than relying on ANI, or caller ID, to help pinpoint the location of an emergency call, Vonage (and practically every VoIP service provider that offers emergency calling services) dictates that 911 must first be activated through a registration process that involves the customer telling the service provider the physical location of the line. After that, the 911 calls go to a general access line at a Public Safety Answering Point (PSAP). This is different from the 911 Emergency Response Center where traditional 911 calls go, and also requires that the caller must state the nature of the emergency, including location and telephone number, as Public Safety Answering Point (PSAP) personnel will not have this information on hand. Other drawbacks are that power and broadband service outages will prevent 911 calls from being made, and if you travel with your Vonage adapter to another location, you must update your account with the new address information.

Granted, these are not the makings of an optimal emergency service offering, but it is certainly better than no emergency service at all. In order to diffuse the inevitable backlash from regulators and the general public, there are a few proactive things broadband telephony service providers can and should do:

1. Be completely forthcoming about the drawbacks of current 911 service offerings, and provide a link to a full description of the service right from the Web site home page.
2. Make the emergency service registration mandatory, not optional. Refuse service to anyone who is not willing or too lazy to sign up for it.
3. For a provider that hasn't yet adopted emergency services, this needs to be a number one priority.
4. Continue to push forward on newer and alternative methods of providing emergency service.

Unless providers effectively address and communicate the issues surrounding 911, the likelihood is that many consumers will be hesitant to adopt broadband telephony service as their primary line of communications — especially if they are a one-line household — and broadband telephony will be relegated to secondary-line status. IT

Marc Robins is Chief Evangelism Officer of Robins Consulting Group, which offers an array of services to the IP telephony industry. He has been involved in the telecommunications industry as a reporter and analyst, trade show producer and publisher, and marketing executive and consultant for more than 24 years. For more information, call RCG at 718-548-7245 or e-mail robinsconsult@optonline.net.

By Tony Rybczynski

Making Work Something You Do, Not Somewhere You Go

The key to business success in moving to convergence is to identify a very specific first phase that delivers real business benefit. Just because you can do something, doesn't mean you should! For many enterprises and government agencies, the low-hanging fruit of the convergence tree is making the increasingly mobile and distributed workforce more effective, and saving money along the way.

Targeting important user segments, such as executives, sales, IT, and telecommuters with unified communications capabilities is an important element in adopting a virtual enterprise model. This model extends connectivity from traditional data VPN (define - news -alert) remote access in two primary ways: by enriching the communication capabilities with voice and multimedia; and by integrating presence across a broad range of user devices (e.g., phones, PCs, laptops, and PDAs) and activities. Unified communications embraces multimedia conferencing, collaboration, presence, session control, and IM services. It gives workers the ability to communicate more effectively with others — anywhere, anytime. In the longer term, these always-on integrated communications systems will interface into enterprise business applications, such as workflow systems, customer service, and supply chain management, leading to a fundamental transformation of how businesses and government work.

Unified communications solutions change the way business is conducted across the virtual enterprise by enabling work to be truly an activity and not a destination. SIP (the Session Initiation Protocol) is pivotal to this open standards-based system and is the protocol glue across these environments. SIP (define - news -alert) is a signaling and control protocol for initiating sessions between and among users. It is independent of the media being used. Presence information is central to this environment and is associated with a person's activities, whether being on a voice call, participating in a multimedia session, actively using a desktop or business application, or being in a particular location. Some vendors have embraced SIP across their business telephony, rich media, and contact center solutions, since SIP-driven communication solutions connect users over any device, anytime, anywhere, and anyhow.

Unified communications gives workers the ability to communicate more effectively with others — anywhere, anytime.

The User Is In Control

Unified communications is changing how users work. It's a real-time, always on integrated model. "Real-time" speeds decision-making and closure. "Always-on" provides secure converged access regardless of location — in the office, WiFi conference, public hot spot, hotel room, or at home. "Integrated" provides a single user interface and a seamless user experience. Roaming between WiFi (define - news -alert) and second- and third-generation public cell networks will extend this connectivity to truly mobile environments. In this new world, users have a high degree of control as to how their presence is communicated (or hidden), and how incoming calls are handled based on caller, media used and time of day.

The Business Benefits

Nortel has equipped over 20,000 of its mobile employees with multimedia clients for their laptops. This resulted in an overall 52 percent reduction in calling card, long distance and telephone charges for executives, 42 percent reduction for sales people, and 90 percent for telecommuters. Instead of incurring traditional phone charges, these workers can have secure access to the converged enterprise network by simply using DSL or cable modems at home and wired or wireless Ethernet connections in hotels, airports, and WiFi hot spots. The IT group demonstrated a payback of only eight months by adding unified communications solutions and IP enabling PBXs. In addition, moving hosted audio-conferencing solutions to an in-house implementation resulted in a payback of four months.

Through this example, you can see some of the hard benefits of equipping mobile users with unified communications capabilities. Specifically, these include: elimination of home second lines, substantial decreases in long distance, 1-800, calling card and possibly cell charges. A dramatic reduction in audio, video, and Web conferencing fees is also possible if these services are being outsourced to a service provider. The soft benefits include faster decision making (e.g., shorter sales cycle and problem resolution times), more effective group project execution (e.g., less wasted time, higher quality outputs and/or faster time to completion); and higher employee satisfaction by eliminating time wasted through ineffective communications.

Unify your communications, mobilize your employees, and watch your business grow. **IT**

Tony Rybczynski is Director of Strategic Enterprise Technologies at Nortel. He has over 30 years experience in the application of packet network technology. For more information, please visit http://www.nortel.com.

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By John Cimko

"Keep What You Use"

The FCC launched a proceeding last September to explore steps to speed the delivery of broadband and other spectrum-based services to rural America.

If you flip back to page 80 of the notice of proposed rulemaking, you find a very interesting idea.

The FCC asked for comment on whether it should adopt a "keep what you use" licensing rule to promote services in rural areas. Under this approach (which already applies to cellular service), if a carrier holds a license that includes rural service areas, and the carrier fails to use the spectrum to bring service to those areas within specified time periods, the carrier must return the unused spectrum to the FCC, which would then make the spectrum available to other carriers either via competitive bidding or by using non-auction mechanisms.

The FCC noted that "the rapid provision of broadband and other wireless services to rural areas is of critical importance in accomplishing our statutory and public policy objectives." And the FCC worried that "there may be instances where . . . market-based policies may not be adequate to promote access to spectrum in rural areas."

That's a nice way of saying that "market-based policies" enable some carriers holding spectrum licenses that include rural areas to concentrate on serving the more lucrative urban markets but then sit on the spectrum they hold in rural markets, choosing not to construct facilities to serve rural customers.

The result is that people living in less populated areas (where cable and DSL (define - news - alert) services, and sometimes even basic telephone service, may not be available) don't receive wireless services. The spectrum licensed by the FCC to provide broadband and other spectrum-based services instead lies fallow.

As the FCC recognized, the market may be failing to give carriers incentives to use their wireless licenses to bring broadband and other services to rural customers. Many of these licensees have concluded that it's not cost effective to serve rural areas. In addition, the carriers generally aren't interested in leasing or otherwise making their rural spectrum available to other carriers. An industry association representing rural carriers commented in the FCC proceeding that its members have described spectrum leasing opportunities from large carriers as "virtually nonexistent."

The strategy of the licensees seems to be to hold onto the spectrum, even though they're not doing anything with it, and then wait to see if using the spectrum to provide services in rural areas might become profitable at some time in the future. In the meantime, of course, if you live in a rural area, you lose.

But now the FCC is considering whether to discourage this spectrum warehousing by changing carriers' incentives. If the FCC adopts a "keep what you use" rule, three things could happen.

First, carriers holding spectrum covering rural areas might take a closer look at the economics of bringing service to those areas, because the alternative would be to surrender the unused spectrum to the FCC.

Second, these carriers also might become more receptive to leasing or otherwise making their unused spectrum available to other carriers on reasonable terms.

And third, if unused spectrum does revert to the FCC, this would give carriers with greater incentives to serve rural areas a chance to acquire the spectrum and deliver broadband and other services.

But if the current licensees don't have incentives to provide service in rural areas, why would it be any different for other carriers? A commenter in the FCC proceeding explains it this way. Large carrier licensees don't provide service in less densely populated markets because they want to avoid the risk of low returns on their investment. But small rural carriers have different incentives. They often are based in the areas they serve. Many of these carriers are organized as community-owned cooperatives. This means they pay attention to community needs, and they have an incentive to use spectrum to meet these needs.

The FCC did acknowledge that a "keep what you use rule" could have some drawbacks. For example, some carriers have argued that, regardless of what rural carriers say, if the economics do not support providing service in a rural area, then pulling back the spectrum and giving it to a new carrier won't ensure that service actually is delivered. Other carriers argue that the proposed rule could deflate spectrum values and dampen wireless service investment. The rule also could lead to uneconomic construction by carriers seeking to "save" their licensed spectrum, and could result in steep administrative and legal costs (such as costs associated with figuring out whether spectrum is being "used" by licensees).

But, as the FCC rulemaking moves forward, there is a strong case that the agency should adopt its "keep what you use" proposal. For one thing, since the spectrum is a publicly owned resource that the FCC is charged with managing for the benefit of the American public, the agency always should pursue policies that discourage letting spectrum lie fallow. The new rule also would help promote the FCC's objective of bringing broadband services to all Americans, including those living in rural areas. **IT**

John Cimko served for fifteen years at the FCC, and currently practices law at Greenberg Traurig LLP in Washington, D.C. The views expressed are solely those of the author and should not be attributed to his firm or its clients. For additional information, visit the firm's Web site at http://www.gtlaw.com.

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By Hunter Newby

Recognize The Rise Of VoIP Peering

There are patterns in history that if recognized, can result in substantial gain for the trained eye. For our time in history and for our industry, perhaps the patterns to recognize are the rise of VoIP Peering and the success of meeting points.

Throughout history, waterways served as conduits for commerce. Along the banks of intersecting rivers, trading posts were established. Paths to and from these posts were worn into roads. Critical mass was reached and the metropolis was born. Other marketplaces in the vicinity could not compete. The prime metropolis became a flourishing hub of the community. Cities, railroad stations, airports… all follow the same great "meeting point" patterns. And as we turn to the technology industry, we see the same: core voice and transport networks connecting in prime meet-me areas like 60 Hudson Street in New York and 1 Wilshire Blvd. in Los Angeles and public Internet networks connecting in the major Internet peering points in Ashburn, VA and Palo Alto, CA. Once critical mass is reached, these geographical locations become fixed. There is no need or reason (and in some cases, no ability) to move these locations. The economics take hold of the consumer. "This is where you go to buy XYZ products at the best price." The key is volume and traffic flow, which creates a competitive pricing environment.

