TMC
INTERNET TELEPHONY
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Selection Guide!
We Round Up Over 25 IP PBXs
(page 50)

The VoIP Authority Since 1998™

The World Is Your Office With Mobile VoIP

Also In This Issue:

- Spirent Communications' Anders Gustafsson Speaks Out
- The Fourth Age of VoIP
- TMC Labs Reviews VocalTec & Shunra

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The VoIP Authority

By Greg Galitzine

Mobile VoIP: The World Is Your Office

Eight years ago, CTI Magazine featured a cover story on Data Race, entitled “The World Is Your Office.” Data Race made a product that essentially allowed folks to telecommute using ordinary analog phone lines to access the corporate network, thereby giving them access to e-mail, files, faxes, etc. This product also featured some computer-telephony integration extras, such as a PC-phone GUI and a contact manager. So in 1997, you could use IP to make office-based voice communications resources available to remote workers. It was genuinely big news at the time.

This past month saw announcements from 3Com and Nortel proclaiming partnership agreements with Research in Motion, the company that manufactures the increasingly ubiquitous Blackberry device. Essentially the gist of these agreements is that you can now use a wireless LAN to make office-based voice and data communications available to mobile workers on a campus environment.

The Nortel announcement focused on interoperability and integration between RIM’s Blackberry 7270 device and Nortel’s Multimedia Communications Server (MCS) 5100, a critical element of Nortel’s enterprise IP telephony solution. By integrating these systems, enterprise customers will be able to use the new BlackBerry 7270 to access IP telephony, e-mail, and other business applications via the wireless LAN.

The 3Com/RIM union is similarly focused on giving organizations the flexibility of a BlackBerry handheld that is tied into their existing PBX telephony solution. 3Com’s Convergence Applications Suite (and their wireless LAN switch) and RIM’s BlackBerry 7270 solution will enable on-campus employees to use always-on applications such as e-mail, VoIP, etc…, by integrating advanced SIP standard-based products.

Both agreements illustrate initiatives aimed at delivering secure, full-featured, SIP-enabled business communications over wireless LANs.

Our industry has always focused on providing knowledge workers increasing access to those applications that allow them to be as productive as possible. In 1997, the Data Race solution enabled telecommuters to access PBX functionality from a fixed alternate location, say, a home office. Today’s solutions are focused on giving employees full mobility within their WLAN-enabled enterprise campus, allowing them to take advantage of all the applications that are available to them on their IP PBXs as they move from building to building. This means they will be able to leverage presence and wireless VoIP to maintain full control of their telephony environment as they roam away from their desks.

Moving beyond the confines of the WLAN-enabled campus is the obvious next step. Through advances such as mesh networking and WiMAX, truly mobile workers will soon have access to all of their desktop apps while out in the field.

-Greg Galitzine

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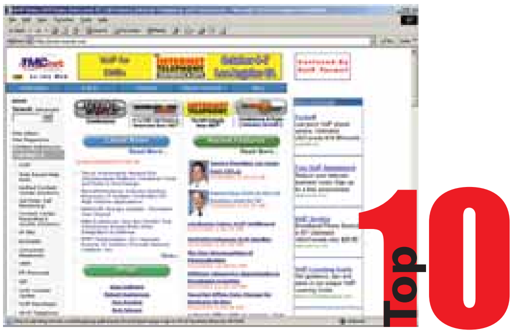

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8. Middletown, New Jersey, USA
9. London, EN, United Kingdom
10. Englewood, Colorado, USA

QUOTE OF THE MONTH:

"It is not a question of whether VoIP and mobile services will converge — but a question of the best path to take. While several approaches have emerged, the choice to use open standards, primarily SIP, which provide the call control flexibility to implement many different user experiences and service provider business models, will accelerate the development of the ecosystem of vendors and technologies that carriers will use to implement mobile VoIP."

– Sanjay Jhawar

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.

Buffet & Soros Like Cable

Two of the world's richest people and most savvy investors are eying the cable industry. Is this a sign that video-on-demand and other such services will be highly profitable or that a wave of consolidation will make the big bigger and stronger?
http://tmcnet.com/91.1

The Battle Of The VoIP-Call-Blocking Telecoms: Round One

Vonage Holdings Corp. claims to have discovered that some of its phone calls are being blocked by an undisclosed major telecom, causing a reported disruption of service for customers.
http://tmcnet.com/92.1

WiMAX: Nirvana Delivered?

Whether or not the putative threats to WiMAX materialize or not, all players within our space must seriously consider the impact on their businesses that will result from WiMAX implementation.
http://tmcnet.com/93.1

Looking To Hook Up At Cannes

Some execs have pretty specific goals in mind for that special someone they want to meet at 3GSM World Congress 2005 running from today through Thursday.
http://tmcnet.com/94.1

Report: IBM Sees Surge in Security Threats For Mobile Devices

IBM announced the results of its 2004 Global Business Security Index Report and presented an early look at potential security threats in 2005.
http://tmcnet.com/95.1

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.

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By Rich Tehrani

Microsoft, Google, Vonexus & VoIP. Oh My.

I was recently invited to keynote the first-ever Vonexus Connection conference. Vonexus, an Interactive Intelligence subsidiary, offers a Microsoft-based IP telephony (define - news - alert) solution for SMBs. It seems to me that they are onto something big. Many people in the industry fear Microsoft will be stalking the VoIP (define - news - alert) market ready to reach out and snatch the best and brightest in IP telephony. After all Microsoft has shown it can take out any company it wants. I give them credit for killing the VoIP client market that VocalTec created by launching their free alternative — NetMeeting — shortly after VocalTec came into existence. This is ancient history now… Going back to the late '90's but instructive nonetheless.

Microsoft lost interest in NetMeeting. If they hadn't, it could have been 100 times more successful than Skype (news - alert). The point is Microsoft understands VoIP. They see the opportunity and they are ready to pounce on the market and take advantage of it in any way they can. Vonexus has such tight integration with Microsoft products it literally fuses telephony into your corporate ecosystem. This is why Vonexus should have tremendous Microsoft support.

I believe the future of IP telephony is VoIP 2.0. To me, this concept means, "What's next?" "What will we see in the future of communications?" Without a doubt, the integration of telecom into our applications is a logical progression from where we are today.

First, Some History

The 1980's saw the advent of the predictive dialer. This device fused telephony tightly with desktop applications in high-volume outbound calling environments. Productivity soared hundreds of percent in call centers as a result. Agents didn't need to dial, redial wrong numbers, or even listen to telephones ring as they were immediately connected to customers as soon as they picked up the phone and said "Hello."

Fast forward to today where we have an IM application, a telephone, a cell phone, a VoIP client, a home phone and about a half dozen message stores from corporate e-mail to G-mail to Hotmail to cellular voice mail to Skype voice mail to Vonage (news - alert) voice mail… Are we more productive? No. Are we going in the wrong direction? Yes. How much worse can we make our communications when it takes up to four calls to get a hold of a colleague? I find myself many times IM-ing, e-mailing, and calling a colleague simultaneously to get a hold of them.

The future is in the integration of telephony and desktop applications

And Now, The Future

It seems obvious that the future is in the integration of telephony and desktop applications — simplification, as it were. I believe Microsoft understands this and I believe Vonexus may be the beneficiary of a potential partnership with the Giant of Redmond.

If it isn't Vonexus, it will be someone else. Microsoft will partner with a company that does something similar and improve the integration over time. They may even decide to (re)invent it themselves.

Surely Microsoft isn't the only company that can integrate telephony tightly. Oracle can do the same with their applications and already does with their CRM products and services. They can easily port telephony functionality across their product line enabling you to dial from a database app or an accounting program.

Another intriguing idea that embodies the integration of VoIP 2.0 is something that I mentioned in my blog in late January (http://tmcnet.com/87.1), the idea of Google getting into VoIP. There is much speculation about this and the rumor started as a result of a personal ad Google ran regarding the need for dark fiber and a person who can manage it. Whether the rumors are true or not is anyone's guess but what is worth thinking about is the fact that Google is slowly building desktop alternatives to Microsoft from desktop search to e-mail. Will VoIP integration with their current search, e-mail, and address book products be the way Google jumps ahead of Gates' baby?

If you think about it, isn't Google in an interesting position to do a search on incoming callers? Imagine when your phone rings and the caller ID says "Jim's Mortgage Company." Wouldn't it be interesting if you received a screen pop (a technical term used in the contact center space to denote bringing up customer history when the phone rings) with the results of a Google search? What if Google organized all of the results by tabbing different results into related categories, so I could

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click on a tab marked “Better Business Bureau,” or “Legal Issues,” or on a positive note, “Case Studies,” or “Customer List?”

Even if Google weren’t to get into VoIP directly but instead chose to work with other providers…

Wouldn’t it be great to integrate this search functionality with Caller ID? Imagine if they connected such a system to their corporate search appliance? You could also have Tabs for “Invoices Past Due,” “Orders Sold,” and “Notes” to name but a few.

I am more enthused now about the future of communications than at any time in the past.

I am more enthused now about the future of communications than at any time in the past. I see true openness afforded by VoIP and I see the ability for us to communicate in a much better way than ever before. What is different about communications today than at any time in the past is that any company is able to drastically alter the communications landscape. Furthermore, any company can change the way we think about communicating. Vonage popularized VoIP, Packet8 popularized videophones, and Skype showed you can have millions of people download your VoIP software in a matter of months with no marketing and no sales force.

Contrast this to ten years ago when innovation was only done by large telecom companies and was sold to you as part of a closed system from a single vendor. VoIP 2.0 is more about where we are going, it is about the minds that are unlocked to try new business models on a whim. Some will fail, some will be tomorrow’s Skype and some will evolve slowly and steadily to make our lives better as we communicate more effectively and efficiently. No company or country will dominate the VoIP 2.0 landscape… It will be led by the best and brightest minds of tomorrow with few restrictions on their innovative spirits. I look forward to bringing it to you monthly in this column or hourly in my blog at www.richtehrani.com. IT

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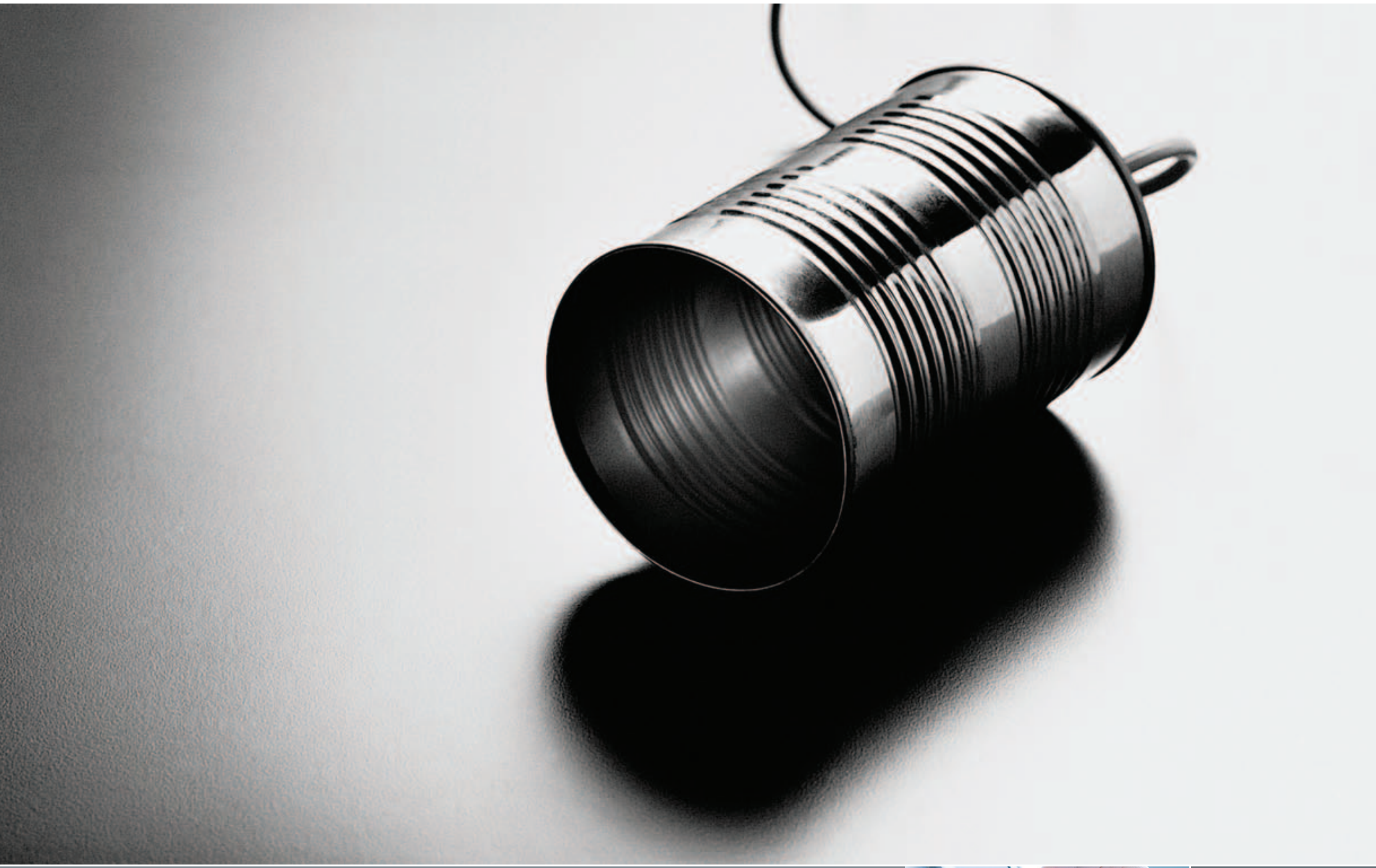
The New Headsets are In



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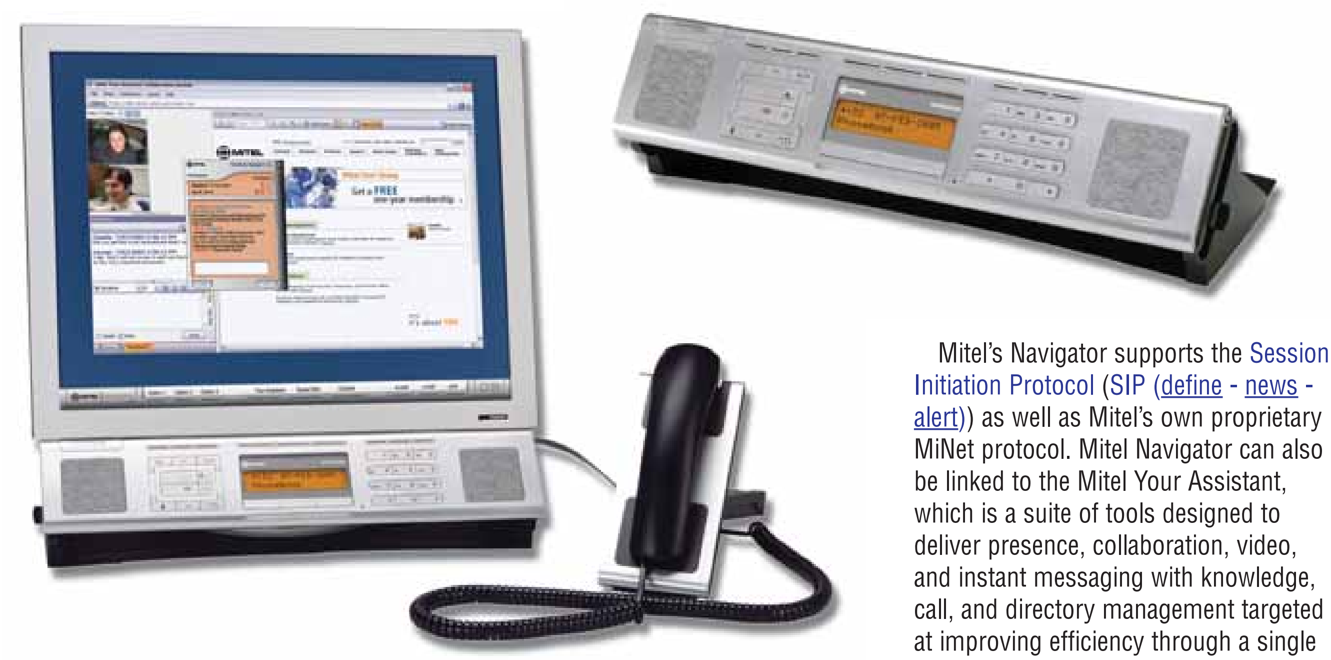
Mitel Navigator Sets Sail For Desktop Apps

Mitel, a provider of IP communications solutions, recently launched Mitel Navigator, a breakthrough communications device whose design changes the meaning of voice, video, and data convergence on the desktop. With an entirely new form factor, Mitel Navigator delivers a novel user experience and can be personally tailored for specific horizontal and vertical market applications.