The reality of ENUM (define - news - alert) VoIP (define - news - alert) Peering is the same. In order for the maximum value to be extracted from the technology, there should be one central resource for the information. It is possible that there will be several ENUM registries representing different types of services, functions, and users groups, but even within those sub-sections, the greatest return will come from eliminating disparity. Disparity is what causes delay and drives up prices. The key benefits to the direct users of a central ENUM database are significant cost savings, efficiency and accountability for the service. It is very comparable to a major airport. The user knows that's where to go because there are the greatest number of choices, highest probability of the lowest cost, and an organization with a clear mission that can answer questions about the service.

The accountability benefits are essential for a "meet-point" type of operation — and possibly even more so for a global phone system database to function properly. The accountability benefits translate into responsibilities for the database operator; they include quality of service, accuracy of user information, security of the information and privacy of the users. The users entrust these critical elements in return for certainty. The certainty has three parts: that the users will be protected, that the service will function properly and that the users are getting the greatest possible economic gain from the collective system. For the early adopters, this is a calculated risk; for the followers, it becomes a necessity.

Usually critical mass not only implies a group of users of a particular service, but also that intangible trust. They believe the service will work, they rely on it and it becomes an increasingly important part of their business. Once the trust in the community is established, the foundation is built for growth. To oversimplify the cycle: service creation, limited awareness, early adopters, trust, mass awareness, and then mass adoption.

Currently this pattern can be seen developing with a couple of the established ENUM registries. The technology exists, has been productized, and is being used. The early adopters have moved in and are taking advantage of free inter-network calling with the other early adopters. They are also taking advantage of every new number that enters the database as soon as it registers. They realize that it is in their best interests to get as many other networks to participate as possible. This will raise the probability of an on-net call and lower their operating cost. In the short time span of a few months, the viral word will spread and the market itself will begin to draw the line between those that are in and those that are out. Once the line becomes public knowledge, the masses will be enlightened.

It's at that point that the economics of a dominant ENUM registry cannot be denied or ignored. As an overall business enabler, the registry will become an essential, proven benefit and have a built-in ROI. The larger the registry gets, the faster the ROI is reached. Once the brave, early adopters have waded in to the waters, the followers will have little choice but to jump right in behind them. To not do so would be like trying to get a cheap, direct flight to London from the East Hampton airport.

Disparity is what causes delay and drives up prices.

Although we are in the midst of significant change and there is a good deal of uncertainty in the process, we should all feel secure that everything that is happening now has happened before in some other form. The pattern now manifests itself in to another rudimentary, man-made system. That system is inefficient and the laws of nature and evolution are setting in to change its course. Rest assured that we are all part of this progress and that the end result will be a better system than the one we have today. IT

Hunter Newby is chief strategy officer at telx. For more information, please visit http://www.telx.com.

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Part IV: To Host Or Not To Host… That Is The Question

By Shawn Lewis

The overall market for VoIP (define - news - alert) services is rapidly increasing both in the U.S. and worldwide, and VoIP has been described as the next "killer application" for telecommunications services. Consumers are demanding more services, more features and more choices, at a lower cost. The true killer application is not VoIP, but rather how service providers will bundle and host these advanced services. IP Centrex features are in high-demand by the end user, and service providers face the challenge of hosting the services either at their facilities or at the customer's premises.

Hosted IP Centrex services enable service providers to immediately provide Primary Branch Exchange (PBX) functionality for phone service in SOHO (define - news -alert), small enterprise, and medium enterprise business environments. Phone service is provided over IP connectivity from the customer to the Internet, a provider's private IP network, or VPN, and all standard features of PBX functionality, as well as more advanced calling features are provided to the customer.

Hosted IP Centrex provides an immediate cost savings to both the provider and the customer, reducing CAPEX (define - news -alert) costs, sustained OPEX (define - news -alert)costs and additional telecommunications and other service charges immediately. In many cases, Hosted IP Centrex also adds flexibility and feature sets not found in many smaller Key or PBX systems. In this column, we will address:

- Operational overhead considerations.
- Speed to market.
- High-demand features.

There are many benefits to hosting application services through wholesale providers versus investing in your own application platform.

So, we find ourselves asking… Should we host or own our IP Centrex and PBX applications?

One answer is a low-cost, high-quality hosted solution that speeds time to market, assures carrier-grade quality, and enables early market entry without the CAPEX requirements associated with investing in your own applications.

Operational Overhead Considerations

Should you invest in and manage your own high-end feature servers?

The rapid adoption of IP-based applications such as IP-Centrex, PBX, and unified messaging in the SMB (define - news -alert)and enterprise market has sent carriers and service providers scurrying to provide these features as layered applications to their voice offering. There have been two traditional approaches:

- Purchase high-end application servers from key vendors and offer the applications directly;
- Choose a wholesale vendor to host the features for you and eliminate the CAPEX.

Both of these are viable options depending on the carrier's or service provider's overall business plan as well as their financial and operational wherewithal. The market however has pointed out that there are limitations to hosting your own applications, primarily from an operational perspective. Level 3's announcement this January that they are getting out of the hosted application business is a testament that even the largest carriers can experience operational challenges when hosting their own applications. It was reported that the operational overhead was not in line with their core business objectives.

Speed To Market

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Gaining market share early is critical.

The VoIP market is expanding at a fever pace. Carriers and service providers are racing to get into the business to capture their market share as quickly as possible. From a historical perspective, the first provider that acquires customers en masse with a new service offering that bundles sticky applications has the greatest chance of long-term market dominance. With numerous wholesale partners available in the market today, the best answer for most service providers is to consider going to market immediately by purchasing the VoIP application services from a wholesale provider, at least initially. Once they have their product defined and begin a large scale roll-out, they can then analyze whether it makes sense to buy their own applications to address their new customers.

The market has pointed out that there are limitations to hosting your own applications.

High-Demand Features

Staying ahead of the technology curve is a must.

Delivering VoIP applications requires technical expertise in software applications. Enhanced features and application services in the VoIP world are radically different than in the circuit-switched TDM world. The technology landscape is rapidly changing, making many of the technologies outdated before they have provided a positive return on investment. One of the biggest value propositions in using a wholesale hosted platform is that the hosting vendor is responsible for developing and/or providing the feature upgrades. In a hosted applications environment, the technology refresh costs are the responsibility of the hosting vendor, thus eliminating the investment required to stay ahead of the technology curve.

That said, using a wholesale hosted applications platform provides a significant business proposition because it enables carriers and service providers to have a market advantage by offering the latest features without the investment. At the end of the day, carriers and service providers have to ask themselves whether or not they are better off spending their financial and operational resources on acquiring profitable customers or are they better off building every aspect of their network and application offering. As the market has demonstrated, size doesn't matter… Operational capabilities do!

Shawn Lewis is the CEO of Volo Communications, a wholesale provider of advanced voice and data services and applications including broadband VoIP service. Mr. Lewis also wrote the first two patents for softswitch and media gateway technologies. For more information please visit http://www.volocommunications.com.

INTERNET TELEPHONY EDITORS' CHOICE AWARD 2005

VegaStream Vega400

VegaStream, Inc.
10445 Pacific Center Court
San Diego, CA 92121
Tel. US: 1 877-834-4470
Tel. UK: +44 1344 784900
Web: http://www.vegastream.com

List Price: $4,000 to $12,000

When you choose a VoIP gateway there are three important items to consider including ease of use and installation, standards support (SIP and/or H.323, codecs…), and voice quality with minimal latency. The Vega 400 from VegaStream Ltd. fits these requirements to a "T" and in fact exceeds them. TMC Labs examined the Vega 400, a gateway that supports four E1/T1 interfaces and can be configured to support one to 120 VoIP channels (four E1s).

In fact, the Vega 400 has a very distinguishing feature that differentiates it from many of its competitors. That is, many competing gateways can also do 120 channels, but only using the G.711 codec (not other codecs). On the other hand, the Vega 400 can do 120 channels using G.711 but also 120 channels using, say the G.729 or G.723.1 codecs. Due to the DSP or processor requirements many competing VoIP gateways can only do perhaps 90 out of 120 channels using the G.729 codec. Similarly, the 400 offers support for T.38 faxing and it supports the full 30/24 channels per E1/T1 of fax simultaneously where as some other gateways only do a few.

DSPs (define - news - alert) are on PCMCIA cards, which slide in the back of the chassis. Upgrades (i.e., from one T1 to four T1's) can be done without opening the unit. This flexibility allows customers to start with a low-density gateway at a low cost, and add capacity as needed.

When you first boot the Vega 400 you can select either a SIP or H.323 stack which is a really nice feature. Many other solutions require that you manually download separate firmware to install a different VoIP stack. Currently, you cannot split the channels with some configured as SIP and some as H.323, though it's probably very rare these days that a customer hasn't standardized on one or the other.

Upgrading the firmware is very easy to do — you simply specify the TFTP (define - news - alert) server to upgrade firmware. When upgrading the firmware it downloads to a second partition then it swaps it with the active partition. This way if there is a problem you can switch back to old firmware. On a related note, configuring and updating multiple boxes is very easy to do. You can just edit the config files (text format), change the IP address of the unit, and then download to multiple boxes. Thus, upgrading multiple boxes was a fast and painless procedure.

RATINGS (0–5)
Installation: 5
Documentation: N/A
Features: 4.75
GUI: 4.75
Overall: A

The Vega 400 comes with a plethora of features that TMC Labs liked. For instance, it supports SNMP (define - news - alert), and you can map ports (8081) for HTTP administrative access. We liked its powerful dial plan, which had wildcards, ranges, etc., making it easy to define your prefixes and routes. We were able to for example define "99" prefix as a SIP-based call. It also supports RFC 2833 (RTP payload for DTMF digits) and SIP INFO method. The SIP INFO method allows for the carrying of session related control information that is generated during a session. One example of such session control information is ISUP (define - news - alert) and ISDN (define - news - alert) signaling messages used to control telephony call services. Importantly, it supports Mode 2 and can interoperate with Cisco equipment which according to VegaStream (news - alert) is not 100 percent SIP standard.

After configuring the Vega 400 via the Web interface (Figure 1) we set the SIP registrar to a VoIP gateway located in VegaStream's remote offices. Next, we connected the Vega 400 to a Gordon Kapes T1/E1 930 Simulator using a standard T1 cable. We should point out that the Gordon Kapes unit can act both as a T1/E1 simulator as

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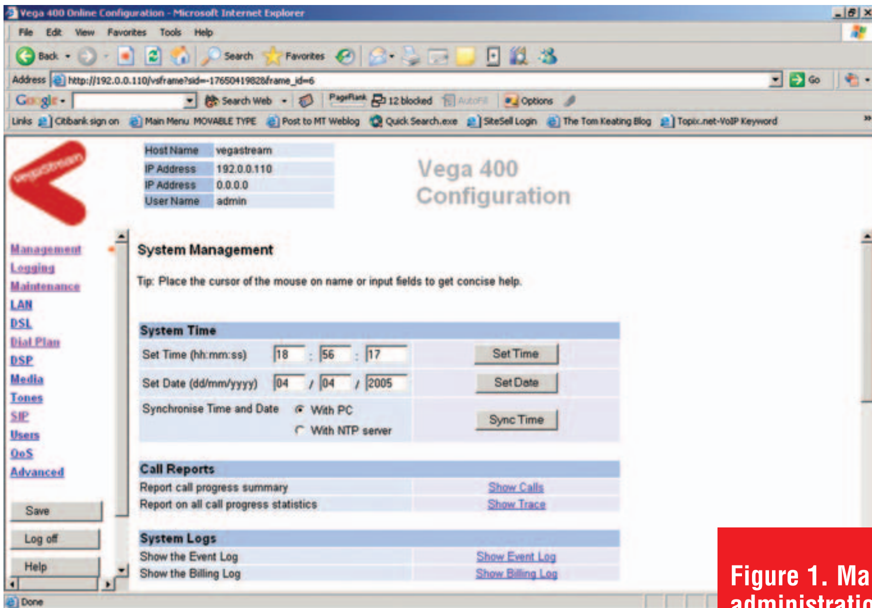

Figure 1. Main administration interface.

well as a simulated PBX, which was useful because the Vega 400 can interface with a PBX and/or the PSTN (TE or NT) on all four interfaces. We then hooked up an analog phone to one of the analog ports on the Gordon Kapes and then we initiated an outbound call from the analog phone that was routed across the T1 connection to the Vega 400. The Vega 400 determined the dialing plan from the touch tones we pressed (an outbound call) and it routed the call across the Internet to VegaStream's office where the call hopped off onto the PSTN. The voice quality was excellent and latency was negligible. Next, we tried the reverse. We had the remote VegaStream office call our Vega 400 gateway (over the Internet/IP) which then routed the call to the Gordon Kapes box and which was set to automatically ring the analog port. We accepted the call and once again the voice quality was superb.

The Vega 400 passed all our tests with flying colors, so we proceeded to examine some of the other features. One important one is that it has two profiles — one for voice and one for data/fax. Thus, you can turn off VAD and echo cancellation for data/fax profile so they are not affected by these mechanisms. Another important feature is that you can prioritize codecs by figuring out which codecs the remote gateway has in common and then choosing the preferred codec. The Vega 400 has SIP over UDP and TCP support, advanced SIP parameters for fine tuning, and you can set one physical network port for VoIP traffic and the second network port for management, which is important for security reasons.

It supports QoS by enabling DiffServ or configuring a VLAN. The unit has QoS statistics and you can set thresholds that notify you via SNMP, log file, or syslog server. The log file can be saved and loaded onto a second box which is useful for tech support. You can also see the default setting followed by new setting which is also useful for tech support to determine what was changed by the user

Other features/specs:

- 4 E1/T1 interfaces
- 15 - 120 VoIP calls. Field-upgradeable through expansion modules
- G.711, G.729 and G.723.1
- SIP and H.323 VoIP signaling
- N. American and ETSI PRI signaling, QSIG, T1 CAS

(The full listing of features can be seen at www.vegastream.com/pdfs/Vega-400_Brochure.pdf.)

Conclusion

As previously mentioned, when choosing a VoIP gateway you want to choose one that follows all the appropriate standards, is scalable, easy to administer, and has good security. Also, when comparing several gateways, don't limit your comparison to just price — make sure that when it says it supports 120 ports that you get a full 120 ports of VoIP using any codec. Otherwise, you are overpaying for ports that are not fully functional. The Vega 400 gives you fully functional ports so you get exactly what you would expect. TMC Labs was impressed with the performance, voice quality, ease of administration, and adherence to appropriate standards by the Vega 400 and we would not hesitate to recommend it. IT

PROS	CONS
Ease of use & installation	Can't run SIP and H.323 stacks simultaneously
Standards support	(minor) Web GUI uses frames
Good voice quality with minimal latency	

U.S. Navy Steams To Victory In "VoIP Becomes Reality" Contest

Quintum Technologies (news - alert), and TMC (news - alert) , publishers of Internet Telephony magazine, announced at the Spring Internet Telephony Conference and EXPO that the U.S. Navy was selected as the winner of their joint "VoIP Becomes Reality" contest.

The contest submission from Callis Goodrich, an Engineer in the SPAWAR Systems (news - alert) Center in San Diego, described how the Automated Digital Network System (ADNS) allowed the Navy to use Quintum's Tenor switches to eliminate outmoded, inefficient time division multiplexing (TDM) voice network connections and to converge both voice and data communications over a IP-based satellite link.

The Winner

The U.S. Navy ADNS program conducted an experiment to use VoIP to carry ship-to-shore telephone calls over a 64 kbps or 128 kbps Inmarsat satellite link. The goal of the experiment was to leverage the existing base of telephony gear on the ship (traditional phones connected to PBXs) while using VoIP to more efficiently use the limited WAN bandwidth.

The previous design used TDM (define - news - alert) to carry circuit-switched voice trunks between the ship and shore PBXs, and bandwidth reserved for voice circuits was not available for other applications when the phones were not in use. A Quintum Tenor A400 VoIP (define - news - alert) gateway was used on the ship and an A800 at the shore, and each was connected to the existing PBX (define - news - alert) via FXS (define - news - alert)or FXO lines. No call manager was used because each line trunk on the ship corresponded to a matching port on the shore PBX, and no changes were need in the PBX call plan. This was referred to as *VoIP trunking*, with VoIP only used for carrying separate voice trunks across the WAN. No end-to-end (or "native") VoIP handsets were used in this design.

Quality of Service guarantees for low-jitter transport across the limited bandwidth, high-delay link were provided by Cisco IP routers, using low-latency queuing and link fragmentation and interleaving (LLQ/LFI). A G.723 codec was used for regular telephone calls and a G.726(16) codec was used for secure telephone calls and faxes. A new feature in the Tenor known as "modem relay" allowed all voice ports to be configured the same, with the initial low-bandwidth G.723 codec initiating the call and then automatically switching to the G.726 codec when it sensed the initiation of a secure call or fax. The Tenor allowed adjustment of the VoIP payload in order to overcome the penalty of added IP overhead caused by the requirement to connect the VoIP gateways through a VPN device. The increased processing delay for the large payload was negligible compared to the large delay through the satellite.

Call quality was judged to be as good as that in the previous TDM design and secure telephone calls were established with greater reliability than had been observed under the TDM design. Use of an IP transport allowed multiple satellite links to carry the voice calls, which improved overall reliability from the TDM transport, which was tied to a specific satellite link. Overall link efficiency was improved through the recovered use of bandwidth that was previously idle for dedicated voice circuits.

According to Mr. Goodrich, "We are pleased with having won the contest, in one way I consider it an acknowledgement that the design constraints for military environments are unique and require creative thinking to adapt commercial technology that was not specifically developed for that environment. Using VoIP technology is consistent with the direction given from Dept. of Defense leaders to move towards a converged IPv6 network environment. Efficient use of scarce network bandwidth is a top priority with limited defense resources, and I expect to see continued use of VoIP technology in military communications."

Honorable Mentions

Quintum and TMC also selected four entries as Honorable Mentions. Syed Muhammad Zakaria Al-Hady of the Learn Foundation, a charitable foundation in Bangladesh, implemented several internet cafés through which people could make phone calls from remote villages to the U.K., the U.S. and Middle East.

LEARN Foundation (Sylhet)

According to Mr. Zakaria Al-Hady's contest submission, LEARN Foundation is a non-profit charity trust. Its mission is to impart ICT training to underserved communities in rural Sylhet, build market-driven wireless networks and develop sustainable businesses for the rural communities over the networks.

Over the last seven years LEARN has trained several thousand rural students in the greater Sylhet districts in the use of the Internet and other relevant ICTs and motivated rural communities to adopt applications over wireless networks to be applied in their day to day lives. Sylhet has hundreds of thousands of expatriates living in the U.K., U.S. and the Middle East. They primarily communicate with their relatives abroad over VoIP. The outbound traffic is channeled from about 1,000 rural telephone shops (RTS) in the district (an average of three villages per call shop). The volume of outbound traffic is about 50,000 minutes per day from Sunamganj district and about 200,000 minutes/day from all of greater Sylhet.

Since there is no Internet in rural Sylhet, the prevailing VoIP access from the RTS to the VoIP Internet gateway located in Sylhet city is largely via mobile GSM (define - news - alert) networks or via long-distance (30–70km) cordless telephones. The bulk of the outbound voice traffic originate from the villages and they are aggregated over these Cordless and GSM networks from as far away as 70 km and converged into VoIP Internet gateway infrastructure located in Sylhet city.

LEARN's strategy is to bring the VSAT (define - news - alert) gateway infrastructure near the villages and to establish a call carrying infrastructure based on VoIP. H.323 has been chosen as the technology for this purpose, while SIP is being considered for future porting. The goal is to serve rural telecom on GSM and PSTN for communication to and from UK, USA, and Middle East by providing:

a. direct access for RTS to the satellite gateway bypassing the city based Telephone shops;

b. its own Internet VoIP gateway infrastructure between UK, USA, and Middle East with Sylhet cutting international carrier costs for the village people.

Stratgez (Dubai)

The next was earned by Faraz Ghani of Strategez, LLC in Dubai, which sought to implement a ship to shore VoIP system between oil exploration ships in the Gulf of Oman and the local offices. They connected calls via IP to the GSM and fixed line subscribers, locally in the United Arab Emirates, and internationally to the PSTN (define - news - alert) from the PABX (define - news - alert).

The requirements were to provide communications over IP to an oil exploration ship belonging to one of the leading oil exploration companies sailing in the Gulf of Oman for oil exploration by the government of Sultanate of Oman in their water territories.

The requirements were to provide the following:

1. Connect the PABX Extensions on the ship to the PABX located at the exploration company's regional headquarters in Dubai, United Arab Emirates via IP to enable extension calling from the ship to mainland offices and vice versa.

2. Connect (via IP) the onboard

extensions to local GSM and fixed line subscribers within UAE and internationally using the regional headquarters' PSTN CO lines connected to the PABX.

3. Connect (via IP) the onboard PC workstation to the regional office network and share all network applications of the company, including providing Internet access using the DSL links available in the regional office to enable onboard staff to browse the Web and access e-mail.

DataTEK (Romania)

The third Honorable Mention was submitted by Gabriel Prodanescu, of DataTEK Group, in Romania. DataTEK implemented Internet Cafes with a multi-tiered partner structure, using Vox Carrier, Inc. as the service provider. DataTEK proposed the use of Dialexia's Dial-Café software bundled with Quintum gateways. DataTEK distributes their solution in Romania and neighboring countries. The Dial-Café solution is designed to enable Cyber shop owners and ITSPs to generate revenue by adding voice services to their basic Internet offering. Dial-Café is designed to support SIP and H.323 gateways, with the gateway functioning as an interface between the callers' phone and the IP network. Dial Café is a PC-based client server application that allows owners to set up routing/rate tables and to control customer calls. Calls are monitored on a real-time billing platform. It has a built in IVR for prepaid calling card usage and allows operators to generate bills and receipts..