"Convergence is a concept that can be applied not only to voice and data, but also to devices — particularly when it comes to the desktop communications device of the future," said Ronald Gruia, Program Leader for Enterprise Communications Solutions, Frost and Sullivan. "With the Mitel Navigator, Mitel is definitely pushing the innovation envelope towards a unique endpoint form factor. It is fully integrated with the Mitel Your Assistant, delivering new features, SIP functionality, and a powerful user interface."



Mitel's Navigator supports the Session Initiation Protocol (SIP (define - news - alert)) as well as Mitel's own proprietary MiNet protocol. Mitel Navigator can also be linked to the Mitel Your Assistant, which is a suite of tools designed to deliver presence, collaboration, video, and instant messaging with knowledge, call, and directory management targeted at improving efficiency through a single interface. The Mitel Navigator can also be used with Microsoft's (quote - news - alert) Live Communications server and Instanbul client.

"As part of the Microsoft Istanbul Technology Adoption Program, we are delighted that Mitel is contributing as a software partner and is exploring how converged technology can be addressed from the desktop", said Marc Sanders, senior product manager, Real-Time Collaboration Group, Microsoft Corp."

Nine programmable, self-labeling buttons are accessible from the Mitel Navigator or via the click of a mouse. These buttons invoke telephony features, PC applications, or commands. One can be assigned for scrolling giving the user an ability to access up to a total of 27 commands and tasks. The Mitel Navigator also supports PC speakers and headphones that can be used with the embedded full-duplex microphone and speakers for hands-free or conference calling. When the PC is being used to play music or other audio streams, Mitel Navigator normalizes the output using a phone-sensitive volume control in the event of an incoming or outgoing call.

According to Don Smith, Chief Executive Officer, Mitel, "Mitel Navigator is all about making sophisticated communications as simple as a phone."

The product will be available in Q3 2005.

http://www.mitel.com

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Kentrox QoS Access Router Aids Growing Businesses

Kentrox has announced the availability of the Q2400 QoS Access Router with a flexible, multi-port architecture and other key features designed to accommodate growing networks for small to medium-sized businesses.

The Q2400 QoS router, with dual T1 and Ethernet WAN ports, gives businesses choices as they expand their networks and transition to Voice over IP (VoIP). Network access needs grow over time as businesses add new locations, install new applications, and deploy more users. Additional equipment to accommodate expansion is costly. The Q2400's flexible multiple port architecture is designed to enable network growth and application upgrades including:

- VoIP migration;
- Virtual LAN support;
- Low-cost redundancy and flexible backup;
- Multiple connections;
- Secure access; and
- Scaling networks.

The Q-Series Q2400 access router integrates six networking solutions in one box — high-performance IP router, QoS appliance, VPN appliance, firewall, WAN access device, and Ethernet switch.

"No matter which VoIP equipment you plug into the Q2400 — VoIP PBXs or gateways — the system will automatically prioritize voice, and provide traffic graphs to let you view and distribute bandwidth as you need it," explained Dan Murray, Vice President of Marketing for Kentrox.

http://www.kentrox.com

Inter-Tel Unveils New 5000 Series Solutions

Inter-Tel, Inc., released its Inter-Tel 5000 Network Communications Solutions, a family of IP-centric communications systems developed specifically for small and mid-size businesses.

"We speak with hundreds of businesses on a regular basis that are eager to leverage powerful VoIP applications like presence management and collaboration to help increase revenue, manage operations more efficiently and control costs," commented Craig W. Rauchle, Inter-Tel's chief operating officer. "We have developed Inter-Tel 5000 Network Communications Solutions with these customers in mind."

Inter-Tel 5000 Network Communications Solutions include two platforms designed to deliver robust and reliable communications along with advanced VoIP-enabled applications to customers. The Inter-Tel CS 5200 is geared for smaller businesses, scaling up to 25 IP endpoints, while the Inter-Tel CS 5400 can accommodate up to 110 endpoints. In addition to the supported IP endpoints, both platforms can support up to 96 digital endpoints via optional Digital Expansion units.

The CS-5200 and CS-5400 platforms feature a 1U data-centric form factor that is designed to fit into 19" racks. The Linux-based platforms provide integrated support for Session Initiation Protocol (SIP) trunk gateways, and offer survivable node capabilities for remote offices to ensure reliability and 911 access. Both platforms feature four-port voice processing, full transparent networking over IP, and Compact Flash storage for voice mail. Both platforms can accommodate a combination of 74 IP, T-1 and analog trunks.

Inter-Tel 5000 Network Communications Solutions seamlessly support Inter-Tel's Model 8600 series IP endpoints, as well as the company's presence management and collaboration tools, such as Unified Communicator and Enterprise Conferencing. Additionally, these new platforms can be transparently networked with Inter-Tel's flagship Axxess communications platform, providing businesses with more choice and flexibility in addressing business challenges.

http://www.inter-tel.com

RADVISION Brings 3G Video To Microsoft Desktops

RADVISION announced a 3G video services solution for the Microsoft desktop multimedia communications architecture. This integrated solution enables real-time video calling between 3G video handsets and PCs running Microsoft's integrated communications client, code named Istanbul.

"Whether they are at their desks, on the road, or in the home, business professionals today need to stay in touch," said Marc Sanders, senior product manager, Real-Time Collaboration, at Microsoft. "Microsoft's real-time collaboration architecture for the desktop, featuring Live Communications Server 2005 and Istanbul, when combined with presence-based 3G video telephony from RADVISION, is a powerful step forward towards the industry's vision of true mobility — where you can connect to any one, using any device, through any media including instant messaging, voice, and even video."

"Over the past year we have seen huge interest both by the enterprise IT manager for personal desktop multimedia communications and from the 3G operator for real-time 3G video calling," said Boaz Raviv, general manager of RADVISION's Networking Business Unit. "The integration of these two worlds now enables enterprises and service providers to quickly deploy real-time visual communications networks that spans from the employee's desktop to his or her 3G phone."

The RADVISION SCOPIA 3G Video Gateway is designed to enable PCs with the Istanbul desktop client to connect to 3G handsets. The RADVISION 3G gateway bridges video calls between 3G-324M enabled mobile video phones and PDAs with IP-based video answering machines and other IP (SIP and H.323) and ISDN-based videoconferencing end points. The platform also enables mobile videophones to utilize additional resources on the IP network including multipoint conferencing bridges, which host three or more parties in a single session, voice and video gatekeepers, and terminals.

http://www.radvision.com
http://www.microsoft.com

Sphere Demonstrates New Features, Expands Partner Universe

Sphere Communications announced recently a new release of its Sphericall enterprise telecommunications software, and expanded its list of interopeable endpoints and gateways such as Cisco, Polycom and Sayson IP telephones; AudioCodes and Allied Telesyn media gateways; WiFi-based IP telephony using Spectralink 802.11 IP wireless IP phones; and WiFi-based PDAs running XTen Pocket PC SIP softphone.

"Sphere continues to provide organizations an ever-expanding range of options for their enterprise telecommunications, delivering more flexibility for budgeting and deployments than ever before," said Todd Landry, VP of marketing and product management for Sphere Communications.

Sphericall version 4.1 software, now shipping, offers numerous new features and enhancements, including a communications API for enterprise software integration. This new release further expands the options enterprises have in deploying and customizing IP-based telephony solutions to fit their exact needs and enable tight integration with key business processes.

http://www.spherecom.com

Johns Hopkins Deploys Netcordia's VoIP Module

Netcordia recently announced that Johns Hopkins Bloomberg School of Public Health has deployed its VoIP module as part of NetMRI. NetMRI for VoIP is an automated enterprise network appliance designed to proactively identify difficult to find issues that can significantly affect VoIP quality.

NetMRI's VoIP module determines the actual quality of IP phone calls by evaluating Call Data Records (CDRs) for delay, jitter, and dropped packets. It also correlates abnormalities with other contributory network events, and provides helpful detail to solve the issues. NetMRI's VoIP module pulls CDRs from Cisco's CallManager.

The VoIP module provides assessment of whether the existing IP network can support VoIP, using Cisco's SAA protocol, already built into most Cisco routers and switches, to create 'synthetic transactions' or simulated voice traffic to determine viability based upon the measured delay, jitter, and dropped packet counts.

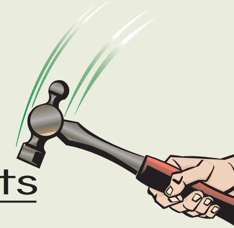
http://www.netcordia.com

Rivermine Targets Growing Enterprise VoIP Market

VoIP services are moving into the corporate mainstream and enterprises are facing new and complex IT business demands. To more efficiently manage the transition process, Rivermine Software announced the VoIP Implementation Kit to address the four phases of deploying VoIP: Business Analysis, Network Analysis and Planning, Implementation, and Ongoing Operations. The kit provides robust migration management tools designed to facilitate corporations' IP convergence. The Rivermine solution was created to help enterprises decrease expenses, maintain services, and mitigate risks.

"As enterprises are thrust into converged networking, it is imperative that organizations adopt a methodical process to their overall deployment," said Mark Logan, Rivermine Software's CEO. 'With Rivermine's rational approach to VoIP, we expect to be a crucial part of the growing enterprise VoIP systems market, which is anticipated to reach $4.2 billion by 2007."

http://www.rivermine.com



Quick Hits

Qovia Announces "Powered By Qovia" Program

Qovia, Inc., recently launched its "Powered by Qovia" program that will make Qovia's products and intellectual property available to select partners who wish to embed Qovia management tools and technologies into their product offerings. . The first participant in the "Powered by Qovia" program is NEC Unified Solutions.

http://www.qovia.com
http://www.necunified.com

AnchorPoint Offers Advanced Analytics

AnchorPoint announced the availability of Advanced Analytics, a new set of business intelligence tools, including dashboard, benchmarking, and ad-hoc reporting, designed to simplify and improve the management of enterprise communications environments. AnchorPoint Advanced Analytics is based on technology from Business Objects and is designed to offer users of the AnchorPoint TFM suite new ways to track, monitor and control communications-related assets and expenses.

http://www.anchorpoint.com

APEX Adds Conferencing To Omnivox

APEX Voice Communications has added Business/Multiparty Conferencing to OmniVox AES, its latest multi- services platform for enhanced services. Key features include a Web-based reservation system, support for resource and time management, and a new group of conferencing-specific command icons. OmniVox AES supports a wide variety of worldwide telephony protocols along with a number of signaling protocols.

http://www.apexvoice.com

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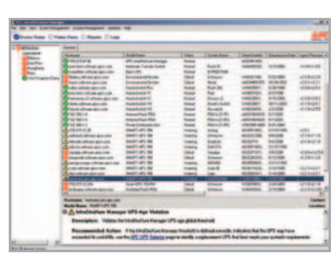

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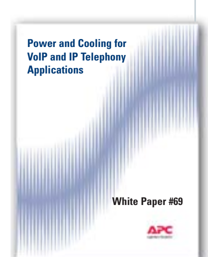

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SBC To Acquire AT&T

SBC Communications, Inc., and AT&T announced today an agreement for SBC to acquire AT&T. The transaction combines AT&T's global systems capabilities, business and government customers, and Internet protocol (IP)-based business with SBC's local exchange, broadband and wireless solutions.

The combined company will have robust, high-quality network assets, both in the United States and around the globe, and complementary expertise and capabilities. The merger is designed to deliver the resources and skill sets to innovate and more quickly deliver to customers the next generation of advanced, integrated IP-based wireline and wireless communications services. For SBC, the combination provides immediate global leadership in the enterprise segment where corporations and governments require complex communication solutions and services and access to advanced national and global networks.

Under terms of the agreement, approved by the boards of directors of both companies, shareholders of AT&T will receive total consideration currently valued at $19.71 per share, or approximately $16 billion. The acquisition, which is subject to approval by AT&T's shareholders and regulatory authorities, and other customary closing conditions, is expected to close by the first half of 2006.

"Today's agreement is a huge step forward in our efforts to build a company that will lead an American communications revolution in the 21st century," said Edward E. Whitacre Jr., SBC chairman and chief executive officer. "The communications industry is undergoing a profound transformation as it transitions to unified, IP-based networks capable of delivering a host of integrated services. To manage this evolution, customers need a partner with the resources to provide new service platforms and product sets, while maintaining world-class reliability and security. This merger creates that company."

Said David W. Dorman, AT&T chairman and chief executive officer, "Together, SBC and AT&T will be a stronger U.S.-based global competitor capable of delivering the advanced network technologies necessary to offer integrated, high-quality, and competitively priced communications services to meet the evolving needs of customers worldwide."

http://www.sbc.com
http://www.att.com

Lucent/Juniper Solution Selected By Shenandoah Telecom

Lucent Technologies announced it is deploying Juniper Networks routing platforms in Shenandoah Telecommunications' network to support the growth of Shentel's DSL business and future advanced services. As part of the agreement, Lucent Worldwide Services will provide Shentel with installation, integration and remote technical support services. Shentel is a Sprint PCS Affiliate.

The Lucent and Juniper Networks solution is designed to enable Shentel to cost-effectively boost the capacity of its IP network to meet increased business and residential demand for its broadband services and deploy the foundation for advanced services such as bandwidth-on-demand to support video-on-demand and IPTV applications.

"Demand for our products and services continue to grow, and Lucent and Juniper Networks provide us with a reliable broadband solution that will help us meet that demand," said Chris French, president of Shenandoah Telecommunications. "This IP routing solution also positions us well for the future as we introduce new revenue generating services."

"Together Lucent and Juniper Networks delivered a secure and assured networking solution that enables Shentel to expand their current DSL service offering, while also providing the capability to support enhanced service offerings such as IP TV in the future," said Christine Heckart, vice president of marketing at Juniper Networks.

Lucent will provide Shentel with Juniper Networks E-series edge routing platform and the M-Series multiservice edge platform. The Juniper Networks E-series will offer Shentel scalable capacity for tens of thousands of users and aggregates IP Traffic from the broadband access network. Shentel will deploy the M10i as the core routing platform in its network, steering all traffic from its data network to the Internet.

http://www.juniper.net
http://www.lucent.com

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VERSATEL NETWORKS

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Last Mile Announces e-Marketplace For Telecom Bandwidth Services

Last Mile Connections (LMC), a New York-based neutral telecommunications exchange dedicated to helping telecom carriers and enterprises access bandwidth in the simplest and most-cost effective way, has launched the Online Exchange, a neutral e-marketplace utilizing reverse auctions for selling and buying telecom bandwidth services. The Online Exchange, is designed to be a powerful customer-focused application designed to connect buyers and sellers of bandwidth from around the globe through a single point of contact, thus dramatically streamlining and speeding the entire sales process and revolutionizing the way the telecom industry conducts business.

"We created the Online Exchange to help those who were inundated with thousands of choices and overwhelmed as to where to go for the best selection of bandwidth services," said James Martino, President and CEO of Last Mile Connections. "E-marketplaces and reverse auctions are used by eight out of 10 Fortune 1000 firms to save millions of dollars each year and increase profitability. Industries where these auctions are used in strategic sourcing include energy, manufacturing, office supplies, IT and more. We believe the time is right for the telecom buyers and sellers to embrace this concept because of the characteristics of telecom services and the many benefits."

According to Martino, the Online Exchange is open to all carriers, service providers and enterprises. The Online Exchange will deliver true market pricing and cut the negotiation and sales process from up to two weeks to 24 hours or less.

The Online Exchange is designed to benefit buyers by guaranteeing them quick and easy access to true market prices from bandwidth suppliers from around the globe, while eliminating the hassle of multiple contacts and lengthy sales processes for daily telecom service needs. Conversely, sellers are offered the opportunity to offer their services to qualified, serious buyers. Last Mile Connections ensures the confidentiality of all involved as the identity of the seller is never revealed to the buyer and buyer identities are kept hidden from other bidders during the entire sales process.

The Last Mile Connections Online Exchange is active today with pre-qualified buyers and sellers requesting and bidding for private lines, Ethernet services, wavelengths, collocation and Internet access, via a secure, confidential e-marketplace.

http://www.lastmileconnections.com

Excel And DaVinci Integration Provide Services To Infiniroute

Excel Switching and DaVinci Integration jointly announced that InfiniRoute Networks a carrier-neutral VoIP interconnect service provider for wireline, wireless, and emerging carriers, has selected Excel's new IMG 1010 media gateway for worldwide interconnectivity and termination of VoIP to support its carrier-grade, converged network solution.