Creative Technologies (Philippines)

The fourth Honorable Mention was awarded to the submission from Virgilio Almeda of Creative Technologies in the Philippines. His company implemented a VoIP network in a bank with IP phones and Cisco's Call Manager. The design provided direct dialing from the IP phones in the home office to the analog phones in branch locations, hop-off calling in the remote offices, and fax transmissions.

"We were amazed at the level of interest that this contest generated," said Chuck Rutledge, Vice President of Marketing for Quintum Technologies. "We had a very difficult time selecting the winner and Honorable Mentions from the many submissions we received. The variety of applications submitted really reflects how far VoIP technology has advanced." IT

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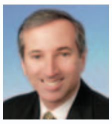

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Hosted VoIP Over UNE-L:

A Promising UNE-P Exit Strategy

A clear path is open to lower wholesale costs, make peace with incumbents, and develop affordable digital broadband facilities — without the truck rolls.

All the king's lawyers and lobbyists appear unable to put UNE-P (define - news - alert) back together again. Yet a viable alternative to the Unbundled Network Element Platform (UNE-P) approach may lie with a twist on everybody's darling: voice over IP (VoIP) services.

The twist — hosted VoIP delivered over the incumbent's unbundled network element-loop, or UNE-L — would require minimal competitive local exchange carrier (CLEC) (define - news - alert) facilities investment in data center VoIP facilities while still leaving copper loop maintenance, customer equipment, and truck roll costs to the RBOC. It would answer regulator calls for such facilities-based investment, and it would promise continued customers for RBOC wholesale businesses.

Broad-Reaching Benefits With Limited Challenges

In short, at workable costs, CLECs can host VoIP and advanced IP multimedia services at a centralized data center and use inexpensive voice gateways to convert those services to traditional time division multiplexed (TDM) switched circuits from the data center to the customer. They, in turn, get advanced VoIP (define - news - alert)and IP multimedia services over their old black phone or standard PBX (define - news - alert). This hosted VoIP model would enable CLECs to boost revenues through delivery of both digital voice and broadband data services while shedding dependence on RBOC Class 5 TDM switches. In the bargain, the CLEC gains control over its own fate.

Here's how it can work:

(1) Utilize Unbundled Network Element-Loop (UNE-L) regulations to buy incumbent copper loops wholesale, typically for under $10 per month.

(2) Deploy or purchase wholesale media gateways inexpensively to convert from VoIP to switched circuits over those cheap local loops.

(3) Provide VoIP application servers to host and manage basic and next-generation voice services at the data center.

(4) Deploy or purchase wholesale digital subscriber line access multiplexer (DSLAMs) ports to deliver both voice *and* high-speed data over DSL (define - news - alert).

Further, it is a strategy that benefits every critical constituency:

CLECs

Through a combination of economical hosted VoIP facilities, low-cost UNE-L access, and broadband DSL delivery, CLECs will gain the ability to offer "double-play" broadband data and advanced voice services enabled by VoIP, neither of which were facilitated under UNE-P. Further, the CLECs' required investment in facilities and associated staff will be limited to the central office and largely free of last-

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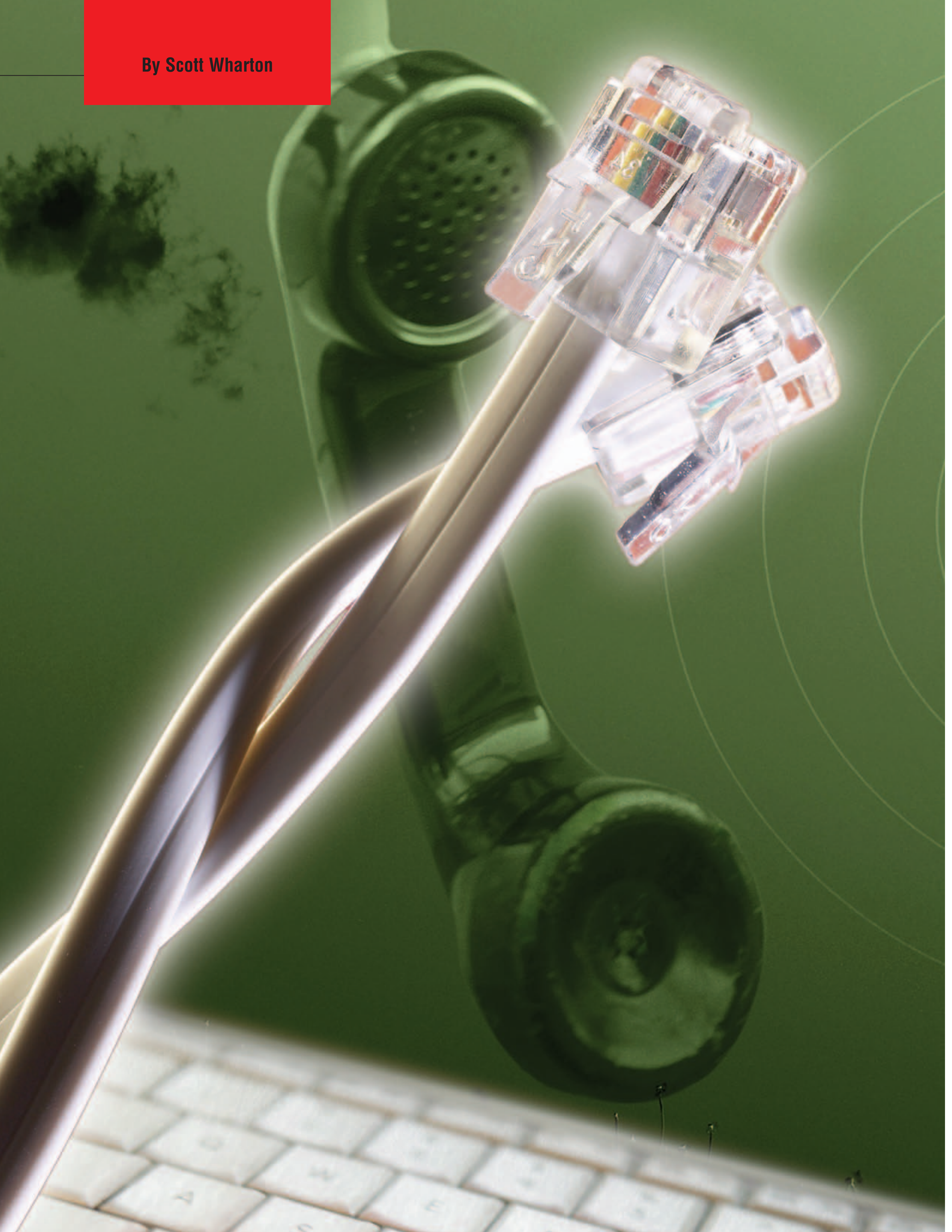
By Scott Wharton



mile network and customer equipment maintenance.

End Users

At the same time, business and residential buyers of telecom services will cheer the retention of voice and data provider choices beyond the Bells and cable operators, and many will gladly hang on to their standard phones over which they'll enjoy a host of next-generation services beyond dial tone.

Regulatory Agencies

The U.S. Congress, the White House and the FCC will likely welcome a way to maintain incentives for competitors in addition to Bell and cable operators, as well a means to encourage facilities investment at a realistic cost.

RBOCs

The RBOCs themselves may enjoy taking credit for enabling facilities-based competition without having to unbundle all their services as well as the loop. Indeed, the model may also prove an attractive approach for the RBOCs (define - news - alert) themselves.

Vendors

Likewise, other interested parties could gain from a hosted VoIP over UNE-L model. For vendors, it could break a logjam of uncertainty that has held back innovative next-generation last-mile services and sales of the equipment required to deliver them. And venture capitalists and financial institutions may again smile on a sustainable competitive provider model based on capital assets and undistorted pricing and earnings.

A facilities-based approach based on hosted VoIP and UNE-L will not be without challenges. It will require additional technological skill sets in the central office and collocation hotels, as well as in data services. Incumbents could push back with higher UNE-L pricing.

Nevertheless, no present alternative offers lower costs or a higher probability of success for all parties concerned. With the new ability to offer both advanced telephony and high-speed data services, CLECs can expect greater ARPU (average revenue per user) and a competitive advantage through service differentiation.

UNE-P Withdrawal

Since the passage of the 1996 Telecom Act, AT&T (quote - news - alert) , MCI (quote - news - alert) and scores of smaller competitors have used UNE-P to buy discounted TDM (define - news - alert) voice switch ports and services, including last-mile access to customers, from incumbent telcos. The promise of UNE-P lay with the notion that CLECs could bootstrap investment in their own facilities over time by using a non-facilities-based approach — discounted wholesale access — to immediately build customers and a revenue base, thereby raising the money to invest in switches and access technologies over time. The formula certainly has created an alternative to the incumbent for many customers: CLECs claim some 20 million UNE-P access lines. However, so long as the number of UNE-P access lines continued to grow without parallel growth in direct sales, UNE-P looked more and more like perpetual resale, and less and less like a wholesale business.

There's also an addictive element to UNE-P. It allows competitors to cling to an arguably artificial price structure while perpetually putting off the day when they direct profits into facilities of their own. In this sense, while the courts may have vacated UNE-P rate rules based on legal issues, the FCC and Congress appear to have abandoned the UNE-P regime at least in part because it has not produced the broad strategic aim of the Telecom Act: facilities-based competition.

"There's no need to pronounce UNE-P dead," says Dana Frix, a Washington, D.C.-based telecom attorney and co-chair of Chadbourne & Parke LLP's telecommunications and technology practice. "It has been an uncertain thing for two years, and for carriers uncertainty is tantamount to death, because how can you plan or persuade investors?"

CLECs can expect greater ARPU and a competitive advantage through service differentiation.

Under these circumstances, large CLECs including AT&T and MCI have all but abandoned UNE-P as an avenue to local market entry, turning instead to partnerships with another group of facilities owners: broadband cable operators. Their un-endorsement of UNE-P deflates the legitimacy and momentum of the approach considerably. More critically, at a time when bundled broadband access and next-gen IP services are becoming competitive table stakes, a carrier continuing to ignore the limitations of UNE-P — it enables no voice messaging or message waiting indicator, no data/broadband, no advanced features — amounts to a prescription for demise.

Given this downward spiral, it has probably become inevitable that the remaining independent CLECs are preparing to accept one form or another of painful withdrawal from their dependence on facilities-less operations.

The Lowest Cost, Highest Payback Avenue To Facilities

With so many millions of UNE-P lines established, the stakes are high, and the options are limited.

Staying the course looks grim. With no more government restraints on the incumbents' wholesale rates, UNE-P carriers could face a squeeze from approximately 50 percent currently to 15 percent margins typically found in resale. Alternatively, investing in one's

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own Class 5 TDM voice switches would be a commitment to a strategy that has already proven itself inadequate. Another alternative — adopting an end-to-end VoIP approach — also appears daunting, as it would comprise a major departure from UNE-P operator skill sets, which typically do not entail maintenance of outside plant, network power, and equipment inside of thousands of customer premises.

The hosted VoIP over UNE-L approach solves most of these challenges.

- It brings the benefits of VoIP to customers, without the hassles of bringing VoIP all the way to the customer premises.
- It provides customers with the simple user experience provided by traditional TDM service.
- By utilizing the UNE loop, it solves VoIP's most vexing challenges: backup power, E-911 and lawful intercept (CALEA).
- It provides the lowest-cost bridge to bona fide facilities ownership by limiting the facilities to the VoIP platform in the data center and by affording an incremental pay-as-you-grow facilities investment path that can start with gateways as small as eight lines each.
- By placing the IP-to-circuit gateway in the data center, it avoids placing and maintaining a terminal adapter at every customer site for voice services and limits required expertise to within the CO's boundaries.
- It paves the way for years of VoIP-based IP multimedia innovation beyond standard enhanced voice features.
- It opens a path to the most desirable premium customers by enabling the "double-play" of voice-plus-high-speed data by leveraging DSL wholesalers such as Covad Communications and DSL.net who can offer already deployed DSLAM ports — and who will see value-add in hosted VoIP.
- For data services, it requires only simple DSL modem CPE that can be drop-shipped and maintained remotely.
- It greatly improves the carrier's business case by using much higher ARPU to justify facilities investment.

"There's no question that competitive carriers must develop some kind of combination of facilities and Internet Protocol," says Frix. "Customer demand for local and long-distance service remains extraordinarily high, even after what has amounted to a depression in the market, so the question is how you provision it. It's certainly a truism that, to the degree that you control the facilities, you are generally better off than if you are only reselling. You gain more resources to move with changing technologies and control your own fate."

Other barriers may present themselves. Not all customers will want advanced features or high-speed data access. For many CLECs, their established UNE-P customers won't neatly match up geographically with new central office deployments. These are not insurmountable challenges. VoIP deployment architectures end the days of big iron through a highly distributed design, enabling targeted deployment of the most promising geographies. Customers who do want premium VoIP and the double-play may well amount to the 20 percent in the well known 80/20 rule that most revenue comes from a minority constituency. That premium minority can be migrated to hosted VoIP first, as the CLEC focuses on COs with the most lines.

"Combining the assets of hosted VoIP with UNE-L is an intriguing option for competitive carriers struggling to find a stable business plan in an always-shifting and uncertain regulatory climate," says Joe McGarvey, senior analyst with Current Analysis. "While this particular variation on the VoIP theme may not strike the right chord for all CLECs or their subscribers, it demonstrates the versatility and flexibility associated with packet-based voice technology."

Purists will argue it's circuit switched to the home, instead of VoIP. But who will care? The copper's there. The network power is there. And so are all the CLASS features, as well as untold advanced applications to come. In time it may prove itself an effective new model for VoIP providers as well as a new model for UNE-P providers.

It's worth considering: a hosted VoIP model offers a viable facilities-based option that will allow UNE-P providers to become facilities based while maintaining access to the incumbent's already installed last mile, for ILECs to migrate CLECs off of UNE-P and for the FCC to promote healthy local competition without the need for regulatory distortions to the market. IT

Scott Wharton is vice president of marketing at BroadSoft. Scott was honored with the Frost & Sullivan Market Engineering 2003 IP Telephony Executive of the Year Award. For more information, please visit the company online at http://www.broadsoft.com.

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Network Assurance And Testing On The Road To Enterprise IP Telephony

Choosing the proper converged IP telephony solution for your company based on features, compatibility, and scalability is a challenge in and of itself, but in the excitement over the opportunities presented by the purchase of a new system, it's important to not lose sight of just how critical proper implementation remains.

With Opportunities Come Challenges

Making the leap to any type of IP-based communications solution can be a very real and very frightening prospect, and in making such a leap, it's important to remember that your data network will need to serve double duty when you begin running real-time voice and video applications over it, whether you're converting to a pure IP solution or working with existing PSTN (define - news - alert) equipment and a gateway.

Migrating to a converged network opens the door to a New World of IP-based enhanced applications and services, as well as cost savings and productivity increases. But like all major advances in technology, the converged network also brings with it a unique set of challenges as well as opportunities. Careful planning throughout the purchase and implementation processes can help enterprises meet these challenges and prepare for them before they turn into quality of service issues.

Indeed, a slew of quality of service issues brought on by improper testing or failure to test at all can turn your converged network into the equivalent of a small town overrun by mischievous reptilian monsters, resulting in dropped calls, unacceptable quality levels, and general chaos throughout your enterprise communications. Such a situation can negatively impact productivity and efficiency, and ultimately your organization's bottom line. And that's no fun.

Quality Of Service Problems That Can Plague An Untested Network

By adopting a comprehensive testing program that tests your network infrastructure and taking the appropriate measures to ensure acceptable levels of service for all your applications, these QoS (define - news - alert) problems can be tamed, controlled, and prevented from multiplying.

Packet Loss: Back in the old days, when you were sending e-mail and HTTP traffic over your data network, a few lost packets were no big deal. But a converged network deployment is an entirely different beast, and packet loss equals unacceptable degradation of voice and video traffic quality. It results in everything from a patchy call in which voices can be garbled to a completely botched call.

Jitter: Voice and video applications have maximum latency levels that can be reached before a VoIP call crosses the line to an unacceptable quality level. Jitter is a variation of delay/latency levels over time, caused by actions like queuing and routing that affect the path of packets as they travel through the network. Testing is essential for deter-

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By Andy Huckridge

When it comes to buying equipment, there's a difference between vendor claims and sales tactics, and actual equipment performance.

mining how much jitter your network will experience and how it should be handled.

Delay/Latency: Exceeding the acceptable delay/latency levels for your voice or video application can make for an unpleasant call in which parties are forced to pause each time they make a statement and wait for the other party to hear what they've said. The network must be thoroughly tested under varying traffic conditions to accurately determine the maximum level of latency that can occur, and to properly configure quality of service solutions to accommodate that level.

Echo: Few of us who have participated in packet voice calls have escaped the effects of echo. This phenomenon occurs when participants can hear their own voices coming out of the earpiece, and can make a voice call completely intolerable. In a hybrid network, analog phones can be the source of echo, but a pure IP telephony configuration is also susceptible, since some calls will inevitably interface with a TDM (define - news - alert) network at some point.

Clipping: Clipping within VoIP calls occurs when either the beginning or end of words, or whole words, seem to be cut off during a conversation. This can occur when voice activity detectors and other solutions that work in tandem with echo cancellers are thrown out of sync. A careful balance must be struck, ensuring that proper levels of latency, jitter, packet loss and echo are maintained throughout the network at all times.

Primary Considerations When Migrating To VoIP

To avoid the risk of communications disruption and failure, it's important to ask the right questions and take into account a number of important considerations before choosing and deploying your new enterprise IP telephony solution.

Assessing Existing Network Performance: How will your voice and data communications operate in tandem on a broad scale? To successfully pair the two most important backbones of your company's communications, you will need to ensure your new converged network is capable of running at peak capacity and reliability at all times. You absolutely cannot guess or come up with a ballpark estimate of how much traffic will be sent over the network at any given time.

Ensuring Interoperability: Are your data and voice equipment interoperable with each other, as well as any legacy equipment you may be keeping in the mix? Packet telephony standards and protocols are changing and expanding each day, and you must understand that Vendor A's media gateway may not work with Vendor B's SIP (define - news - alert) proxy server. The alternative could mean costly last minute equipment purchases or worse — network failure.

Determining the Impact of Security Solutions: Adding voice and real-time video to your data network adds a layer of complexity to packet traffic, and you had better be prepared to test and overhaul your existing security solution to meet these new demands. Firewalls and other network security measures can seriously affect the quality of voice and video traffic by introducing additional jitter and latency, and in some cases make effective communications impossible.

Determining the Impact of Software Upgrades and New Hardware: When making changes within the network, what is the impact these changes will have on network performance? The last thing you want to do is to make these changes on-the-fly. By testing in a lab or in pre-deployment, you will be able to avoid network service interruption, and help ensure that your network has the capacity to process it's regular traffic flow while simultaneously undergoing a minor or substantial upgrade.

Assessing Real-World Performance Versus Vendor Representations: When it comes to buying equipment, there's a difference between vendor claims and sales tactics, and actual equipment performance. Indeed, many vendors make claims such as "guaranteed interoperability" with Product Type A or guarantees regarding load capacity or scalability. These products may have actually achieved interoperability or scaled in the lab — at least once before, if not twice. But what happens when products are deployed in the real world?

Right Sizing the Equipment Investment: What is the amount of equipment and the scalability you will require? You may already know what size budget you have to work with, but circumstances change, and you must ensure you purchase a system that is flexible enough to expand or be downgraded to match your company's needs today — and certainly several years down the road. Having a certain amount of foresight is a requirement as you begin the process of migrating into a new era of IP communications

The Benefits Of Testing A Converged Network

As we discovered, there is no room for compromise when it comes to testing your network infrastructure: Comprehensive lab testing, or at the very least pre-deployment testing, must be performed, and it must be done correctly for a number of reasons.

Optimizing Network Performance: You want your network to be running at peak performance at all times. But without proper testing, you run the risk of being inundated by the issues described above, or perhaps overcompensating for wild card problems in your network by allocating too many of your resources to quality of service assurance.

Maintaining Security: While you may be able to identify potential problems before you roll out your network and establish solutions for dealing with them, the only sensible way to identify these issues is to perform comprehensive testing in the lab or in pre-deployment testing to simulate real-world traffic on your network and analyze how it interoperates with your installed security solutions.

Driving Cost Efficiency: If testing is performed correctly, it will result in an efficient converged infrastructure that is scaled to accommodate the correct number of users and traffic load throughout your enterprise. If miscalculated, your new network could wind up being a costly mistake for your company. Failure to test is a major risk that will likely end up costing more money to fix in the long run than the incremental costs of testing.

The Bottom Line

The decision to roll out a converged communications solution is a solid investment and a no-brainer for you and your company. But to do so without properly testing your network first is an enormous gamble, and one that could hurt you and your company in the long run.

Testing is without a doubt the most reliable way to ensure your network will operate successfully in its new configuration. It ensures a network that can handle the amount of traffic that will be sent over it, that it can prioritize that traffic properly, that quality of service issues are accounted for in routing traffic, and that potential security hazards are discovered and taken care of before it's too late. It can simulate traffic patterns on your network you never dreamed would exist and enable you to prepare for their eventuality.

Testing can also play a role in ensuring your new communications equipment is interoperable in the real world, under varying conditions and in varying configurations, and help ensure that your CEO will always get a dial tone and crystal clear communications when he needs it. IT

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The Promise Of Presence

Combined with mobility, fixed/mobile convergence, VoIP, and multimedia, rich presence could become the key challenger to e-mail as the Internet's true killer application.