Excel's IMG 1010 is designed to provide gateway functionality for SS7 to H.323 signaling networks and TDM-to-IP interconnections at a value level that complements InfiniRoute's existing SS7 to H.323 interworking capabilities, while enhancing its ability to interconnect and terminate IP-based calls for an expanded range of carriers. In addition, the IMG 1010's 1U form factor allows InfiniRoute to easily add network points of presence (POPs) in a space-efficient, yet manageable way as they build out in cities around the world.

"Our distribution relationship with DaVinci enables us to extend our collective VoIP expertise deeper into our carrier relationships. It further enables us to take advantage of DaVinci's partnerships with best-of-breed technology providers, including Excel, to maximize legacy network infrastructures while providing a path to next generation IP architectures," said Robert H. Turner, Chief Executive of InfiniRoute Networks. "Excel's overall experience in global SS7 protocols and H.323 signaling, combined with DaVinci's VoIP networking expertise truly allows us to extend our cost-effective and scalable worldwide voice network to quickly capitalize on growing VoIP opportunities."

http://www.excelswitching.com.
http://www.davincint.com
http://www.infiniroute.com

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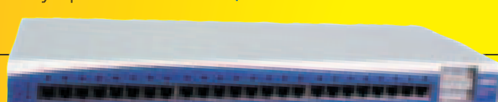

4200512L1
NetVanta 1224R Integrated Switch Router with T1/FT1 Interface and QOS
The Netvanta 1224R is a 24-port managed Layer 2 Ethernet switch with an integral IP router. Its single WAN interface accommodates NetVanta NIMs and DIMs for DSU/CSU functionality and optional dial backup. It supports all NetVanta 1000 Series Layer 2 Ethernet switching, access routing, and system level functionality. It has a 4.8 Gbps non-blocking switching capacity **UNDER $1000!**

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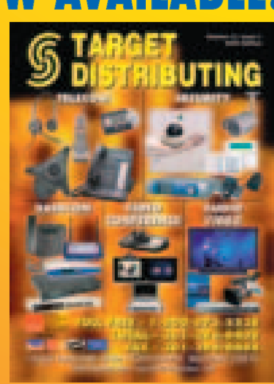

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GlobalTouch Chooses Telco Systems For SIPTalk Service

GlobalTouch Telecom, Inc., has selected Telco Systems' Access211 analog telephone adapter (ATA) for its SIPTalk VoIP calling service.

GlobalTouch CEO, Greg Welch, says, "The Access211 is the perfect device for our SIPTalk customers because it combines a VoIP ATA with built-in router and life line POTS support, all into one device. Life Line POTS support allows the customer the ability to utilize a standard phone line to access E911 service in the event of power or Internet outages."

Telco Systems' Access211 VoIP ATA is designed to enable GlobalTouch SIPTalk subscribers to make and receive phone calls and faxes over broadband Internet connections using standard telephones and fax machines.

http://www.siptalk.com
http://www.telco.com

Teltronics, Inc., Intros Site Event Buffer Devices

Teltronics, Inc., has announced the release of SEB NET-PATH and SEB NET-PATHm. Both Site Event Buffer (SEB) monitoring devices are the latest in Teltronics' Intelligent Systems Management products designed for monitoring legacy telecom systems and IP-based datacom elements — serving as an SNMP Proxy Agent or a Network Segment Manager, providing ASCII Datastream analysis, monitoring environmental sensors, controls, and more.

The SEB NET-PATH is is designed to addresses virtually any monitoring requirement, while SEB NET-PATHm has a reduced size and feature set to provide a cost-effective alternative for monitoring solutions not requiring the full-featured SEB NET-PATH unit. Both are fully integrated with IRISnGEN, Teltronics' centralized alarms management collection system and together they provide the customer a comprehensive suite of management tools available for highly reliable, centralized monitoring, management and maintenance of remote devices or systems. In addition to their alarm event detection and reporting, both systems are suitable for data collection and polling, and secure access for remote devices.

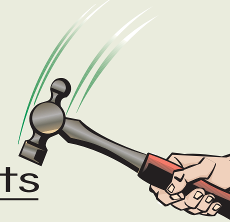
http://www.teltronics.com

SunRocket Selects Netrake Session Controllers

Netrake recently announced that SunRocket, the 'no gotcha' phone company, has selected Netrake to support SunRocket's residential flat-fee VoIP phone service. According to the company, Netrake's session border controller was selected based upon its ability to satisfy SunRocket's immediate and future requirements as it introduces new VoIP services. The ability to seamlessly support Hosted Firewall and NAT traversal for offering new, advanced IP services to customers behind enterprise and residential firewalls was critical to the selection process.

"The proven scale and five-nines reliability of nCite's security, quality of service and network management components support SunRocket's goal of delivering feature-rich, easy-to-use, 'no gotcha' phone service to consumers throughout the United States," said SunRocket chief operating officer Rob Mainor.

http://www.netrake.com
http://www.sunrocket.com



Quick Hits

Volo Completes Fixed Wireless VoIP Testing

Volo Communications has successfully completed a month-long test of VoIP services over fixed wireless with Blue Wave Networks, who is now offering VoIP services commercially using Volo's VoiceOne broadband voice services solution. Blue Wave Networks, who offers residential and business customers a variety of Internet access and data services, is now able to expand its service offering to include nationwide voice services.

http://www.volocommunications.com
http://www.bluewavenetworks.com

Microsoft Launches Connected Services Framework

Microsoft announced availability of the Microsoft Connected Services Framework, an integrated software solution designed to allow telecommunications operators and service providers to deliver converged communications services across multiple networks and a range of device types. The goal is to allow wireless and wireline operators to reduce the cost and time of service deployment, more efficiently utilize existing infrastructure, and generate additional revenue through new services.

http://www.microsoft.com

Verizon To Acquire MCI

Verizon Communications, Inc., and MCI, Inc., announced that Verizon has agreed to acquire MCI for $4.8 billion in equity and $488 million in cash. The transaction is designed to add new strength to the telecommunications services both companies provide. It ensures that consumers and businesses will have a supplier with the financial strength to maintain and improve MCI's Internet backbone network.

http://www.verizon.com
http://www.mci.com

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*Requires MAX EX Base for each room (Part No. 910-158-941).

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Itronix Selects Sierra Wireless EM5625 Embedded Module

Sierra Wireless and Itronix Corporation today announced that Itronix has selected the Sierra Wireless EM5625 embedded module to provide its GoBook series wireless rugged notebooks access to CDMA 1xEV-DO networks. The dual-band EM5625 embedded module will enable users to connect to e-mail, the Internet, and information residing on corporate servers, including data-heavy applications like streaming video and large graphics files, using one of the fastest wireless wide-area network technologies currently available.

"Based on our past experience with using earlier Sierra Wireless embedded modules in our GoBook series laptops, we are confident in our choice to migrate to the next-generation EM5625 embedded module," said Matt Gerber, Senior Vice President, Product Line Management for Itronix. "Sierra Wireless offers the expertise and support we need to integrate the wireless functionality seamlessly into our products, letting us concentrate on getting to market as quickly as possible."

"Original equipment manufacturers put a lot of trust in our embedded modules when they integrate them into their own products, and that is something we don't take lightly," said Dan Schieler, Senior Vice President of Sales, Americas and Worldwide OEM for Sierra Wireless. We are pleased to work with Itronix to deliver EV-DO connectivity in the GoBook III notebook computer and the full series of GoBook notebooks."

The dual-band Sierra Wireless EM5625 embedded module provides a powerful, reliable, wireless engine that can be integrated into a wide variety of mobile devices and offers "always-on" voice, data, and messaging functionality at typical data speeds of 300–500 kbps, with bursts up to 2.4 Mbps. Additional features of the EM5625 module include enhanced power management to optimize battery life, as well as gpsOne capability, which complies with E911 requirements for emergency location. Sierra Wireless also offers a comprehensive suite of integration services for original equipment manufacturers (OEMs), including a Universal Development Kit to simplify the integration process and accelerate time-to-market.

http://www.itronix.com
http://www.sierrawireless.com

TeleSym Announces Mobile VoIP Solution Support For Symbol MC50

TeleSym, recently announced support for the Symbol MC50 enterprise digital assistant (EDA) with its SymPhone solution. The SymPhone solution is designed to allow employees to make and receive secure calls over a corporate wired/wireless LAN or via the Internet. Incorporating both patent pending call quality technology and RSA security features, SymPhone also extends the enterprise PBX to mobile workers and professionals.

"Symbol's MC50 is designed for mobile professionals seeking to capture, move and manage information at the point of business activity in order to increase productivity and efficiency, while improving customer responsiveness and satisfaction," said Doug Lloyd, director of mobile computing, Symbol Technologies. "With SymPhone's validated voice solution, the MC50 is an even more powerful and versatile tool enabling on-the-spot decision making with real time access to people and information in a mobile computer — especially when combined with an organization's data applications."

Introduced in October 2004, the MC50 is specifically designed for business-essential applications within the enterprise environment with a particular focus on durability and life-cycle management. Featuring industry-standard-based and Symbol wireless security capabilities, the MC50 also includes integrated voice capability with a designed-for-voice electrical and mechanical design. With the MC50, Voice over IP (VoIP) solutions can be deployed enabling in-building and on-campus communications for real-time responses and improved customer satisfaction.

http://www.telesym.com
http://www.symbol.com

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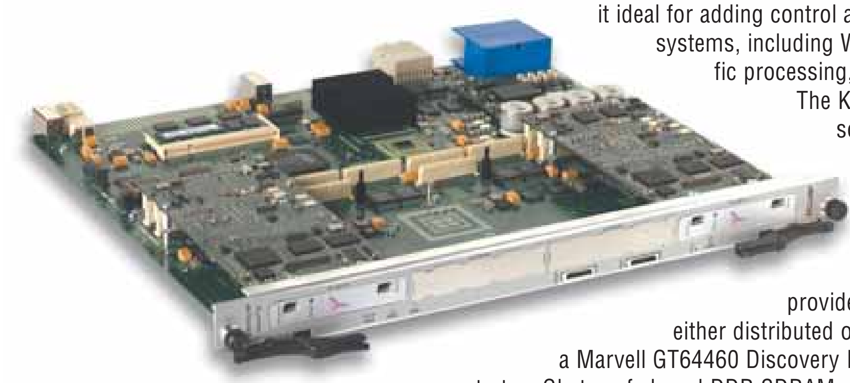
Artesyn Announces ATCA Packet Processing Blade

Artesyn Communication Products, has announced the KatanaPPB. The KatanaPPB features up to six PowerPC processors, a versatile multiprocessor interconnect with separate control and data planes, and redundant IPMI-based system management interfaces. The KatanaPPB also features a high-speed PICMG 3.1-compliant ATCA interface with ten Gigabit Ethernet channels.

"ATCA is the consummate open architecture telecom platform, and KatanaPPB is the industry's fastest ATCA telecom blade," said Todd Wynia, vice president of marketing at Artesyn Communication Products. "Katana's six PowerPC processors, versatile data flow architecture, and high-speed switched fabric make it ideal for adding control and packet processing power to a wide range of telecom systems, including WAN access, SS7/SIGTRAN signaling, media gateways, traffic processing, wireless base station controllers and softswitches."



The KatanaPPB features six Freescale PowerPC 7447A processors, each equipped with an Altivec vector processing unit, 32 kbytes of on-die L1 instruction/data cache, 512 kbytes of on-die L2 cache, and cache coherency mechanisms for symmetric multiprocessing. The six processors are divided into two complexes.

Complex one, mounted directly on the baseboard, provides a pair of MPC7447A processors, which can be used for either distributed or symmetric multiprocessing (SMP). The processors share a Marvell GT64460 Discovery III system controller, which provides high-speed access to up to two Gbytes of shared DDR SDRAM memory and 128 Mbytes of flash memory. Complex two provides four MPC7447A processors. Each processor, mounted on a separate PrPMC (ProcessorPMC) mezzanine card, is equipped with a dedicated Discovery III system controller, two Gbytes of DDR SDRAM and 64 Mbytes of flash memory.

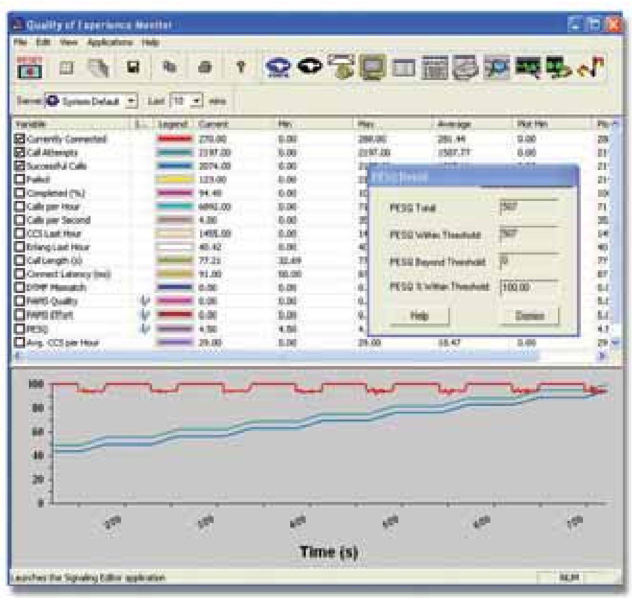
OEM pricing for the KatanaPPB equipped with a pair of MPC7447A processors starts at $4,407.

http://www.artesyncp.com

Empirix Speeds Enterprise VoIP Deployments

Empirix, Inc., a provider of integrated test and management solutions for VoIP, contact center,and Web environments, today announced a comprehensive solution designed to ensure successful deployment of enterprise VoIP infrastructure and applications. The Hammer VoIP Test Solution for Enterprises was created to allow enterprises to reduce risk and speed the rollout of VoIP services and IP telephony applications (such as messaging, speech self-service, conferencing, and CTI) by accurately assessing how their infrastructure and applications will perform as a live, enhanced IP service.

Traditionally, enterprises deploying VoIP have performed network readiness assessments and/or used test tools supplied by VoIP equipment vendors. By enabling enterprises to test applications running on VoIP networks in their own environments, Empirix provides a comprehensive approach, and the capability for enterprises to perform upgrade and migration testing on an ongoing basis. In addition, Hammer VoIP Test Solution for Enterprises can be used to analyze and troubleshoot both pure VoIP networks and hybrid IP/TDM environments.



Jessy F. Cavazos, Industry Analyst and Communication Test Sector Leader for research firm Frost & Sullivan, noted, "Empirix has been testing VoIP technology since it first emerged in labs years ago — extending its capabilities into the enterprise is a natural next step."

The Hammer VoIP Test Solution is comprised of three elements:

- Hammer FX-IP, a feature test platform for IP environments that generates test IP calls and evaluates voice quality;
- Hammer CallMaster, a graphical scripting and reporting tool for creating test call flows; and
- Hammer Call Analyzer, a diagnostics and troubleshooting solution that enables users to visualize and debug signaling and voice quality problems in VoIP networks.

http://www.empirix.com/e-voip

Texas Instruments, Wintech Launch Videophone Development Platform

Texas Instruments Incorporated and Wintech Digital System Co. released the Videophone Development Platform (VDP), a development platform for designing point-to-point IP-based videophone systems. The VDP is designed to enable developers to roll out real-time, high-quality consumer videophone systems on TI's digital media processor, substantially lowering system cost and reducing development time.

Traditional barriers to adoption of videophones include jerky video quality and high unit cost. However, the introduction of new video codecs which require half the bandwidth for video transfers, the continuing penetration of IP broadband connectivity and the availability of single-chip encode and decode implementation have overcome these last-mile barriers. The increasing market demand for IP consumer videophones with superior audio and video quality is driving the advent of new and improved systems.

The VDP from TI and Wintech is an integrated hardware/software development platform reducing both design complexity and total system bill of materials, including everything developers need to begin designing point-to-point videophone systems immediately. All application system software runs on TI's 600 MHz DSP-based TMS320DM643 digital media processor, including audio/video compression, networking stacks and control protocols.