Today's PC users are becoming accustomed to instant messaging and presence as a new way to reach out and touch someone. Presence has proven itself to be an intuitive mode of communication that feels natural to use. Once people experience it, they expect it.

IP-capable mobile phone users soon will expect this too.

Instant messaging (IM) is poised to leapfrog standard e-mail communications by providing something that e-mail cannot: IM empowers users to share information about themselves with a select 'buddy list.' Presence puts the 'instant' in instant messaging by telling a message sender in real time whether or not the recipient is actually available. This ability to advertise activity can even be customized by the end user to provide yet more detail on his/her current receptiveness to messages — e.g., 'don't bug me right now' or 'only accepting CEO messages.' Avatars and .gif images can communicate mood.

In 2005, as 3G wireless infrastructure deployments rapidly increase around the world, presence is quickly becoming integral to the suite of broadband mobile IP applications that both enterprise and consumer subscribers demand.

The marriage of mobility with presence will revolutionize the ways in which businesses, families, affinity groups, and friends use networks. Mobile technology suppliers and international standards bodies have made significant strides toward enabling just that.

The Untethered 'Presentity'

Two international standards bodies are well on their way to turbo-charging presence and making it a ubiquitous fixed and mobile networks tool. These bodies include the Internet Engineering Task Force (IETF), which has long set IP-related specifications and extensions, and the 3rd Generation Partnership Project (3GPP), a consortium of wireless standards bodies defining the evolution of the GSM technologies. Both bodies are seizing on a common denominator, the IETF's Session Initiation Protocol (SIP) (define - news - alert), to develop further symbiosis between IP and mobility.

Specifically, the IETF (define - news - alert) and 3GPP (define - news - alert) are borrowing and sharing specifications to marry presence capabilities with VoIP, location, and IP multimedia capabilities.

SIP, the rapidly emerging IETF signaling standard for VoIP call set-up and tear-down, has enabled the IETF's SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) Working Group to build upon this platform to make VoIP and other wireless IP applications presence- and instant-communications-enabled. SIMPLE has extended the Rich Presence Information Data Format (RPID), adding optional elements to the Presence Information Data Format (PIDF). These extensions provide additional information about the 'presentity' (presence entity) and its contacts.

RPID (define - news - alert) allows mobile users to expose much more information about their willingness to communicate — with obvious mobility implications. It can speak to a user's current activity (in a meeting), location (on the exhibit floor of a conference), environment (noisy and distraction-filled), and even mood (harried). Further, this information can be structured to make it easy to provide different views to different groups of watchers, such as coworkers and clients, and designed so

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By Erik Lagerway

that much of it can be derived automatically from calendar files or user activity.

SIMPLE also has created the XML (define - news - alert) Configuration Access Protocol (XCAP) (define - news - alert) with two critical applications: management of contact lists and management of presence data exposure. The latter of the two includes the ability to protect access to sensitive geographic location information, as specified by the GEOPRIV Working Group.

Finally, the IETF's SIPPING Working Group's SIP CERT specification paves the way for user access to presence rules, contact lists or other data stored by the service provider as the user moves between devices. The goal is for any client to become synchronized with the latest presence information by subscribing to the server via the same mechanism used to get awareness of others' presence.

On the 3GPP side of the ledger, SIP is integral to the signaling aspects of the IP Multimedia Subsystem (IMS) standard. While the IETF standards promise to make presence transparent across fixed and mobile networks, IMS promises to make application access transparent across fixed and mobile networks.

The truly revolutionary thing about these complementary efforts is that for the first time, both the subscriber and IP multimedia applications are becoming untethered from any specific fixed or mobile network or end device. In effect, applications, as well as the subscriber's authentication identity and all his presence, location, and other data effectively travel with him across any network or device. Better still, because the presentity and buddy awareness data reside on the network, subscribers can enjoy single-login access to it all from any mobile or fixed device.

Opportunity: Anywhere, Anytime Presence

Significant numbers of businesses and families are likely to jump on the ability, via presence, to follow, find, and communicate with their colleagues and

Integrating Videoconferencing Into IP-Powered Presence And Collaboration Tools

By Scott Moule & Tom Falk

As powerful IP-powered applications like presence management, collaboration, and messaging continue to gain mind share in the business community, emerging video solutions embedded in these applications offer a whole new set of benefits to businesses. The prospect of merging video, desktop video to be exact, into IP-enabled productivity tools promises to entice customers with the prospect of easy-to-use videoconferences and video calls.

In its own right, the rationale for incorporating video communication capability into the enterprise network can be very appealing to a business. Video communications, when delivered at the highest levels of quality and reliability, is a cost-effective alternative to live meetings. According to a recent study from Harvard and Columbia Universities, videoconferencing results in a 38 percent increase in retention among participants. Desktop video adds value to business processes by impacting communication and collaboration patterns in very practical, visible ways that can positively affect a company's sales, operations, and expenditures.

Certainly, all of us are familiar with the traditional benefits driving the explosion of videoconferencing solutions. Yet the real power is when desktop video can be deployed to empower employees in remote locations. This ability to enable team members to meet one-on-one or in a group as needed from virtually any location with an Internet connection, rather than scheduling a conference room, and pulling all the participating parties into central location, can become a key factor in a company's decision making process to leverage a converged communications network.

What's more, combining video with IP-enabled presence management, collaboration, and messaging tools adds a new set of robust solutions that can tangibly help a business address their most fundamental needs, such as increasing revenue, managing operations, and controlling costs. It's a brave new world for IP communications, and video plays a major role in this growth.

Video Plus Presence

The effectiveness of utilizing videoconferencing is amplified by the existence of presence information in the enterprise network. As SIP (Session Initiation Protocol) and SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions) continue to gain acceptance in the marketplace, new standards-based applications that support presence will continue to drive interest. The implications are powerful.

The ability to instantly see the availability of the people you're trying to reach and create an ad hoc videoconference call eliminates the cumbersome task of trying to coordinate the schedules of team members scattered throughout an office, or around the globe and securing videoconferencing rooms in multiple locations. By leveraging presence information and new IP video applications, users can bring the appropriate participants into a meeting, sales presentation, or another dialogue in an expeditious and efficient manner, maximizing time and resources.

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An easy to use client interface that includes presence, collaboration, text chat, and desktop video capabilities also empowers team members to choose the appropriate medium in which they are most comfortable to communicate, either one-on-one, or in a group. This enables users to forego having their image broadcast on a conference call — if they desire. Through these easy-to use client interfaces, a person can elect other forms of media — such as a voice calls and audio bridges, Web collaboration, text chat, messaging, and other options to communicate. Presence availability is not bound to specific technologies — it allows individuals to choose the right communications vehicle at the most appropriate time.

Collaboration Is Key

Integrating video communications into a converged communications network leveraging advanced collaboration capabilities can make an even more profound impact on communications. Video calls, combined with whiteboarding, document sharing, on-the-fly conferencing, messaging and other tools redefines the nature of business communications. Colleagues can be brought into videoconferences on a moment's notice from any location, sales presentations can be modified on-the-fly — despite the fact that participants are several hundred miles apart. Through the combination of presence tools, collaboration and videoconferencing, the age-old business barriers of geography and time become non-existent. In today's environment, all employees need is an Internet connection, a Web camera, and the appropriate software to turn any location, like an airport terminal or a hotel room, into a full-blown conference room, demonstration center, sales office, and training site.

While the power of this new communications tool is undeniable, companies looking to leverage this technology should be mindful of several factors that can impact reliability, security, and network efficiency.

A Matter Of Bandwidth

As with most IP-enabled applications, ensuring network availability and reliability is of paramount importance when using next generation tools like video and collaboration. In general terms, the combination of desktop videoconferencing and IP-enabled applications do not require the significant bandwidth that have caused some businesses to shy away from these solutions. Today, video encoders can support up to 1,000:1 compression, delivering acceptable quality with as little as 128k bits per second. For example, for five meeting participants utilizing video, voice, and a document sharing program, about half a T-1 line would be required to handle the IP traffic for this scenario. When doubling the online participants to 10 guests, a T-1 line should provide sufficient bandwidth.

In terms of wireless connectivity, most 802.11 networks offer appropriate network availability to handle video for several users, although quality could be impacted when integrating other presence and collaboration tools. Usually, however, Web and video conferencing in remote sites involve only one or two people per location, which should result in more than enough availability to deliver all these value-added IP applications.

Today's online video technology offers selectable frame rate speeds up to 30 frames per second that creates almost broadcast quality images, with some providers delivering up to 32 frames per second, resulting in a superior picture and excellent audio. Again, the rule of thumb of a T-1 line for about 10 video and collaboration participants and three or so using an 802.11 network would be the appropriate network availability required for this activity.

Security Considerations

As might be expected in today's business environment, companies utilizing IP transport need to pay close attention to the security of their infrastructure. These considerations become even more crucial as companies add videoconferencing into the mix of mission-critical data and voice communications they place on a converged network.

In certain industry sectors, the emphasis on securing the network is even more paramount. In the healthcare and financial sectors, for example, the need for secure communications isn't just necessary, it is mandated by regulatory agencies who demand that the privacy and integrity of individuals and their personal records remain insulated from the innumerable attacks on both electronic communications.

Perhaps the simplest way to ensure the security of the network is through encryption. Although encrypting voice communications has not yet reached the mainstream, the technology has achieved broad acceptance for data communications for both large and small companies.

Likewise, encrypting video is also a reasonable solution to securing the network. For instance, adding CAST 128-bit encryption into the security configuration is a simple, inexpensive, and efficient solution that can help a customer meet all industry and government mandates. Although relatively few companies now use encryption as a security measure for voice and video, this will no doubt change as more and more companies look to integrate desktop video into their networks. In most cases, they're already encrypting their data; adding video to this security configuration just makes technical and economic sense.

In addition, there are a number of other measures that business may pursue in order to protect their vital communications. Some companies elect to password-protect their desktop videoconferencing, collaboration, and presence tools, while others may choose to have their network administrators maintain a lockout system so that only certain individuals can access desktop video and conferencing communications. In either event, companies have a number of security solutions to choose from that maintain the integrity of the infrastructure and traffic, while not impacting the quality of the communications and collaboration experience.

Evolution — Not Revolution

As advances in SIP, SIMPLE, and other standards evolve, there will likely be a continued escalation in the power of presence, collaboration, and messaging technologies. Desktop video will certainly play an integral role in this development as businesses look to identify new means to streamline and enhance communications. Video quality will no doubt increase as technology matures. But perhaps more significant, however, will be the growing acceptance of multimedia collaboration within the enterprise market, and the realization that desktop videoconferencing — coupled with these advanced IP applications — can help businesses address their most pressing business challenges in a tangible, practical, and cost-effective manner. IT

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loved ones wherever they go.

The ability to self-manage and expose one's data to select buddies will just as likely quell any fears of 'Big Brother' tailing one through life, whether a boss or literal big brother. One might expose availability to the boss, but hide one's current location at the beach.

Buddy lists combined with selective exposure of presence information also enables subscribers to form affinity groups, such as family, colleagues, project partners, clients, or sport team fans at playoff time. Adding mobility to the equation multiplies the effectiveness of managing and communicating with such groups. After all, a colleague or parent can more fully account for and reach an entire project team or family if every member can expose his presence data not only from his office desk, but also from his cell phone, laptop, or PDA — ultimately around the clock. Meetings and notifications become easier to arrange, ahead of time or on the spot.

Reception of a communication also becomes infinitely more certain. Conversely, presence enables one to avoid the uncertain, delayed responses associated with 'non-instant' e-mail or voice messages. The IP network user wastes less time and effort on attempts to communicate with absent interlocutors. From the service provider's point of view, a reduction in attempted communications with absent recipients will help limit dead-end network capacity usage — a bonus for wireless operators, particularly at a time when music, video, and other multimedia data sessions threaten to clog the pipe.

Once subscribers get practice with presence, they are also likely to alter significantly how and when they communicate with their peers or colleagues. An e-mail message goes off into the ether. A busy signal resulting from a phone call offers only the options to hang up or leave a time-sensitive voice message to uncertain reply. With intelligence about a person's availability, whereabouts, activity, and even mood, the options multiply. Communication itself can become more selective. One can wait for the right presence conditions to attempt contact.

Mobile service providers will not be slow to grasp the opportunities afforded by anywhere, anytime presence to develop affinity group notification, currently available contact lists, auto-conferencing, targeted marketing, and other presence-powered, value-added services.

Opportunity: Instant, Mobile Multimedia Communications

Mobile presence also ushers in possibilities for the intelligent selection of not only when, but how to communicate.

IM service providers have for several years integrated VoIP calling with IM in the fixed-line IP realm. In recent months, this voice-enabled IM phenomenon has shifted from proprietary technologies to compliance with SIP. As a result, some 40 million subscribers to Microsoft Corp.'s MSN Messenger service, for example, can now choose to make an instant VoIP call rather than send an instant text message. In effect, the VoIP-enabled phone can, through presence, tell the caller when nobody's home.

Where fixed and mobile service providers employ IMS application servers, an instant IP video call across mobile or mobile and fixed networks becomes a third possibility.

In other words, the marriage of network-hosted rich presence with VoIP and IP multimedia opens opportunities for users to structure conversations, from simple voice to multimedia sessions, based on unprecedented intelligence about whom one is calling, what they're up to, where they are, and how they feel.

For example, if a client's presence data indicates she is busy and hurried, a quick IM may have to suffice. However, if she's enjoying the luxury of time, privacy, and a receptive state of mind, an instant IP video conference becomes an opportunity to communicate with more power and immediacy. Indeed, rich presence can enable automatic notification when such ideal presence conditions occur.

Take a poll in early 2005 asking for the communications industry's killer application to date, and the strongest response will probably be for e-mail — an application virtually everyone uses daily. Take the same poll a few years from now, and e-mail just may come in second. Rich presence — empowered by mobility, VoIP, IP video, and the next iterations of 'family and friends' service packages — will prove hard to resist. IT

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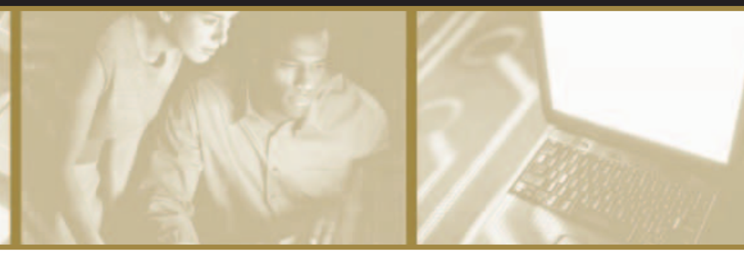

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Delivering Secure And Assured Voice Over IP

It is widely accepted that VoIP (define - news - alert) offers many business benefits, particularly lower operational expenses, but with these advantages come new security and IP routing challenges. Since security devices and enterprise routers were originally designed to protect and transport best-effort IP networking data, they are not necessarily up to the task of delivering the reliability, security, and performance required by converged voice and data applications. Routing performance and reliability are absolutely critical because they impact quality of service (QoS) and ultimately end-user customer satisfaction. Security also presents additional challenges with VoIP because attacks have the potential to affect not only the voice service, but also the entire IP network.

These additional requirements mean that enterprises deploying VoIP need to evaluate their entire network infrastructure from both a routing and a security perspective. Simply selecting the ideal PBX (define - news - alert) — while a good start — is not enough to ensure the overall security, quality, and reliability of VoIP services.

VoIP Security Requirements

Traditional voice services never required security, but VoIP is entirely different. VoIP creates many new security challenges because the convergence of voice onto the standard IP network means that any security breaches or attacks targeted at the voice service can affect the network as a whole: e-mail, data backup, etc., could all be compromised by security breaches or attacks. The reverse is also true: in a converged network, any IP-based threat is a threat to the voice service, including viruses, denial of service (DoS) attacks, and the like.

Closing The Back Door

Perhaps the two most common security threats are DoS attacks and PBX hacking. A DoS attack on the PBX involves a flood of requests from an attacker that overwhelms the PBX, limiting or disabling its ability to set up calls or support features for existing calls. Typically, general DoS and Distributed DoS (DDoS) attacks are launched from outside the network.

PBXs contain a variety of confidential enterprise information that could be devastating if the information ends

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By Scott Heinlein

up in the wrong hands. Common information includes call logs, employee directories, customer contact information, voice mails as primary examples. Imagine someone hacking into an enterprise PBX days before the company's earnings announcement and listening to a voice mail from the CFO to the CEO. Or someone getting a hold of the enterprise customer contact list and selling it to a competitor.

A PBX that has been hacked could also be used to make unauthorized phone calls, commonly referred to as toll fraud. Toll fraud can quickly become costly if undetected for an extended period of time.

Firewalls provide the first line of defense against both these attacks by rate limiting certain protocols to acceptable thresholds. Firewalls that do not "understand" VoIP protocols can leave the network open to attackers and hackers because they typically require a range of firewall ports to be opened to allow VoIP calls in and out of the network. Leaving parts of the firewall open for VoIP is dangerous, because unnecessarily open ports are an open door into the corporate network.

One method of closing these doors is by using Application Layer Gateways (ALGs) to interoperate with PBX vendors using H.323 (define - news -alert) and SIP VoIP protocols. ALGs (define - news -alert) allow ports to be opened when a call is initiated or received and subsequently closed when the call is complete.

Keeping the PBX virtually separated from the rest of the enterprise network can also contain hackers and DoS attacks targeted at other portions of the network, preventing them from affect-

Encryption is perhaps the best prevention against voice interception.

ing VoIP sessions. Firewalls can place the PBX, IP phones, and other network components into separate logical zones and apply specific security policies that must be satisfied to gain access to individual zones.

Zones can be created in a variety of manners. Virtual LANs (VLANs) are very effective inside the network and VPNs (define - news -alert)work well across the WAN. In either case, the ability to apply specific security policies for access to a specified 'zone' provides an extra level of security by applying policies to stop unauthorized attackers from accessing the PBX zone.

Voice Call Interception

Interception of voice calls in the legacy voice environment typically required physical access to the voice network, but interceptions of VoIP calls is much simpler because physical access is not required. Someone across the globe can intercept a VoIP call and eavesdrop or perform a 'man in the middle' attack (in which attackers intercept a call and input words into the conversation, unbeknownst to the parties talking).

RELATED LINKS

Voice over IP Security Alliance
www.voipsa.org

Voice over Packet Security Forum
www.vopsecurity.org

Encryption is perhaps the best prevention against voice interception. Many enterprises encrypt voice conversations across the wide-area network (WAN) using IPsec or another encryption technique.

While encryption eliminates most concerns about eavesdropping, it can also create latency, affecting voice quality. If latency levels become too high, voice communications become difficult and eventually impossible. Enterprises need to ensure network devices are capable of encrypting and decrypting the high volume of small packets required by voice so latency remains minimal.

Delivering High-Quality VoIP

Along with securing voice communications, ensuring VoIP performs with high levels of quality and reliability is essential for successful VoIP deployments. Most legacy enterprise routers and network designs were designed to deliver best effort IP performance, but voice is one instance where best effort is not good enough. Voice and other real-time applications are easily affected by latency, jitter, and packet loss, and therefore demand that routers perform at a higher level.

According to industry research, voice communications begin to erode if latency exceeds 100 ms, jitter exceeds 50 ms, and packet loss is greater than one percent. These stringent requirements mean each network device must be capable of the highest levels of performance to ensure minimal latency, jitter and packet loss are introduced into the network.

Prioritizing Voice

Network congestion is one of the most common reasons for low-quality voice because it can queue voice packets, increasing latency, jitter, and packet loss. Routers with QoS capabilities mark voice traffic and can assign it a higher priority in the network queue. In times of congestion this allows voice traffic to be forwarded before less important traffic such as e-mail and Web browsing.

But in some cases the functions of classifying, scheduling, and queue management can actually degrade overall router performance since it can be processor intensive. Enterprises need to consider the ability of their routers to perform QoS functionality without degrading router performance.

ASIC-based technology can offset the performance issues that can occur with QoS functionality by assigning specific functions to purpose-built chips. Routers that utilize protected CPU resources for specific functions also accomplish the same goal. Both of these acceleration methods can assist network devices efficiently handle voice packets, enabling the prioritization of voice traffic without degrading router performance.

Enhanced Routing Techniques

The unique requirements of VoIP require new routing techniques that were not required for best-effort IP networks. Traditional IP networks take a number of seconds to detect and recover from an IP link failure. A few seconds is not a considered too long for a best effort data application such as e-mail or Web browsing, but would certainly cause a noticeable interruption in a voice call.

New techniques such as enhanced hello timers or Bi-directional Forwarding Detection (BFD) can bring this IP link failure detection and recovery to well under the threshold required by voice communications.

Ensuring a secure and high-quality VoIP service requires careful examination of the entire network.

Large enterprise networks can also leverage MPLS (define - news - alert)technology for traffic engineering, reservation of bandwidth, VPNs, and high-availability recovery mechanisms for voice packet traffic. These are all techniques available today and designed for supporting real-time applications.

IP Link Services

Voice packets are smaller than data packets, and this small packet characteristic places additional requirements on the network. Small packets can be blocked by larger data packets, which can introduce latency and jitter, and ultimately affect the quality of the voice service.

Link Fragmentation and Interleaving (LFI) breaks up larger data packets and interjects voice packets, helping to eliminate jitter. Compressed Real Time Protocol (cRTP) can also be used to compress packet header from 40 to two bytes, improving VoIP performance and capacity on links. cRTP can be used in conjunction with LFI to reduce delay and jitter on lower speed links.