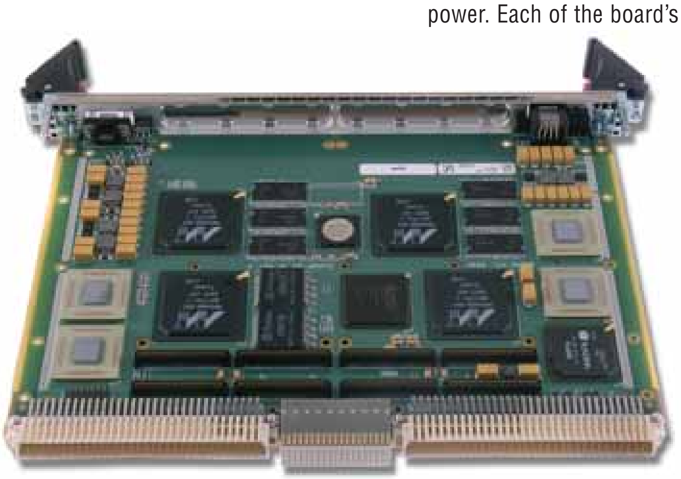
"The VDP provides developers with the tools necessary to create viable and commercially ready IP-based video client devices," said David Dong, president, Wintech. "Instead of spending months integrating different software algorithms, developers begin the development process starting with high-quality video and tight audio/video sync, enabling them to focus on product differentiating features, such as the user interface and out-of-box experience."

http://www.ti.com/vdppr

Quad 1.5 GHz PowerPCs Drive New VME DSP Board From Curtiss-Wright

Curtiss-Wright Controls Embedded Computing (CWCEC) has announced a new high-performance quad processor DSP VMEbus board. The CHAMP-AV IV features quad 1.5 GHz FreeScale MPC7448 PowerPC processors, doubling the processor performance of its predecessor, the CHAMP-AV II, while reducing cost by almost 1/3. This AltiVec-based DSP engine also features CWCEC's industry leading QuadFlow architecture, and builds on the success and popularity of the previous generation CHAMP-AV II and -AV III products.

The CHAMP-AV IV drives DSP applications with up to 48 GFLOPs of peak computational power. Each of the board's four processing nodes consists of a 1.5 GHz 7448 processor, 256MB or 512MB DDR-250 SDRAM (2GB/sec), and dual 100 MHz 64-bit PCI-X interfaces. It is also available configured with quad 1 GHz MPC 7447s. To provide maximum multi-processing application performance, the board's QuadFlow architecture supports four simultaneous node-to-node transfers to deliver a total data flow of 3.2GB/sec peak. Each of the four nodes transfers data to adjacent nodes at speeds up to 1.6GB/sec peak.



Each processing node of the AV IV is provided with a Gigabit Ethernet connection. System expansion on the CHAMP-AV IV is supported with two PCI-X interface compatible PMC sites. Each expansion site supports the latest PCI-X specification, and enables 64-bit/100MHz operation, while retaining compatibility with legacy 33MHz and 66MHz PMC modules. The AV IV's total 1.6 Gbyte/sec peak of I/O bandwidth provides the throughput required to match the DSP processing power of its quad 7448s.

http://www.cwcembedded.com

Centrepoint's TalkSwitch® Ships With SIPconnect Interface

Centrepoint Technologies, recently announced the full compliance of the TalkSwitch 48-CVA with the SIPconnect Interface Specification. Centrepoint customers will be among the first to take advantage of the benefits of this new specification — a best practices document that guides interoperability between IP PBXs and VoIP service providers.

Centrepoint's TalkSwitch is a customer installable and configurable telephone system designed for business with up to 32 phone users per location. It offers combined access to the public switched telephone network (PSTN) and Voice over IP networks.

"Centrepoint's compliance with the specification means that our customers are able to jump to the front of the line to receive the benefits that an end-to-end IP connection to a service provider network brings. It saves money and offers a dramatically higher quality of service, it also opens the door to a host of enhanced features," said Jan Scheeren, President and CEO, Centrepoint Technologies. "We've recognized that more and more small businesses are deploying IP PBXs. A consistent method for integrating them with a VoIP service network is crucial."

TalkSwitch is built on a flexible SIP-based architecture that was designed from the outset to support integration with VoIP network services.

In conjunction with the SIPconnect initiative, Centrepoint has worked with Cbeyond Communications to carry out customer testing of the TalkSwitch 48-CVA with Cbeyond's "BeyondVoice with SIPconnect" service offering. Centrepoint customers can easily connect their systems with Cbeyond's network without investing in any additional equipment.

The SIPconnect Interface Specification defines a common set of implementation rules for directly integrating an IP PBX with the network of a SIP-enabled VoIP service provider. The specification builds on the SIP and VoIP protocols published by the Internet Engineering Task Force (IETF).
http://www.talkswitch.com
http://www.sipconnect.info

Navtel Platform Now Offers SIP Expert Analysis Tool

Navtel Communications Inc., a developer and provider of test equipment for the laboratories of network equipment manufacturers and operators have announced the immediate availability of "SIP Discovery." This SIP expert analysis tool is designed to set new standards of excellence for users who require the most complete, automated, easy to use SIP testing solution available on the market today.

Providing extensive statistics on the status of a network, or any entity that composes this network, as well as all the necessary troubleshooting information in an intuitive graphical user interface, SIP Discovery is the only SIP expert analysis tool that significantly reduces fault identification/resolution cycles, ultimately saving costs to both carriers and equipment manufacturers.

Capable of measuring any device's load and message transfer/response latency over long periods of time, SIP Discovery will prove itself extremely useful to network planners and Q&A teams. The call tracing, graphical representation of captured call flows and packet processing, used to determine and display each session's status [Initiating, Incomplete, Failed, Established, or Cleared] provides installation, maintenance and Q&A teams with pre-analyzed data, saving long hours of browsing through thousands of lines of raw traffic.
http://www.navtelcom.com

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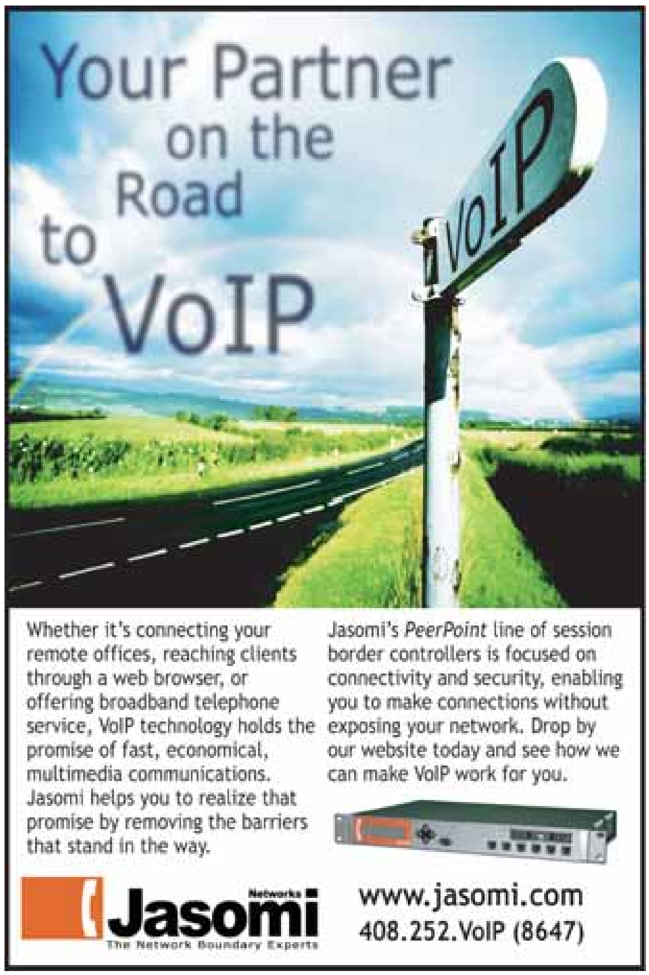
Amdocs Reveals Amdocs 6 Product Portfolio

Amdocs recently launched Amdocs 6, a pre-integrated portfolio of modular billing, CRM, self-service, order management, mediation, and content revenue management software products designed to ease and accelerate adoption of an integrated customer management strategy for telecommunications companies. The ultimate goal is to deliver an intentional customer experience that creates stronger, more profitable customer relationships.

"The concept of an intentional customer experience is one that can create significant differentiation in the communications industry," said Rob Rich, executive vice president of the Yankee Group. "Companies who can help service providers break down silos and align their organizations around an integrated customer management strategy, adapt business and operational systems to become more agile, and place the customer at the center of the business will create exceptional business value for those service providers. There are very few companies with the breadth of products, professional services, industry know-how and partnerships to make this happen."

http://www.amdocs.com

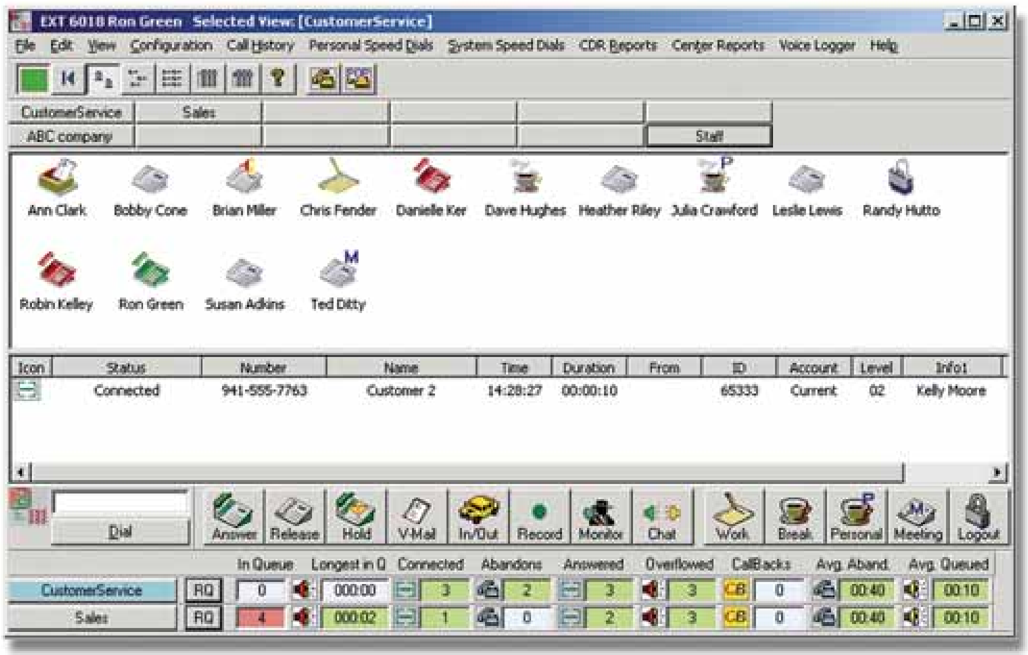


Comdial Releases Solution for Small To Midsize Contact Centers

Comdial Corporation has announced a new solution for the small-to-midsize business contact center market. The CONVERSip Contact Center is designed to offer the tools to manage external call volumes and improve the efficiency and performance of employees.

The CONVERSip Contact Center, an integrated communications suite consisting of skills-based routing, unified voice messaging, call recording, and historical and real-time reports, is designed to efficiently capture all incoming calls, process high call volumes, and manage calls to deliver high levels of service without any additional software at an affordable cost for inbound, outbound or blended contact centers.

This solution assists management personnel in maximizing their resources by providing easy-to-use, real-time reporting tools that graphically demonstrate call volumes, queue times, productivity levels, and agent activity. In addition to its advanced reporting capabilities, one of the CONVERSip Contact Center's unique features is a new call back feature, which allows callers to leave a message to hold their place in the queue without remaining on the phone. Callers are then automatically contacted by an agent when it is their turn in the queue.
http://www.comdial.com



Digisoft Releases Telescript 5.6 Call Center Software

Digisoft Computers, Inc., has just announced the release of Telescript 5.6. The new version includes increased integration with MS Outlook, additional Do Not Call list capability and telephony server enhancements designed to improve automated outbound dialing productivity.

Designed to operate with myriad products, new integration with MS Outlook enables callbacks that are set in Telescript to automatically appear in the Outlook calendar with reminders set. This provides added assurance that callbacks are properly handled in a timely manner.

Increased Do Not Call capability within the application now enables the call center to disqualify an entire exchange rather than importing each number individually. Telescript can now expand the list from a single exchange to include all numbers within the exchange for the Do Not Call queries.
http://www.digisoft.com

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Polycom Intros Service Partner Certification Program

Polycom has expanded its core channel programs with the introduction of the Polycom Certified Service Partner (CSP) program. The CSP program certifies Polycom's best-in-class service and support channel partners by verifying their performance in providing customers with 24x7 support, fast response times, call center support, and stringent training in Polycom solutions and IP networking.

The CSP program is unique in requiring CSP certified partners to maintain a high level of customer satisfaction in order to achieve annual recertification. Polycom and its CSP partners are jointly able to offer maintenance and diagnostic service and support. This approach is also unique among industry support programs for its tight integration of vendor and channel partner support, which assures the customer the right skill set and best practices for each stage.

"Real-time conferencing and collaboration technology is a mission critical application for many customers and becoming an integral part of our customers' core IP network environments," said Geno Alissi, vice president and general manager of global services at Polycom. "The Polycom Certified Service Partner program ensures our certified channel partners are able to deliver the high level of expertise, service and support that customers demand for their core business applications."

http://www.polycom.com

NetFabric And Williams Bring VoIP To Small Offices

NetFabric Corporation has announced the signing of a master distribution agreement with Williams Telecommunications Corp. of Ontario, Canada for resale of its FUS1ON line of Intelligent Call Directors which are designed to enable small offices to quickly and cost-effectively adopt VoIP services.

Williams Telecommunications Corp. distributes products from manufacturers such as Nortel, Lucent, Avaya, Mitel, Nitsuko, TIE, Vodavi, GN Netcom, Panasonic, as well as a complete line of peripheral products. From telephones, voice mail systems, and headsets, and more, Williams Telecommunications serves the needs of businesses of all sizes.

FUS1ON provides for reliable access to VoIP Service Providers, maintains the quality of the phone service business customers demand and maintains reliable access to E-911. It is also a platform for new telephony applications such as automatic Client Record Pop-Up and Call Accounting. FUS1ON can be installed with any PBX, KTS or even discrete phones in minutes.

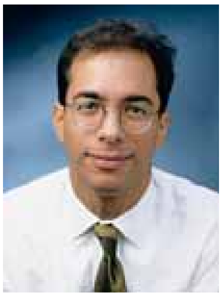
http://www.williamsglobal.com
http://www.netfabric.net

Critical Telecom Announces Global Agreement With Ericsson

Critical Telecom Corp. today announced a definitive, multi-year global agreement with Ericsson, under which Ericsson will integrate Critical Telecom's GEmini Remote Ethernet DSLAM into its global Ethernet DSL Access (EDA) broadband portfolio to help service providers deliver such new generation broadband access services as IPTV.

GEmini is a scalable Ethernet-based remote DSLAM. Based on the company's new generation architecture, GEmini is the first of a new class of product that is purpose built for the marriage of fiber and copper anywhere in the access network enabling the cost-effective delivery of high bandwidth services. The platform provides non-blocking ADSL2+ bandwidth using standards-based Gigabit Ethernet in a physically small shell that is environmentally hardened.

http://www.criticaltelecom.com
http://www.ericsson.com



By Marc Robins

SIPconnect: Taking VoIP To The Next Plateau

VoIP (define - news - alert) network solutions that use gateways and other equipment to enable IP networks to interface with TDM networks have been essential in the evolution of VoIP and enhanced IP communications. Beyond any doubt, these bridging solutions between legacy networks and new IP networks have greatly contributed to the growing deployment of IP PBXs and other IP-based communication solutions.

At the same time, however, one of the major drawbacks of today's next-generation communications architecture is its reliance on offloading packets to traditional TDM networks at various points during a transmission. While this is necessary to interface with analog equipment at termination points, it is also a limited approach to next-generation IP telephony where a pure end-to-end IP connection is possible.

Indeed, routing VoIP traffic over TDM networks introduces a number of QoS issues, such as latency and echo, and is inadequate for supporting the full capabilities of VoIP. Intelligent end-user identification information can be lost, and network issues such as quality of service (QoS) and application layer security are not consistently addressed, which ultimately limits the ability of service providers to deliver the rich set of features enabled by IP-based communications, and to be able to extend these capabilities beyond the enterprise and over the wide-area network to some other remote end point.

A much more efficient and cost effective approach, which also enables the full capabilities of packet-based communications, would be to enable IP PBXs to peer, or connect directly, with VoIP service providers, eliminating the need for gateways and TDM traffic routing altogether. Of course, to accomplish this, the equipment and service providers must use common standards so that direct IP peering may be accomplished.

Enter The SIPconnect Interface Specification

For most of the industry, SIP (define - news - alert) is now considered the logical choice for routing a variety of IP traffic, and the protocol already defines ways for accomplishing interconnection between SIP-enabled IP PBXs and service providers' SIP-enabled networks. However, these methods do not address the entire scope of interconnection such as required features, security, and quality of service issues, and do not offer a common, predictable solution for retaining intelligent end-user identification throughout a network transmission.

The draft SIPconnect Interface Specification, launched by managed service provider Cbeyond Communications with support from equipment vendors Avaya (quote - news - alert), BroadSoft (news - alert), Centrepoint Technologies (news - alert), Cisco Systems (quote - news - alert), and Mitel Networks (news - alert), seeks to address these thorny problems. SIPconnect defines a method for interconnection between IP PBXs and VoIP service provider networks that builds on SIP and other VoIP protocols, and specifies a reference architecture, required protocols and features, and implementation rules necessary for seamless peering between IP PBXs and VoIP service providers. Importantly, these recommendations use and build upon guidelines published in RFCs dealing with the SIP family of protocols. The SIPconnect guidelines are intended to compliment (and potentially guide implementation) of these RFCs, and not to replace them in any way.

The Benefits Of Direct IP Peering Using SIPconnect

SIPconnect offers a number of important benefits to IP PBX vendors, VARs and Interconnects, service providers, and business customers, and stem from the following benefits of a direct IP peering approach. These benefits include:

Universal Approach. SIPconnect fills a current void in the industry by providing clear instructions for IP peering between IP PBX and VoIP service providers. This will accelerate adoption and reduce development costs for PBX developers and service providers.

Customer Cost Savings. Direct IP peering makes VoIP gateways unnecessary, and extends the benefits and savings of VoIP communication systems (i.e., DID and conferencing) into the network and beyond to compliant destinations.

Transparent Feature Transport. Individual end-user information is passed from the IP PBX to the network intact (and with application layer security intact) rather than being lumped into one account. This enables important information for presence and other user-based applications to travel through the network to terminating IP PBXs without being stripped out.

Quality of Service. Important transport layer issues are defined, including: QoS configuration, echo cancellation, method for DTMF relay, packetization rates, codec support, and dealing with fax and modem traffic.

Security. Service providers act as the public interface (for DNS queries, etc.) for communicating with SIP devices via their SIP proxies (i.e., a session border controller). This enables them to add security at the application layer for customer communications.

Join The Initiative

While Cbeyond is making the draft SIPconnect Interface Specification public in order to gain industry-wide approval of the interface and service, the next step is up to the industry at large. Ideally, more equipment vendors and service providers alike will contribute to the development of SIPconnect, resulting in a working standard that will benefit the entire industry.

To learn how your company can become part of the SIPconnect initiative, and to access the complete draft SIPconnect Interface Specification and related documentation, visit the SIPconnect Web site at http://www.sipconnect.info. **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.

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By Tony Rybczynski & Edwin Koehler, Jr.

IP Video Surveillance

World events have raised the societal need for increased electronic surveillance in various forms, including video. The requirement for video surveillance does not stop at homeland defense. For example, many metro areas are installing traffic monitoring systems and multi-angle street corner cameras for both traffic control and law enforcement purposes. Furthermore, recent events within our schools have highlighted the need for these types of capabilities in schools. Sadly, our society is at a point where these systems are deemed a requirement even at the expense of some loss of privacy.

Fortunately, from a deployment perspective, the implementation costs of video surveillance systems have been dramatically reduced due to three major technology developments.

Going Digital, The PC Platform, And IP Networking

The first technology to help drive the cost of video surveillance is digital imaging. Traditional analog systems required dedicated broadband facilities (e.g., coax) for transmission, and magnetic tape for storage. The move from analog to digital not only reduces the amount of data required to transmit the captured video through compression, but also creates a very portable and limitless storage format that is ageless. This provides two financial benefits, the amount of networking bandwidth to move the content is reduced and the cost of maintaining analog tape banks and repositories is reduced or even eliminated.

The second technology change that has had a beneficial impact on the cost and functionality of video surveillance systems is the use of the PC platform. PC architectures have become sufficiently robust and powerful to handle video surveillance system requirements with ease. The marriage of high-quality video-capture hardware and PC-based application software creates a perfect implementation for the use of digital video technology.

The third technology change is in the way that the systems communicate; this is quite possibly the biggest area of improvement in the implementation costs. Traditional systems moved the surveillance data back to the security console via dedicated coax-based infrastructures and analog head-ends. As the move to PC-based platforms came about, the next obvious step was to incorporate PC-based networking for the connection and relay of surveillance data. The dominant protocol for PC communication is without a doubt IP. IP-based communications allow for the use of open systems digital routing and switching technologies that comprise the Internet today. These systems are much less expensive than traditional coax head-ends. The largest benefit of using IP however is not in the significant cost savings, but more so in the additional capabilities that such a system would enable. For example, low cost IP-enabled cameras are widely available, providing wired (with power over Ethernet) and wireless access directly to a client viewer or to a centralized security desk.

Video Surveillance And Convergence

Open PC platforms on both the server and the client side allow for the merging of different applications into the overall security solution. Instant messaging, real-time bi-directional video conferencing, e-mail and Web access for documents and images are all examples of applications that can enhance the power of the site security solution.

PC platforms also allow authorities to address major challenges of video surveillance systems: establishing the context of captured information. Information itself without context is useless; information in the wrong context can be misleading. The correlation of information is one of the most important aspects in crime prevention and forensic analysis. If events cannot be properly identified and sequenced in the data search, then the value of critical captured data may be lost. In order to be most effective, this correlation needs to happen as quickly as possible. Open PC platforms and IP networking allow for the use of storage networking technologies that have vast capacities and allow information to be retrieved on demand. This capability, when combined with object data models and relational database technology, provides the right knowledge base platform. When applications such as face recognition are incorporated into the data model, the resulting surveillance system incorporates the ability to match search criteria with the captured video information.

Securing And Hardening Video Surveillance

IP networking has evolved significantly in the aspects of robustness and security. High-end highly-critical video surveillance systems will be architected in a three-tier design: a front-end data capture IP network running over dedicated Ethernet; secure access to a MAN/WAN enterprise network and the Internet; and a data processing and storage layer. Resilience can be provided at each level to provide the required level of redundancy with a view of fast recovery to achieve 99.999 percent reliability comparable to the phone network.

Video surveillance is exploding, accelerated through video digitization, PC platforms, and low-cost cameras and IP networking. More importantly, convergence has opened the door for increased functionality to allow enterprises to better leverage their investments in video surveillance. IT

Tony Rybczynski is Director of Strategic Enterprise Technologies at Nortel. He has over 30 years experience in the application of packet network technology. Ed Koehler is a senior architect with the Enterprise CTO group, and is focused on multimedia applications and supporting technologies. For more information, please visit http://www.nortel.com.

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By John Cimko

Brand X Heads To The Supreme Court

Brand X Internet Services v. FCC, decided in 2003 by a panel of the Ninth Circuit U.S. Court of Appeals and now scheduled for review by the U.S. Supreme Court, is a strange and positively bizarre result.

At least that's the view expressed by Judge Diarmuid O'Scannlain in his concurring opinion in the case.

Before examining the basis for Judge O'Scannlain's verdict about the court's controversial decision in *Brand X*, here's a bit of history.

The *Brand X* saga involves whether cable modem services providing broadband Internet access should be treated as regulated telecommunications services or unregulated information services. It began in 2000 when the Ninth Circuit issued a ruling in *AT&T v. City of Portland*, deciding that the broadband pipe used by cable modem services constitutes a telecommunications service subject to common carriage regulation.

Two years later the FCC (which had not previously ruled on the regulatory classification of cable modem service) adopted a declaratory ruling in conflict with *Portland*. The FCC held that cable modem service fits within the Communications Act's definition of interstate information services, and therefore is not subject to state or federal economic regulation.

The FCC decision set the stage for a quirky plot twist. Appeals against the FCC decision were filed in three different federal circuits, thus activating random selection procedures overseen by the Judicial Panel on Multidistrict Litigation. The Panel spun the wheel and the arrow landed on — you guessed it — the Ninth Circuit. This led to a showdown between the Ninth Circuit's *Portland* decision and the FCC's ruling, with *Portland* having home court advantage.

Ordinarily, the FCC's decision would be reviewed by an appeals court under the Supreme Court's two-step formula in *Chevron USA v. NRDC*. Step one: Is the statute being applied by the agency clear on its face? If it is, then the agency is required to adhere to the expressed intent of Congress. Step two: If the statute is silent or ambiguous about the issue in contention, the reviewing court must give deference to the agency's interpretation of the statute. The only issue for the court to decide is whether the agency's decision is based on a reasonable statutory construction. Thus, under *Chevron*, if the agency's interpretation of the statute is reasonable, the agency's action must be upheld by the court, even if there could be other reasonable, but different, readings of the law.

But the Ninth Circuit, because of its previous decision in *Portland*, decided not to apply *Chevron's* deferential standard of review to the FCC's ruling. The court concluded that *Portland*, holding that the transmission component of cable modem service is a telecommunications service, was controlling precedent in the Ninth Circuit and required the circuit to vacate the FCC decision without even getting to the *Chevron* test. The FCC and the Department of Justice then appealed the Ninth Circuit decision to the Supreme Court, the Court took the case, and a decision is expected later this year.

Judge O'Scannlain calls the *Brand X* decision a strange and bizarre result because it is "strikingly inconsistent" with the underlying principles of *Chevron*. The *Chevron* rule is that Congress gave the agencies, not the courts, the discretion to interpret ambiguous statutes because the agencies have the requisite expertise. But *Brand X* subverts this rule by holding that, if a court acts first to resolve a statutory ambiguity, a subsequent agency effort to apply its expertise to an interpretation of the statute will be disregarded in favor of the court's precedent.

It is true, of course, that there can be disagreement about the FCC's policy judgment in deciding that cable modem service is strictly an interstate information service and is thus freed from restrictive state and federal regulation. Consumer groups (as well as state and local governments) have opposed the ruling, arguing that the FCC's action will enable cable companies to choke off competitive services offered by other Internet service providers. This, they say, will drive up prices and narrow the choices available to consumers.

Interestingly, incumbent local exchange carriers that compete against cable companies in providing Internet access generally support the FCC decision that cable companies should not be regulated. Some industry observers speculate that the incumbent carriers expect the FCC to provide similar deregulatory treatment for their provision of Internet access, and that the carriers have decided they would be better off if the two behemoths — cable and the incumbent LECs — can square off without any regulatory interference. Whether consumers would ultimately benefit from this unregulated "intermodal" competition is anyone's guess.

In the meantime, however, it appears that the Supreme Court, in accepting the FCC's request to review the *Brand X* case, has signaled its intent to overturn the Ninth Circuit. It's hard to disagree with Judge O'Scannlain's view that the strange and bizarre *Brand X* decision stands *Chevron* on its head by allowing the courts to cut in front of agencies in resolving statutory ambiguities. The Supreme Court may be preparing to push the appeals courts back to the end of the line, which is the position *Chevron* reserved for them. **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

ENUM: Let The Market Decide

Today, we will take a look at a few commonly known ENUM (define - news - alert) registries, examine how they operate, and compare their differences. As it is with many things, one particular application of a technology might suit you better than another. There are many reasons for this, one being risk tolerance. The tolerance is usually directly related to the level of expertise any given organization has with a technology. The greater the expertise level, the greater the comfort, the more willing the organization is to taking risks. With risks, there are certainly rewards, and for the "early adopters" a.k.a. knowledgeable, calculated risk takers, the rewards are great.

We will be taking a look at the VeriSign IP Connect Suite, e164.org, and The Voice Peering Fabric. These companies have different networking backgrounds, which have impacted the way they architected each of their services. Since this is so new there is room for the different uses and implementations of ENUM, but ultimately the market will decide what makes the most sense.

VeriSign IP Connect Suite

VeriSign offers its IP Connect Suite as a means to set up and resolve VoIP calls. The service includes directory (mapping telephone numbers to IP addresses), security (secure authentication and firewalling), and interoperability (mediating H.323 and SIP platforms). The security and interoperability require VeriSign CPE to be deployed to the customer premise. The directory service can be used with or without the CPE.

The directory service is supported by a single data store for location that includes ENUM, SIP and SS7 lookups on that store. The location information is the same — the protocol used to access it results in a different format. The customer can have a single directory service for all of its devices, regardless of whether they support ENUM, SIP, or whatever. VeriSign views ENUM as a protocol for directory access, not as a platform or directory in itself.

VeriSign states that the ENUM service is both multi-lateral and bi-lateral, meaning providers can peer with all members or only in selectively established situations between two parties (end user enterprises, or carriers). In either scenario, it is the responsibility of the customer to establish the network connections (public or private) for the actual traffic flow to complete the peering relationship. In order for any customer to use the ENUM service, they must have their ENUM numbers loaded in to the VeriSign central data store. For this ENUM service, VeriSign charges an annual subscription fee per number.

The VeriSign CPE-based services act as a protocol and codec conversion gateway handling media and signaling traffic. There is a monthly fee from VeriSign for the CPE. The customer is responsible for establishing the connectivity between the CPE and VeriSign. VeriSign is carrier agnostic and does not require a specific backbone provider. The service will accept IP over either a private layer 2 connection or a public IP connection. A dedicated layer 2 link to the data store for signaling does not actually carry the voice traffic. For that, the customer must use a separate interconnection method. It is possible for the customer to use a public Internet connection for both signaling and media.

VeriSign is a public company and has many service lines including managing the DNS lookups for the public Internet.

No matter which implementation fits your comfort level, there are some very compelling reasons to use this technology.

e164.org

The e164.org ENUM service is multi-lateral, non CPE based and free for anyone who wishes to use it. The access method is via the public Internet to the e164.org ENUM database. Users are required to load their ENUM numbers in to the database via a csv file transfer and go through a registration process. Once registered, they have agreed to accept voice traffic sent to them by any other user of the database.

The e164.org service handles media and signaling of the calls but does not provide conversion services. If there are interoperability issues they must be sorted out by the communicating parties. The transport of the call for its actual termination is typically via the public Internet unless both parties agree on an alternative method. Since this service is based on the Internet, then the Internet is the standard choice for transport.

Any network operator can utilize the e164.org ENUM service. This service is currently in use and accessible from any public IP connection. e164.org is a private company based in Australia.

The Voice Peering Fabric

The Voice Peering Fabric (The VPF) is a distributed Layer 2 Ethernet peering fabric for VoIP. It acts as a central point

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for VoIP networks to interconnect and establish VLANs between each other for the purpose of sending and receiving VoIP traffic. This helps to avoid the public Internet and enables private layer 2 Ethernet connections without the need for ordering disparate long haul circuits.

The VPF ENUM registry is a service of The VPF. It is multi-lateral, non CPE based and free to all members of The VPF. The fee to be a member of The VPF distributed Ethernet switch fabric is $1,500 per 100meg FE port per month. The VPF ENUM members register their numbers via the public Internet then so agree to receive voice traffic from the other members. The VPF ENUM registry handles media and signaling but does not provide conversion services. If there are interoperability issues, they must be sorted out by the communicating parties. The distributed Ethernet fabric handles the call transport between sender and receiver. Although The VPF does accept public IP VPN tunnels as an interconnection (and ultimately transport) method, over 95 percent of its members interconnect to the fabric via Cat5 using Layer 2 Ethernet in one of the VPF facilities located within major carrier hotels.

The VPF ENUM registry is a mirrored database that is recreated in each VPF site to keep local traffic from having to traverse the distributed links to a central database. This improves efficiency and keeps costs low.

The VPF currently has 53 members consisting of international wholesale minute providers, CLECs, VoBB service providers, universities and enterprises. The VPF ENUM Registry currently has over four million registered numbers and is carrying on average 10 million minutes per day — a run rate of two billion minutes per year. The fabric and service is accessible from any VPF location via a direct Ethernet connection, through any of its Ethernet transport carrier Alliance Partners, or via a public IP connection. The VPF and VPF ENUM Registry are services of Stealth Communications, a privately held company based in New York.

No matter which implementation fits your comfort level, one thing is clear — there are some very compelling reasons to use this technology and the sooner you do, the more you will save. For more information on any of these services, please visit www.verisign.com, www.e164.org, and www.thevpf.com.