Conclusion

VoIP deployments are booming and enterprises gain many benefits from using VoIP technology. While enterprises should not hesitate to deploy their own VoIP solution, ensuring a secure and high-quality VoIP service requires careful examination of the entire network from both a security and reliability perspective. By employing the technologies described above, enterprises can be assured that their VoIP services are as secure and reliable as the rest of their network. **IT**

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VoIP MARKETPLACE

CEO Spotlight

Joe Licata, president of the Enterprise Networks division, Siemens Communications, Inc.

In the CEO Spotlight section in *Internet Telephony*® magazine, 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 Joe Licata, president of the Enterprise Networks division of Siemens Communications, Inc.

GG: What is Siemens' mission?

JL: In a nutshell, Siemens' mission is to eliminate communication chaos. We call this vision LifeWorks@Com: the development and deployment of collaboration innovations that are independent of networks and devices. Our job is to make communication technologies work seamlessly together. With the simultaneous emergence of VoIP, fixed mobile convergence, expanded mobile enterprise networks, and presence-enhanced collaboration applications, we believe that the day for completely unified communication has finally arrived.

GG: How is Siemens positioned in the next-generation telecom market?

JL: Siemens (news - alert) has adapted quickly to a rapidly changing communication landscape. In October, we merged our fixed-line and wireless networking companies to form Siemens Communications. We understand that there is a lot of competition out there, but Siemens is truly unique in its ability to provide a comprehensive portfolio of solutions for wireless, fixed, enterprise, and service provider networks — all working together. Siemens' full range of expertise is already paying dividends. Recently, for example, we were chosen to help Cingular build out its 3G network. Enterprises are also coming to Siemens, and our new affiliate, Chantry Networks, is helping Siemens to further expand its strategic deployment expertise for a new generation of merged VoIP (define - news -alert), WLAN (define - news -alert), 3G (define - news -alert), and other communication network options. Enterprises and service providers may continue to deploy only pieces at a time, but they now understand the value of a partner who has a comprehensive leadership presence throughout the ever-expanding and ever-changing web of next-generation communication options.

GG: What is it that sets Siemens apart from its competition?

JL: Openness to imagination. While our competitors ask, "Why?" we say, "Why not?" In the communication industry, Siemens was among the first to recognize the disadvantages of one-size-fits-all communication networks. Now people and enterprises are awakening to this idea. As long as systems are open to innovation, communication technologies can be pushed to meet very unique personal preferences and business needs. With the first generation of VoIP, most of the industry only played lip service to the foundation of open, standards-based systems. At the end of the day, it is these same proprietary and point-based systems that have created communication fragmentation — applications, services and devices that don't work together. Convergence, via open, standards-based solutions, will continue to be the Siemens' difference. Customers deserve the right to select best-in-class solutions — even if those solutions come from different vendors — to best meet their specific communication needs.

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?

JL: Outdated regulations and, even worse, new regulations that aren't relevant in the VoIP telecommunications world. The enterprises we work with need communication tools that make employees more effective, more productive and more responsive to customer needs. Carriers and service providers — SBC, Bell South and Cablevision, for example — need to create new services and sustainable business models. End users simply want devices, networks, and applications without restrictions. The U.S. regulatory environment must not inhibit the innovation that's just taking off. If there are new regulations, rules must be applied evenly across network platforms. And, finally, broadband penetration must be encouraged everywhere. The United States, the most innovative country in the world, still ranks only 13th in terms of per capita broadband penetration.

GG: What are some of the areas where Siemens is increasingly focused on and why are these areas

important to the future of your company?

JL: We're seeing a new managed services trend in the enterprise space. Managed services made up less than 18 percent of the total service market in 2003, but managed services are expected to account for nearly 50 percent of the market by 2010. The definition of managed services is also changing. With the next generation of VoIP, enterprises have more deployment options — including the adoption of hosted solutions whereby the communication gear isn't even on the premise anymore. In short, enterprises and CIOs want more choices that allow them to focus on core business issues.

> Our job is to make communication technologies work seamlessly together.

GG: Describe your view of the future of the IP telephony industry.

JL: A lot of new things will continue to happen as we integrate VoIP architectures, voice over WLAN, 3G, and other communication technologies — both in the enterprise environment and in the home. The real breakthrough, however, is the ability to provide complete access to communication features and related applications no matter what device, mode, or network a person finds themselves using. Siemens recently demonstrated such a scenario, combining presence with an IP-based Multimedia Subsystem (IMS) platform to deliver instant messaging, interactive mobile gaming, friend list/network presence detection, video telephony, push-and-talk over cellular and other multimedia IP services across both fixed and mobile networks. At a national cable trade show, we demonstrated the use of a cable television as a real-time source for caller ID and voice mail message information at the home. Wires don't go where people go, but that will no longer be a hurdle. The future: communication without limits. IT

CEO Spotlight

Benjamin Sayers
CEO
VoIP Supply

In the CEO Spotlight section in *Internet Telephony*® magazine, 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 Benjamin Sayers, CEO of VoIP Supply (news - alert).

GG: What is VoIP Supply's mission?

BS: Our mission is to be the "go-to" value added reseller in the VoIP (define - news - alert) industry. We will accomplish this by providing the largest breadth of products and services with competitive pricing, representing these products and services with highly trained sales and customer service representatives and supporting our customers through the uncharted and ever-changing technologies that comprise the rapidly expanding VoIP marketplace.

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

BS: My vision for the future of VoIPSupply.com is clear with respect to achieving our mission and maintaining the status described. With the VoIP industry gaining the traction that is has over the past twelve months and the tremendous potential being involved in at this early stage, my vision is focused on obtaining the required assets to complete the task and exceed the goals set forth. We are experiencing rapid growth, on pace with the industry. To maintain the high level of customer service, provide definitive support, and meet the demands of our customers in all departments, we are continually adding staff, facilities, and the required capital to support the growth.

There are an overwhelming number of opportunities available to cloud the vision and distract from our mission. We will continue to add products and services so long as they fit our business model and do not distract us from completing our mission or diminish our ability to surpass the competition in areas such as support and customer service.

In the next-generation telecom market, just like the previous telecom booms, there are many potential business opportunities and needs that must be filled. With our past business experiences, staff, facilities and specialties, we have selected the VAR model as the need that was not being met by our competition — and our customers deserve more than they tell us they receive from the rest of the pack. While there are many other aspects of the business that are lucrative and tempting to get involved in, there are times when we find it difficult simply to keep up with the demands of customers coming to us for equipment and support. For the time being, we must remain focused on our mission and leverage our early entry into this expanding market.

GG: What is it that sets your company apart from the competition?

BS: For the limited amount of competition that we have, there are many things that set us apart.

• **Knowledgeable staff:** Our sales, support, and customer service representatives are continually trained on VoIP technology and the many products that we sell and service.

• **Availability:** We recognize the global demand for VoIP technology and the required components. Our sales staff is available 24/7 and our support/service staff can be reached by phone or e-mail, just about any time of the day or night. From what most of our customers have told us, any previous experience they have had with a competitor of ours has been less than enjoyable and generally included a lack of availability or unacceptably delayed response times.

• **Price:** Most people looking for VoIP technology are doing so with cost savings in mind. We understand this and cater to that by offering very competitive pricing, bundled with our knowledgeable staff who are available when customers are in need of information. In addition, due to our high volume of sales, we are able to offer reduced pricing and volume discounts to end-users, resellers and service providers.

• **Bundled Services:** One of the core markets that we serve is the service provider. While we can sell the equipment at reduced, volume discounted prices, we can also exhibit the definition of Value Added Reseller (VAR) by going a few steps beyond. With respect to the service providers of the world, we offer a total package. Not only will we provide reduced pricing, but we also provide provisioning of equipment, fulfillment services to their end users, collateral inclusions which promote their services

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to our customers and a hosted solution for their customers to purchase equipment online that in turn is provisioned and fulfilled by our in-house staff.

VoIP has gained its recent popularity and growth as a result of expanded broadband service.

• **VoIP at the Core:** We sell VoIP gear as our core business. For most of our competition, VoIP equipment in just another category in their catalog of equipment; a small aspect of their business offered only as a courtesy to their customers who have come to them for some reason other than equipment. We know VoIP and stick to it as our core line of products sold and serviced.

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?

BS: Yes, the industry is up on the radar and is likely to be subject to increasing regulation and litigation. As with any new industry, it will go through many revisions, modifications, consolidations, and likely some government regulations. VoIP has gained its recent popularity and growth as a result of expanded broadband service, making it more readily available in an acceptable quality. Broadband availability will continue to play a role in some areas of the world. The biggest factor I see and encounter is a general lack of education related to the costs, requirements, availability, and capabilities of VoIP technology.

GG: What are some of the technology areas where VoIP Supply is increasingly focused on, and why are these areas important to the future of your company?

BS: We are continually adding new products as they become available and new manufacturers as they enter the VoIP arena.

• WiFi — There are many new products combining VoIP and WiFi, we are adding these products as they become available.

• Video — A natural addition to VoIP is the combining of VoIP telephony with video capabilities. Several companies are planning the release of Video phones during the next few months. I see these as a being an addition to our catalog of products with a significant demand over time.

• Analog Alternatives — Many of our customers are coming to us for VoIP gear with the primary intention of saving money. Unfortunately, like any new technology, the prices for many VoIP devices are still high and can be a deterrent for some. We have begun to create and market several analog alternatives by combining a low-cost ATA (define - news - alert) with a traditional analog device such as a conference phone. For some it makes much more sense to save a few hundred dollars and putting up with a small device next to the conference phone than spending $800 or more for the latest SIP enabled conference phone. The same concept makes sense with a small business purchasing a single ATA and an expandable cordless phone system than a phone or device for each person in the office.

From a service standpoint, we have added a division of the company dedicated to providing provisioning and fulfillment services to the increasing number of VoIP service providers around the globe. Many of these providers are not set up to do both service and fulfillment. Many are limited in their facilities and most lack experience in logistics, supply chain and other facets of the fulfillment services industry. I used to own an application services provider (ASP) company and know from experience the difficulties that would arise as a result of being in an office complex versus a warehouse. Trying to juggle both the growth of the service business and the synchronized growth of a fulfillment department would not be beneficial to service growth.

These new products and bundles are important to us, not only from a sales vantage, but from a customer standpoint whereas we are providing the largest selection of products and alternate options for both business and home users of VoIP. The service side of the business is very important in building and growing our customer base in the commercial sector, providing an invaluable service to the VoIP providers.

GG: What does the road ahead hold for IP telephony industry?

BS: That's an interesting question, but not one I could answer with any degree of certainty.

• Given the incredible growth of business over the past year or more and the relatively small dent that has been created in the telecom world, I would say that continued steady growth could easily be foreseen by most.

• I would also expect to see increased attempts to regulate and/or tax the service by the government.

• In addition, there will most likely be some heavy consolidation in the industry though that seems a bit far off from now.

• With the continued education of consumers, bringing to light the "no brainer" aspects of VoIP, continued focus on integrated VoIP products for the home will persist and expand.

• As with any technology, in conjunction with demand and competition, the prices for the required devices will likely be on a slow decline. IT

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