Next month, we'll take a look at who is using these services, how they use it and why. IT

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

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By Tim Lorello

Overcoming The VoIP E9-1-1 Challenges

When last we met, we had discussed the very real possibility of VoIP (define - news - alert) adoption being slowed, delayed, or regulated because of the challenges related to providing Enhanced 9-1-1 (E9-1-1) (news - alert) service. We saw how one of the most revolutionary and advanced forms of telecommunications available on the planet often provides no 9-1-1 service; and even when it does so, it is providing access to emergency services in ways that turn the clock back over three decades. In particular, when a VoIP caller dials 9-1-1, the service routes to a 9-1-1 Telecommunicator who likely does not know the location of the caller. The caller's location information is automatically provided today with all wireline and most wireless services, but not with most VoIP-initiated calls. Will this shortcoming slow VoIP adoption? Will the FCC impose regulatory requirements that may impact the rollout of VoIP service? And would you, as an end user, replace your wireline or wireless service today with a new service that did not provide the most effective delivery of location information for this lifesaving tool? Thankfully, solutions are appearing that may make this shortcoming disappear.

E9-1-1 = Locate + Route + Deliver

To better understand the challenges, and ultimate solutions, surrounding E9-1-1 and VoIP, it helps to break down a 9-1-1 call into three components: Locate, Route, and Deliver. Simply put, in order to provide "Enhanced" 9-1-1 service, you must:

- Locate the caller.
- Route the call to the Public Safety Answering Point (PSAP) that handles emergency dispatch to that caller's location; and
- Automatically deliver the caller's location information to the 9-1-1 Telecommunicator at the PSAP.

It is this last step that differentiates basic 9-1-1 service from Enhanced 9-1-1 service. For VoIP, each of these components poses differing levels of challenge.

Locate A VoIP Caller

The ability to locate the 9-1-1 caller is perhaps the greatest current challenge. For a wireline caller, this locate component became a simple matter of assigning a location, the street address, to the caller's phone number. As long as you could automatically deliver the calling party number to the PSAP, this association could be used to dispatch emergency services to the correct home or office address. The introduction of wireless technology created new challenges since the phone number would no longer identify the caller's location. To address this challenge, the wireless industry introduced two main methods for locating the caller: network-based or handset-based triangulation. The former relies upon various, and improving, methods of measuring signal strength and signal timing at the cellular base station to derive the caller's location. This approach has been advertising 50-meter accuracy. The second method, handset-based triangulation, places a Global Position System (GPS) chip in the handset and uses satellites to accurately locate the caller's position. In order to save battery life and reduce the time it normally takes GPS to first locate a device, the industry introduced an Assisted GPS (AGPS) mechanism that would feed critical information to the handset as it attempted to get a satellite fix. Using AGPS, the location of the caller is advertised to be within three to 30 meters in many circumstances.

VoIP introduces challenges very similar to those originally faced by wireless: how do you locate a caller whose location is not tied to a single domicile? Automatic location techniques will likely follow the same path as did wireless: network-based and handset-based. Network-based port-mapping methods already exist for private data networks and can easily be extended to add a location coordinate to the information maintained. As a user "plugs into" a particular port or is picked up by a WiFi base station, the location of that port or base station can be associated with the caller. Simultaneously, new VoIP handsets are appearing on the scene from companies experienced in producing cellular handsets. In the near future, these handsets will likely have embedded GPS chips. In the meantime, VoIP service providers are introducing techniques that allow the user to easily identify his or her current location (and, in most instances, these will be a short list of frequently-visited calling locations).

VoIP introduces challenges very similar to those originally faced by wireless: how do you locate a caller whose location is not tied to a single domicile?

Route A VoIP Call

Once a caller's location is available, it is important to route the call to one of over 8,000 PSAPs in the United States. The jurisdictions of these PSAPs change with the changes of their communities, and the ability to route calls to the appropriate PSAP requires the use of a geographical database that maintains the coverage polygons for each PSAP. This situation is further complicated by restrictions on how to route calls to PSAPs: only qualified Local Exchange Carriers (LECs) are allowed access to the telephone connections serving many of these PSAPs; and many VoIP service providers, usually intentionally, are not considered to be a LEC or a Competitive Local Exchange Carrier (CLEC). Thus, though this very same requirement is already being met for the tens of thousands of 9-1-1 calls being placed from wireless phones each day, there are political challenges associated with granting VoIP service providers access to these route mechanisms. Collaboration is required and will likely overcome this obstacle.

Deliver Location Information Of A VoIP Call

Finally, there is the challenge to automatically deliver the location information to the selected PSAP. Again, this very same challenge has been addressed by the wireless industry. For the wireline industry, the entire network had to be upgraded to support the delivery of the calling party number to the PSAP. This method, which we take for granted in our homes and on our wireless phones, was unimportant to a wireline industry more focused on connecting two parties than on identifying the caller. It took three decades to enhance local and long-distance infrastructures in order to pass the calling party number to the receiving device or system. Unfortunately, for the wireless industry, the phone number alone is insufficient to determine a caller's location. Techniques were created to replace the calling party phone number with a "pseudo-number." This pseudo-number was nothing more than a 10-digit database key that could be used to deliver the original calling party number and the location of the caller. Over 42 percent of all PSAPs currently receive the X/Y (latitude/longitude) coordinates of wireless 9-1-1 callers. Using similar techniques, VoIP service providers can deliver the location information to these PSAPs.

VoIP Can Revolutionize Emergency Services

Our final look at VoIP E9-1-1 should be as provocative as the technology. How can we integrate the concepts and capabilities presented by the Internet telephony world and use them to revolutionize the way emergency services can be provided? This question will be addressed in our wrap-up view of VoIP E9-1-1 in a subsequent article. IT

Tim Lorello is senior vice president and chief marketing officer for TeleCommunication Systems (TCS). For more information, please visit the company online at http://www.telecomsys.com.

A Conversation With Philip Asmundson

Thoughts On VoIP, Michael Powell, And The Outlook For 2005

I recently had the chance to chat with Philip Asmundson, who was just promoted to national managing partner of Deloitte's U.S. Technology, Media and Telecommunications (TMT) industry practice in late January. Here's what he had to say about the current state of our industry.

GG: Please describe the role of Deloitte's Technology, Media & Telecommunications Group.

PA: The TMT Group is composed of service professionals who have a wealth of experience serving technology, media and telecommunications companies throughout the world in areas including cable, communications providers, computers and peripherals, entertainment, media and publishing, networking, semiconductors, software, wireless, and related industries. These specialists understand the challenges that these companies face throughout all stages of their business growth cycle and are committed to helping them succeed. Deloitte is a leader in providing strategic, financial, and operational assistance to its technology, media, and telecommunications clients.

GG: What areas in telecom should my readers keep an eye on in the coming year?

PA: In our predictions reports, we identify three key trends that your readers may want to watch for 2005:

1. Two billion cellular users by end of 2005.

By the end of 2005, there will be nearly 2 billion cellular mobile subscriptions worldwide and in excess of 100 percent market penetration in some markets. Subscriber growth will be strongest in developing countries (including Asia and Latin America). The most compelling and lucrative mobile content will continue to revolve around phone personalization, such as ring tones, real tones, wallpapers, and basic games.

2. Strength in PSTN, VoIP, and Broadband

In 2005, the vast majority of voice calls will still originate and terminate on the PSTN (Public Switched Telephony Network) due to superior call quality and overall reliability. PSTN operators will reduce prices. Key convenience features, such as stored number dialing, text messaging, and conference calling, will stimulate call volume over fixed lines. Meanwhile, VoIP call volume and the user base will increase significantly among consumers and businesses, but adoption and growth will be limited by shortfalls in VoIP (define - news - alert)'s quality, consistency, and reliability and the resulting slightly negative image in the marketplace. Broadband penetration will continue to grow in 2005, with broadband connections finally outnumbering dial-up in many countries.

3. RFID: Zero to 10 billion tags by end of 2005

In 2005, RFID will finally make it out of the lab and into the commercial world. The combined influence of major retail chains, defense contractors, automotive manufacturers and others — all of whom are requiring suppliers to use RFID — will prompt a massive increase in RFID adoption, starting from essentially zero. By the end of the year, billions of RFID tags will have been sold and used.

GG: VoIP is an industry on a tear. To what do you attribute the recent growth spurt of this technology?

PA: We believe VoIP's call volume and growth has been fueled by the proliferation of broadband and growing public awareness. VoIP's call volume and user base will increase significantly among consumers and businesses alike, but few enterprise customers will migrate completely. Although we believe that VoIP is

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inevitable, it will remain a niche product in 2005, but its importance to the global voice market will grow substantially.

GG: What are some of the hurdles facing the industry that would impede its success?

PA: Although VoIP quality will continue to improve, for many it will still fall short of expectations. New features will partially offset the quality issues, but those functionality advantages will often go unexploited as vendors and customers continue to focus on price. For enterprises, cost savings may well be less than anticipated, slowing wider adoption. Instead of taking a chance on VoIP, many enterprises will focus instead on getting more value from the PSTN, hiring a professional negotiator or enforcing governance to drive costs down. An even greater number of companies will opt for a hybrid approach, using VoIP for internal communications and the PSTN for external traffic. Another barrier to VoIP adoption will be the rise of VoIP spam, made possible by low-cost VoIP calls that are not subject to current telemarketing regulations. Although VoIP spam will be a limited trend in 2005, there will be a perception of this new threat to users. Finally, the regulatory framework has to date, taken a hands off approach to VoIP but as VoIP grows we expect regulatory scrutiny to focus on VoIP services.

GG: What are your thoughts on the news that FCC Chairman Michael Powell is stepping down from his post at the agency?

PA: Chairman Powell has led the FCC during a period of extreme turbulence for the industry. He has struggled to get his policies passed and has suffered from an inability to get a coalition among the four other commissioners. In addition, the FCC has been relegated to the backseat in some areas, as the courts have been drawn into not only interpreting policy but in some cases writing it. As a result, very little has been clarified during his administration and the new FCC chairman will face an increasing need to establish regulatory policy in an increasingly complicated environment. Look for the new chairman to come from outside the FCC and for further turnover at the commissioner level of the FCC. IT

Part II: What To Look For In A Wholesale VoIP Partner

By Shawn M. Lewis

Service providers of all kinds, including CLECs (Competitive Local Exchange Carriers), IXCs (Inter Exchange Carriers), ISPs (Internet Service Providers), FBWs (Fixed Broadband Wireless Operators), and cable operators now realize that they must offer a VoIP (define - news - alert) product to their customer base in order to maintain a viable business. The majority of these service providers lack the technical and operational expertise, not to mention the financial wherewithal to build and operate their own next-gen network facilities making it necessary for them to partner with a wholesale carrier.

So, we ask ourselves… What to look for in a wholesale VoIP partner?

Industry analysts agree the wholesale market has created a large business opportunity for new, next generation wholesale carriers. The market for wholesale broadband voice services in the U.S. has been characterized principally by quality, price competition, and least-cost advantage.

But is that enough? How do you know which wholesale partner is the right wholesale partner for you? Let's address what you should look for in a wholesale partner, including technology (customizable applications, service creation environment, scalability); automated provisioning (service provider interface, end user controls, real-time CDR); and fulfillment (rapid deployment, endpoint equipment, and customer support) — all critical to a service providers successful VoIP business.

Technology ownership. Are the applications fully integrated into the network?

One of the biggest value propositions about offering a VoIP product is the ability to integrate enhanced features and applications. These features include unified messaging, Centrex, IP-PBX (define - news - alert) and enhancements to information services. There are several wholesale VoIP carriers offering these enhanced features and applications, but most are offering them through third-party application server vendors. That said, these applications are not fully integrated in to their networks, meaning that the wholesale carriers do not have full control of the technologies. This is a major issue and limitation for the carrier since it is important for them to offer true scalability and seamless interoperability.

It is a technical and operational challenge for carriers to be able to develop, own, and manage the server applications end-to-end. And, as the market is teaching us, service providers need to take strong consideration in looking for a wholesale VoIP partner that owns their technology.

Customizable applications and a true service creation environment.

We believe that in order for a service provider to be successful, they need to be able to customize their applications and feature sets to meet the needs of the market they are attempting to penetrate and to stay ahead of their competition. Microsoft has been the most successful example of what ISVs (Independent Software Vendors) can develop with their creation environments such as .NET and the SDK (Software Developers Kit). Thousands of software vendors have created their own vertical software applications enabling them to be successful. I believe that service providers need to look for VoIP partners that can enable them to do the same thing with voice, data and application features. It is imperative that providers lead with features and services rather than just price.

Automated provisioning and end-user controls are not optional; they are a must.

In today's price-driven market, service providers must keep their operational costs down particularly in customer provisioning and moves, adds, and changes. These are critical areas where new customer acquisition costs can spiral out of control when live customer service agents have to interface with the customer directly. Service providers need to consider the following key questions at a minimum:

• Does your VoIP partner have a fully integrated set of network interface tools that enable you to rapidly provision new

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customers?

• Is there an end-user Web interface that enables users to manage their own account, calling features, billing, etc.?

• Will both interfaces (end-user and service provider to carrier) enable end-users to sign up and have new services provisioned in a fully automated environment, right from the service providers Web site?

• Are these interfaces customizable and able to be private labeled for the service provider?

• Are you able to get real-time CDRs (Call Detail Records) from your VoIP partner to ensure accurate and timely invoicing?

Fulfilling new customer orders requires endpoint devices and customer support.

There are a lot of service providers that have not had to provide endpoint devices in order to provision a customer. As this is a new operational task, service providers need to understand a lot of critical issues in fulfilling customer orders. Here are a few key considerations:

• Does your wholesale VoIP partner have endpoint device vendors that have had their devices tested for seamless interoperability with their network? This is a critical requirement to limit the first time setup issues with the customer.

• Make sure to keep up with developments in the endpoint device marketplace. Have a wide enough selection of vendors and equipment options so you can meet any customer's needs rapidly and efficiently.

• How are you going to physically fulfill the order including warehousing, shipping and device activation? This is a job better left to the experts. There are several fulfillment vendors that can be used that can limit the operational disruption of doing it yourself. In most cases, using an outsourced fulfillment vendor can provide a fixed cost rather than the unknown variable costs of doing it on your own. IT

Service providers need to take strong consideration in looking for a wholesale VoIP partner that owns their technology.

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.

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IP Telephony Improves Productivity And Lowers Costs For The Village Of Lombard

By Christopher Kearney

Known as the “Lilac Village” for the abundance of pink and purple shrubs planted by founder William Plum, the Village of Lombard, Illinois attracts horticultural fans from around the world each May for the annual Lilac Time celebration, a 70-year-old horticultural tradition.

Located 50 miles from Chicago, this scenic village and its 42,000 residents are served from a six-building municipal complex housing town government, police, public works, fire protection, and water works. The phone system used by the village was based on aging PBXs that were experiencing frequent failures. The PBXs didn’t support features that were deemed critical, such as automated voice mail, caller ID, and E-911 support. Two additional PBXs at remote fire stations were incompatible with those at the central village complex, hindering communications. In addition to the voice network, the data network, based on older switches from a variety of vendors, was no longer meeting the village’s needs. Users were experiencing slow application response time and the network was difficult to manage.

To alleviate these problems, and to obtain the communication features needed to manage a 250-person organization, village IT Manager Larry McGhinnis chose to migrate to a converged IP voice/data architecture based on Cisco (quote - news - alert) IP telephony technology.

The GTP Solution

Greenwich Technology Partners and Cisco Systems worked together to design and deploy a complete IP telephony voice solution as well as a new converged voice/data IP network.

The voice solution was based on Cisco Call Manager, Unity voice mail, and Cisco IP fixed and wireless phones, as well as Cisco’s Emergency Responder, an enhancement to Call Manager that enables emergency agencies to identify the location of 911 callers, even when phones or people move from one location to another. The new converged network, based on Cisco routers, switches, and 802.11g (define - news - alert) wireless access points, provides seamless connectivity for both fixed and mobile users throughout the village’s campus facilities.

Security was a major concern for the village. GTP deployed Cisco PIX firewalls to protect the village from Internet-based attacks. Cisco Security Agent, a host-based intrusion prevention solution, was installed on key servers to detect and prevent malicious behavior. New Virtual Private Network (VPN) capabilities were deployed to enable secure access by village staff working from home.

One of the village’s most unique requirements was the need for public works engineers to move freely between desks when reviewing site or construction plans anywhere in the complex. To accomplish this, GTP distributed Cisco wireless LAN phones to the engineers, who are now reachable via their ordinary desk phone numbers regardless of their location within the complex.

Net Result

Employees of the village now enjoy vastly improved voice features with their new IP-based phone system. Features such as auto-attendant and call transfer have enhanced user productivity. Staff can access user directories from the screens on their IP phones and can even check their voice mail via the Web. “Our new IP telephony system has greatly enhanced our ability to serve the community at a lower cost than our old PBX,” said IT Manager Larry McGhinnis. “We estimate that we’ll save upwards of $35,000 this year because of reduced operations costs and the consolidation of phone lines that this system has allowed us to accomplish.” IT

Christopher Kearney is Practice Director, Convergence at Greenwich Technology Partners, a network infrastructure consulting and engineering company. For more information, please visit them online at http://www.greenwichtech.com.

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IP PBX Product Roundup

FacetPhone

FacetCorp

http://www.facetcorp.com

FacetPhone is an IP-PBX telephone system for small to medium size businesses, designed to completely integrate the phone system with the user's desktop computer.

FacetPhone includes standard features such as voice mail, auto-attendant, conferencing, automatic call distribution, Caller ID support, and call detail recording. But it also includes advanced capabilities like: visual voice mail management and forwarding to e-mail; enterprise instant messaging; computer telephony integration with UNIX, Linux, or Windows applications; call recording; presence management; roaming extensions; and built-in branch office and telecommuter integration, and an incredibly powerful IVR system.

The FacetPhone architecture is based on an industry standard Linux or UNIX server with external media gateways making the phone system inherently flexible, scalable, reliable and cost effective.

Sphericall

Sphere Communications Inc.

http://www.spherecom.com

Sphere Communications' Sphericall IP PBX is designed to enable users to deploy enterprise telecommunications as software running on an existing data network with industry standard servers and no expensive proprietary hardware.

Sphere Communications enables enterprises to deliver the complete communications package — telephony, voice mail, unified messaging, on-demand conferencing, text messaging, presence, softphone, desktop video conferencing, advanced forwarding features, smart directories and more. Sphericall scales to over 30,000 ports across any number of locations and integrates seamlessly with legacy environments allowing companies to grow at their own pace.

The solution's open systems approach allows the choice of a wide variety of endpoints (IP phones, analog phones, wireless phones, etc.) to meet the diverse requirements of today's businesses.

Sphere offers an all inclusive user-based pricing model, combining all features into one price, eliminating startup costs and redundancy penalties. This dramatically simplifies purchasing, implementation, and management. Sphericall IP-PBX is certified secure and reliable by the Department of Defense reflecting the most stringent testing available today.

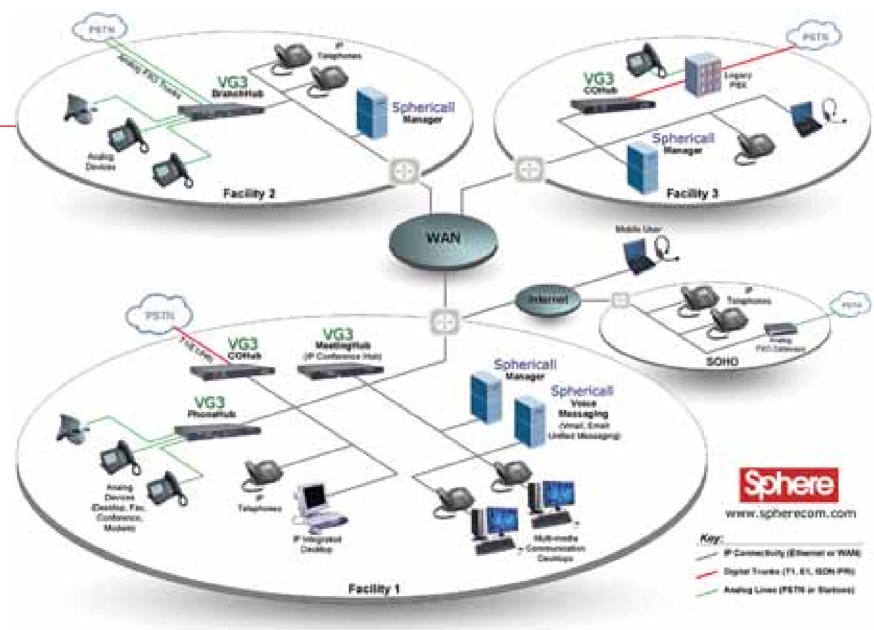
NBX V3000

3Com Corporation

http://www.3com.com

The 3Com NBX V3000 platform with Release 4.4 software features a 1U package with four analog ports, one station-side port for fax machine, and the requisite 10/100 MB Ethernet interface to give your small business call control and gateway platform.

The platform offers built-in call-processing features, including voice mail with 400 hours of storage, automated attendant, hunt/call groups, call detail reporting, computer telephony integration (CTI), PC-based visual voice



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mail/e-mail clients (IMAP4), and optional software that delivers unified messaging capabilities.

The NBX V3000 can be expanded to support as many as 1,500 devices. Additionally, a wide array of advanced functionality for network, remote offices, and T1/E1 access is available through the SuperStack 3 NBX expansion chassis. The platform is available in more than 61 countries with support for more than 11 languages or dialects.

OmniPCX Office

Alcatel

http://www.alcatel.com

The Alcatel OmniPCX Office was specifically designed for small and medium-sized businesses (SMBs). The product provides business telephony, advanced call center (country dependant), controlled Internet access, integrated internal and external communications through e-mail and unified messaging, and a strong IP infrastructure for sharing data resources and running IP telephony and VoIP.

The OmniPCX Office is a modular system that can be customized, so users need only buy and install needed functionality. The modular architecture is combined with a unified software system and can support new features, additional terminals, and increased capacity. Alcatel OmniPCX Office features flexible configurations and services. Three rack-size modules with universal slots provide for numerous board combinations.

The Alcatel OmniPCX Office is based on Linux and is based on open-standards protocols (including CSTA, TAPI, and IP), so that it supports a wide range of applications developed both by Alcatel and its partners.

Allworx 10x

Allworx

http://www.allworx.com

The Allworx 10x system supports both analog and VoIP phones. It includes a customizable auto attendant, call report log with details, call routing, and central office (CO) line support (nine standard, up to 33 optional).

The system also offers a dialing plan option which allows the Allworx 10x to look at the number being dialed and route the call through the lowest — cost option (e.g., long-distance carrier or VoIP service provider).

The Allworx 10x system also offers Direct Inward Dialing (DID), which allows extensions to be used as direct phone numbers, and can give a system the appearance of being a much larger one; fax support, PBX and Key system features, phone support for up to 40 users/100 extensions, presence management, site-to-site and remote user support, text-to-speech capability, unified messaging, voice mail with unlimited storage, and VoIP capability.

AltiWare 5.0

AltiGen Communications, Inc.

http://www.altigen.com

AltiGen manufactures IP call center solutions and IP-PBX telephone systems for small-to-midsize businesses. The company's offering includes AltiWare version 5.0, supporting AltiGen's AltiServ systems; the AltiContact Manager, AltiServ Contact Center and AltiServ IP Business tele-

phone systems. Version 5.0 offers significant enhancements for every aspect of the AltiGen system including call center, distributed communications, Voice over IP, enhanced general telephone systems functionalities, management, and reporting and improved integration capabilities.

AltiWare 5.0 offers enhancements including Modular Licensing with reduced up-front investments; the AltiServ Contact Center now included in all AltiWare systems; Advanced AltiContact Manager with advanced applications available on a single user basis such as Centralized Call Recording and dynamic call prioritization with skills-based routing; IP-PBX with improved IP networking tools and security including E911 support and remote IP extension survivability; Distributed Call Centers with advanced features like "Look Ahead" routing for overflow and remote supervisor and agent support; Multi-site Businesses with enhanced multi-site management tools and an improved multi-site operator console; Desktop PC software and administration interfaces redesign including new capabilities for all PC clients (AltiConsole, AltiAgent, AltiSupervisor, AltiView, ACM Administrator); Reporting with enhanced CDR search and reporting tools; and AltiGen SDK with additional APIs available for developers and integration requirements.

Avaya IP Office

Avaya

http://www.avaya.com

The Avaya IP Office communication solution brings together voice, data, and communications applications for small and mid-size businesses. Optimized for the two- to 360-station market, this converged communications system enables businesses to deploy an infrastructure, that gives full-featured traditional voice functionality today and the choice to deploy or migrate to IP when needed. In addition to the voice and IP applications integrated into IP office, this solution provides customer contact center capability from five to 75 agents.

Avaya also offers a Small Office Edition of the IP Office system, which supports from two to 28 users. The system includes built-in wireless LAN telephony capability, as well as security, conferencing, contact center, messaging and automatic call answering features. It is delivered with a Wizard installation disk for easy setup. The unit is only three inches high with a footprint comparable to an 8.5" x 14" legal pad.

CallManager Version 4.1

Cisco Systems, Inc.

http://www.cisco.com

Cisco CallManager is the software-based call-processing component of the Cisco IP telephony solution, part of Cisco AVVID (Architecture for Voice, Video and Integrated Data). The software extends enterprise telephony features and functions to packet telephony network devices such as IP phones, media processing devices, voice-over-IP (VoIP) gateways, and multimedia applications. Additional data, voice, and video services such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems interact with the IP telephony solution through Cisco CallManager's open telephony application programming interface (API). Cisco CallManager is installed on the Cisco Media Convergence Server (MCS).

Cisco CallManager Version 4.1 provides multiple servers clustered and managed as a single entity. Cisco CallManager clustering yields scalability of from 1 to 30,000 IP phones per cluster, load balancing, and call-processing service redundancy. By interlinking multiple clusters, system capacity can be increased up to one million users in a 100+ site system. Clustering aggregates the power of multiple, distributed Cisco CallManagers, enhancing the scalability and accessibility of the servers to phones, gateways, and applications. Triple call-processing server redundancy improves overall system

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availability.

CONVERSip

Comdial Corporation

http://www.comdial.com

The CONVERSip product portfolio, which converges voice and data networks into a single network, offers a fully-integrated solution designed for stand-alone and networked environments to promote connectivity for small to midsize offices in single or multi-site locations. Featured among its CONVERSip product lineup are the MP1000, which earned a 2004 Internet Telephony Innovation Award, and MP5000 Media Platforms. The MP1000 is an affordable IP telephony solution designed exclusively for smaller VoIP deployments and for small or branch offices and retail locations. With the MP5000, Comdial's converged communication solution, existing customers can leverage their legacy telephone systems while migrating to the latest IP-based technology.

Asterisk

Digium

http://www.digium.com

Asterisk is a complete PBX in software. It runs on Linux and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in three protocols, and can interoperate with almost all standards-based telephony equipment using less expensive hardware.

Asterisk provides voice mail services with directory, call conferencing, interactive voice response, and call queuing. It has support for three-way calling, caller ID services, ADSI, SIP and H.323 (as both client and gateway). Asterisk needs no additional hardware for voice over IP. For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsors, Digium. Digium has single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks as well as a single port FXO card and a one- to four-port modular

FXS and FXO card.

Nexspan

EADS Telecom

http://www.eadstelecom-na.com

EADS' Nexspan is an IP-native suite of network platform elements, productivity applications, and phones that offer IP telephony features right out of the box. The Nexspan family of IP-PBX products offers a converged telecommunications platform that provides a foundation for an IP network and flexibility to unite your current network elements into a seamless converged network.

EADS' basic IP-PBX for the Nexspan suite, a standalone Nexspan S, can handle 48 digital lines or 250 IP subscribers, but can be networked together with other Nexspan elements to create a much larger network. It can also be combined with the Nexspan Communications Server as part of a fuller IP environment.

The Nexspan L is designed for more distributed, larger customer environments, which require more redundancy and reliability. Each Nexspan L can handle 192 digital lines and 250 IP subscribers. And, again, the Nexspan L can be networked with other Nexspan elements to support as many subscribers as you need.

The heart of Nexpsan's pure VoIP is the Nexspan Communications Server. It works with the Nexspan S and L in converged environments or standalone as our full VoIP solution, with each server handling 960 IP subscribers.

Quadro16x

Epygi Technologies, Ltd.

http://www.epygi.com

You can connect the Quadro16x to a broadband Internet service or to your company's gateway router using an Ethernet cable. Next, connect up to 16 ordinary phones and fax machines to the Quadro. In minutes, you are able to make VoIP calls for free.

In addition to its Voice over IP capabilities, the Quadro16x has a host of features for the small business. With its built-in IP router, you can create a firewall protected computer network. In addition, you can establish a highly secure VPN from the Quadro to other Quadros or to the corporate office.

The PBX features of the Quadro like voice mail and auto attendant give you all the phone features you would have at any large corporate office. The SIP voice over IP gateway with STUN allows you to connect to other Quadro users or a SIP server to make virtually free long distance calls all over the world. All this functionality is in one small appliance. The Quadro's FXO port will automatically engage one of the phone ports to provide lifeline support in the event of a power outage. Epygi offers Quadro in four versions including Quadro2x and Quadro2xPro for up to two phones; Quadro4x for up to four phones; Quadro16x for up to sixteen phones.

MX-ONE

Ericsson Enterprise

http://www.ericsson.com/enterprise/products/mxone/index.html

Ericsson launched the primary building block of the MX-ONE platform: the MX-ONE Telephony System. An IP PBX telephony engine, it consists of a telephony server and media gateway — and offers all the business-class features enterprise users are accustomed to, fitted for a standard 19-inch rack. This first release is supported by the existing MD110 management system and end-user applications.

Integrated into the existing IT infrastructure, the MX-ONE Telephony System lets enterprises enjoy all the advantages of server-based IP communication, together with the existing MD110 network, applications, phones and management system. A future-proof investment, it also offers seamless integration between sites using VoIP. For enterprises, Ericsson believes that it translates this into cost-efficient communication, from lower administration costs to billing control and affordable functionality for mobile workers.

The rest of the MX-ONE building blocks in MX-ONE will be released this year and consists of applications like unified messaging, presence management applications and speech tools. All managed by MX-ONE Manager, a new management platform based on active directory.

IP PBX32

I.C.S. Corporation

http://www.icstel.com

The IP PBX32 features a robust flash boot Linux system with application programs. The system also offers full digital IPBX functions with auto attendant, voice mail, schedule call, SMS, phone book, call log checking, status monitoring, and record-on-demand; eight-port VoIP capability, with no A/D loss high-quality digital switching. Its smart system monitoring function ensures stability and reports failures. It's QoS (Quality of Service) bandwidth control capability ensures VoIP quality.

The IP PBX32 has a compact size of 340mm X 260mm X 44mm (1U size); a 90-240 VAC input; and a built-in

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80V/24V switching power for ring/phone use. The system's PBX free port design features eight slots for CO/Ext modules with support of up to 32 ports. It also features a backlit LCD panel for easy system configuration; LEDs for power/switch status; and a built-in 20GB hard drive. The system is equipped with two cooling fans with speed control and one temperature sensor.

Axxess

Inter-Tel, Inc.
http://www.inter-tel.com

Axxess is based on open architecture interfaces and standard protocols, which offer the flexibility to tailor the platform to suit your needs. Support for VoIP protocols, such as SIP, provides a communications pathway — connecting diverse tools together so that they can speak to each other. SIP enables simple, flexible connectivity, which allow infrastructures, applications and endpoints to interact in a standard manner. Axxess also supports IEEE standards, such as 802.11b and 802.3af — enabling your business to provide tools that facilitate the mobility of employees. Inter-Tel offers support for these standards and many other industry-standard interfaces, which will help address your business requirements.

SIPS

IPeya
http://www.ipeya.com

The SIPS software is based on the Linux operating system. The system software resides in flash memory to ensure system integrity. User data files and voice mails reside on the hard drive with hard drive capacity ranging from 40 to 300 GB depending on system configuration. IPeya developed a proprietary encrypted file system to guarantee physical security of the system. In the case of a theft, the system's hard drive can't be read.

SIPS provides a safer migration path from PSTN to VoIP. IPeya's VoIP implementation is 100 percent compatible with Session Initiation Protocol (RFC 3261), therefore allowing users to utilize third-party SIP phones. In addition you can connect regular analog phones via inexpensive FXS adapters.

Web-based graphical user interfaces (GUI) gives users manageability and control. In addition to giving you ability to configure all system functions

remotely it has a unique discovery engine that allows you to find, provision, and troubleshoot phones, and other devices on your LAN from anywhere in the world. IPeya has three versions of the SIPS system — the SIPS S, for offices of up to ten employees, the SIPS M, capable of supporting up to eight VoIP lines, and the SIPS L with optional WiFi access.

Enterprise-CS

Iwatsu Voice Networks

http://www.iwatsu.com

The Enterprise-CS is a converged communication gateway that merges TDM and IP resources onto existing LAN/WAN infrastructures to transmit both voice and data traffic for higher cost-savings, fewer hardware requirements and efficient bandwidth usage. With its dual architecture, Enterprise-CS is a cost-effective alternative to native VoIP systems, while providing all the same power and advantages of VoIP at a lower the cost.

In a single gateway cabinet, Enterprise-CS can support up to 1,024 IP ports and integrates with applications including VoIP networking, unified communications, voice-activated call routing, and presence solutions. The system's platform supports 100 percent TDM, 100 percent IP or a combination of both TDM and IP capabilities and offers an integration with any ADIX system letting ADIX users get an IP upgrade without replacing their current system.

The system's flash-based software allows updates from a remote maintenance console — eliminating the need to modify or replace hardware supporting new software revisions. The Enterprise-CS also supports peer-to-peer communication enabling IP phones to "talk" to each other directly and rely less on system resources.

Mitel SX-200 PBX

Mitel Networks

http://www.mitel.com

The Mitel SX-200 PBX provides voice communications for small-to-medium sized businesses. Whether for a central location or across multiple sites, the SX-200 PBX offers scalability, centralized management, and networking capabilities. Additionally, the SX-200 has an attractive migration path to IP via the SX-200 IP Node that is designed to protect a business' existing infrastructure investment while providing the infrastructure and user benefits of an IP environment.

The Mitel SX-200 IP Node provides Internet protocol (IP) capability via the Mitel SX-200 PBX system, allowing businesses to benefit from a wide array of new infrastructure and user benefits such as IP networking, teleworking and IP desktop devices and applications.

The Mitel SX-200 provides centralized voice mail; support for digital, analog, and IP phones; 500+ telephony features; touch-tone auto and speech-enabled attendant; optional integrated unified messaging; support for automatic call distribution; advanced networking capabilities, including: T1, ISDN and E&M; feature transparency across multiple sites; support for ANI and DNIS to allow individual users to identify important Incoming calls; and support for IP Applications.

NEAX 2000 IPS/2400 IPX

NEC Unified Solutions, Inc.

http://www.necunified.com

The NEAX 2000 IPS is an IP-based communications system that provides users with a choice of time division switching (TDM), pure peer-to-peer IP connectivity across corporate local- and wide-area networks (LAN/WAN), or a combination of both, all in one system.

Combined with the choice of TDM and/or IP, the NEAX 2000 IPS modular design provides customers with the ability to customize their NEAX 2000 IPS to match current needs as well as add capacity and capabilities to meet future business demands.

The NEAX 2400 Internet Protocol eXchange (IPX) features the capability of pure IP peer-to-peer switching in addition to support for a hybrid network with traditional digital/analog switching and IP/TDM switching. This design provides users with the flexibility to continue to utilize their existing equipment while they phase in IP telephony and lay the foundation for future networks.

The system is designed to grow incrementally over its entire size spectrum, ranging from 384 ports to over 24,576 ports. The NEAX 2400 IPX also provides over 780 service features and applications that enhance productivity, reduce operating costs, and improve communications.

V-Box

NetSapiens

http://www.netsapiens.com

NetSapiens V-Box is a broadband phone system easy to deploy with Plug-n-Play installation. The system offers features such as phone authentication, which allows only authorized phones to register to your V-Box; call authentication enabling only authorized phones to make calls; individual extensions assigning unique extension to each phone; single user-multiple phones assigning multiple phones to a single user account and single voice mail inbox; hunt group automatically rolling over incoming and outgoing calls to the next available trunk line; alternate routing for outgoing calls to preferred carriers based on dialed-number prefix; and backward compatibility integrating with non-VoIP phone systems.

The system also supports firewalls and NAT traversal. The V-Box offers portability, a Web-based user interface for users and administrators; and wireless support of WiFi 802.11 phones. The system features caller ID, call waiting, call transfer, call history, call hold, remote office connectivity, auto attendant, customizable greetings, Interactive Voice Response (IVR), Follow-Me call forwarding, Do Not Disturb, voice mail, individual voice mail inbox for each user, voice mail retrieval by phone, Web, SMS, and e-mail, audio conferencing, conference by dialing into assigned conference room extension, conference by Web-initiated invitation, multiple conference rooms, and last conference participant list saved.

Business Communications Manager (BCM)

Nortel

http://www.nortel.com

The Business Communications Manager (BCM) is an integrated voice and data communications solution for small businesses with VoIP gateway functions and data routing features. The system has three different versions — the BCM 200 for five–60 users, the BCM 400 for 40–200 users, and the Communication Server 1000S for 150–1,000 users.

The BCM 200 supports e-business applications that extend network services to remote workers, increase portability, simplify moves and changes, and eliminate toll charges on site-to-site calls. Its reduced-size platform with two media bays is designed to meet the need of smaller sites.

The BCM 400's higher capacity platform features four media bays, enabling larger configurations to be supported at a lower cost.

The Communication Server 1000S is a full-featured IP telephony solution for the medium to large enterprise environment for up to 1000 IP clients per call server

SIPxchange IP PBX

Pingtel

http://www.pingtel.com

Pingtel's Enterprise SIPxchange PBX is an enterprise-grade IP PBX available in open source, based on SIP (SIPxchange PBX works with a variety of Media Gateways, Phones, and Servers.)

SIPxchange PBX is available as a 12-month subscription with certified software, complete product documentation, installation, and management tools, interoperability certification, operating system, and performance metrics. Maintenance and support covered by a 12-month subscription to the SIPxchange PBX Premium Edition includes updates, unlimited electronic self-help, and 7x24 TAC (Technical Assistance Center) support.

SIPxchange PBX provides robust functionality with onboard voice mail and auto attendant for up to 250 seats on a single server. SIPxchange server components (Comm Server, Media Server, and Config Server) operate on a single PC server. Any server selected must be capable of running Red Hat Linux version 7.3. Customers have the ability to choose from a wide array of hardware vendors and solutions; these include, but are not limited to, HP, IBM, Sun, and Dell.

ShoreTel5 Release2

ShoreTel

http://www.shoretel.com

ShoreTel5 Release2 includes foreign-language editions supporting the international offices of global enterprises and expanded legacy integration for easier VoIP migration. The ShoreTel5 Release2 has been enhanced at every level, from architecture and management to the hardware and software that provide an unparalleled user interface.

A new automatic message forwarding feature increases productivity and responsiveness to customers worldwide by providing automatic-forwarding of voice mail useful in a variety of scenarios.

Other new features aimed at international operations are caller-ID and message-waiting support for analog phones, CE marks on ShoreGear hardware indicate conformity with a common set of European regulations, as well as localized tones and cadences for ShoreTel's line of IP phones. The ability of the ShoreTel voice mail application to interface to a legacy PBX rounds out ShoreTel's migration options. ShoreTel already supports site-by-site migration with PBX networking using tie lines and voice mail networking using Audio Messaging Interchange Specification (AMIS). The ShorePhone IP210 is available for immediate shipment at a suggested list price of $249.

HiPath

Siemens Enterprise

http://www.siemens.com

The HiPath was developed in three versions — the 3000, the 4000, the 5000, and the 8000. The system offers you flexible configuration for your network; gateways that serve as interfaces to public telephone network; a common infrastructure; a single system for all locations; a single database; and a single application package.

The HiPath 3000 is suitable for up to 380 users, enabling IP telephony and traditional telephony. It is a successor product to the Hicom 150 system. The HiPath 4000 is suitable for up to 100,000 users, enabling IP telephony and traditional telephony. It is a modular license model, and a successor Hicom 300 system. The 5000 version is suitable for up to 1000 users offering IP telephony, using HiPath 3000 as gateway, and is an all-software solution. The 8000 version offers a carrier-class scalability up to 100,000 users per node and virtually unlimited users per network (depending upon configuration), and is an SIP-based overlay network. This version also offers a standards-based server and OS; a no single point of failure with hardware and software resilience; open industry standards; enterprise class telephony features; as well as VPN support for mobile users and small to medium sites.

Strata CIX

Toshiba Digital Solutions Division

http://www.toshiba.com/taistsd/index.htm

The Strata CIX is a business communications system with FeatureFlex, which allows users to customize or create features enabling greater flexibility and improved efficiency. You can create features companywide, by department, or for individual users.

Designed for small- to medium-sized enterprises or larger corporate users with multiple sites, Toshiba's Strata CIX supports up to 672 ports. It can run as a pure IP system or can be TDM-enabled, allowing enterprises to use IP where it makes sense. Strata CIX also offers all the features of Toshiba's traditional TDM systems in a native IP solution and provides a smooth migration path from Toshiba Strata CTX and Strata DK digital business communication systems.

What begins with eight digital telephone ports, one analog telephone port, and three CO lines with Caller ID can easily double in capacity to include 16 digital telephone ports, six CO lines, and two analog station ports. So you can add employee stations, telephone numbers, fax lines, and voicemail with near plug-and-play ease.

The Strata CTX28 even protects your investment should your needs eventually require a larger Toshiba telephone system. Simply upgrade and take your telephones with you. Virtually all telephones integrate easily with your new equipment.

TeleVantage Enterprise Manager

Artisoft, Inc.

http://www.artisoft.com

The TeleVantage Enterprise Manager enables businesses to connect distrib-

uted, multi-site locations into a single, unified telephone system. The system extends the TeleVantage ViewPoint — the system's desktop graphical interface, to show the phone and personal status of any remote user. With a single view of the entire organization's availability, employees can also point and click to camp on any busy remote extension so they are automatically called back and connected when the extension becomes free. The TeleVantage Enterprise Manager automatically distributes voice messages among TeleVantage servers as needed, and users have access to multi-site distribution lists.

The system automatically configures interoffice call routing over VoIP to eliminate long-distance toll charges between distributed locations providing centralized IVR applications that can be accessed by any office. TeleVantage Enterprise Manager automatically configures TeleVantage servers so they can communicate over VoIP and automatically propagates dial plan changes to every server in the TeleVantage domain.

Abolishing the need to manually enter another extension or user name by optionally importing and synchronizing users from Microsoft Active Directory, the system's Active Directory Synchronization Service automatically imports users into TeleVantage and updates user properties as they change in the Active Directory.

TeleniumIP

Vodavi Communication Systems, Inc.
http://www.vodavi.com

TeleniumIP is the IP-based telephony system in Vodavi's portfolio of communications products. The home office can easily be integrated into the main system with the extended phone applications. Using a standard broadband IP connection, a single IP phone can be a standard extension on the system using the same phone lines and voice mail of the main office.

TeleniumIP is offered in two models with different capacities. Its modular design provides for easy upgrades as your business grows. The systems can also be networked together to achieve greater system capacity and extend the system beyond a single LAN. The system offers networking features, Four Digit Dialing, Busy Lamp Field (BLF), Centralized Voice Mail, Message Wait Indication, Call Transfer, Call Forward, Centralized Attendant, Conference Name Display, and Absent Text Messaging.

Enterprise Interaction Center (EIC)

Vonexus
http://www.vonexus.com

Enterprise Interaction Center (EIC) offers IP telephony functionality based on Windows Server and Exchange Server 2003, including integrated auto-attendant, and call routing and queuing. Out-of-the-box screen-pop and IVR integration will be available for Microsoft Business Solutions products, initially for Great Plains and MS CRM, with support for Navision and Axapta in a future release. An Outlook telephony client will also be released, enabling customers to upgrade Exchange and Outlook from a messaging platform to a total communications solution. Vonexus is also developing new Windows CE and SharePoint interfaces to extend EIC's features to mobile employees.

EIC, which was first made available by Interactive Intelligence in 1997 on a TDM platform, was re-launched by Vonexus in 2004 as an integrated IP telephony solution to address the growing demand in the SMB market for converged voice and data solutions running on the Microsoft platform.

MX250

Zultys Technologies
http://www.zultys.com

The MX250 comes standard with two analog FXS circuits and has three slots to accommodate telephony interfaces. You can use modules to connect to analog (FXS and FXO), ISDN BRA (S/T), T1, and E1. You can configure the digital interfaces to carry voice or data traffic, or mixed voice and data.

The MX250 supports T1 and E1 ISDN PRA and T1 CAS. The CAS protocols are loop start and ground start with caller ID, and E&M wink with DID. The ISDN protocols supported over T1 are Lucent custom, Nortel custom, U.S. National, and Japanese ISDN. The ISDN protocol supported over E1 is ETSI with subaddressing. The ISDN protocols supported over BRA are Japanese ISDN and ETSI, with or without SPIDs.

The MX250 includes fax termination on any telephony interface. The MX250 provides music on hold through an external connection, from the Internet, or from its internal hard disc. It supports overhead paging using an FXS circuit or a 3.5 mm audio output. IT

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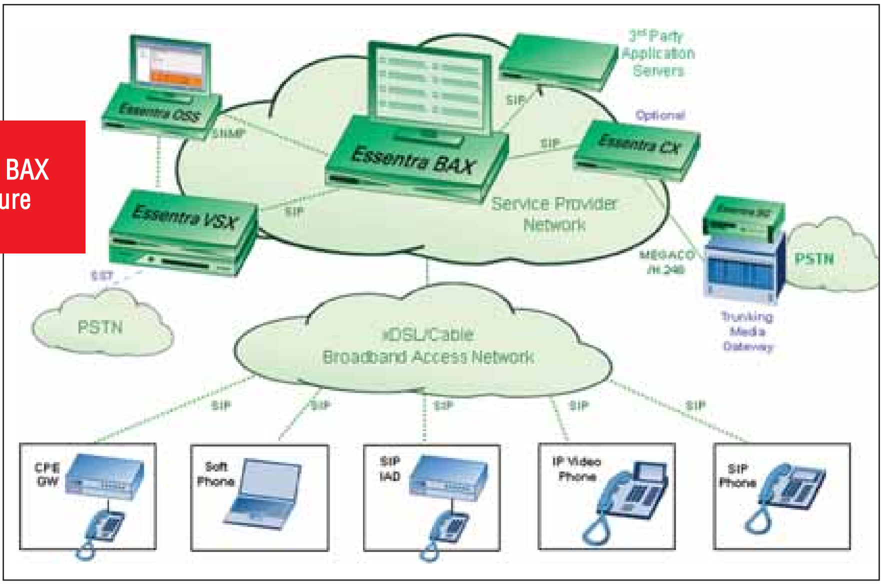
INTERNET TELEPHONY EDITORS' CHOICE AWARD 2005

Essentra BAX

VocalTec
Two Executive Dr.
Fort Lee, NJ, 07024
Ph: 201-228-7000
Fx: 201-363-8986
Web: http://www.vocaltec.com

Pricing: Call for pricing.

Essentra BAX architecture



The broadband VoIP (define - news - alert) market is exploding — service providers from Vonage (news - alert) to Lingo (news - alert) to Packet8 (news - alert) to AT&T CallVantage (quote - news - alert) and even cable companies such as Comcast (quote - news - alert), Cablevision (quote - news - alert), and Charter (news - alert) are jumping on the broadband VoIP bandwagon. Many different types of service providers — including incumbent PTTs, alternative carriers, cable operators, ISPs, and ASPs — are looking to leverage the broadband access market to offer voice over IP.

There is much more to becoming a broadband VoIP service provider than just sticking a few VoIP gateways on your network. To become a VoIP broadband service provider will require provisioning, maintenance, and upgrading of end-user devices, billing, authentication, and a self-service Web page to enable customers to configure their voicemail and calling features.

VocalTec hopes to ease the complexity of integrating these various functions using VocalTec's Essentra BAX — Broadband VoIP Access Platform which utilizes and leverages the SIP (define - news - alert) protocol. The Essentra BAX softswitch platform is interoperable with many SIP endpoints (IADs, SIP phones, etc.) and can communicate with SIP-based trunking gateways for interconnection with the PSTN. An integral part of the Essentra Product suite, it is fully interoperable with the other Essentra components, thus enabling service providers to expand their access network into a full-fledged packet tandem system. The system runs on Linux and VocalTec claims carrier grade 99.999 percent availability

The Essentra BAX Broadband VoIP Access Platform offers service providers a comprehensive package for establishing residential and SOHO broadband services, including support for traditional subscriber calling features such as call forwarding, three-way calling, and dial missed number.

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

Other supported calling features include call waiting, call forward, caller ID sending/blocking, three-way calling, voice mail, return call, repeat dialing, and more. Other types of call features and services, such as prepaid calling cards and conferencing, can be supported through integration with application servers.

A Web-based subscriber self-provisioning interface (Figure 1) enables subscribers to configure which services they need and how they work, such as which phone number to forward calls to. In fact, the Web interface is designed to leverage the same Web interface for both system administrator and the subscriber with the subscriber only able to view and customize a subset of the administrator's capabilities. System administrators are able to set up services, add/delete/modify subscribers and monitor the system. As previously mentioned, subscribers are able to view and modify their own service settings, activate/deactivate services, view call logs, etc. The GUI interfaces are fully customizable and support multiple

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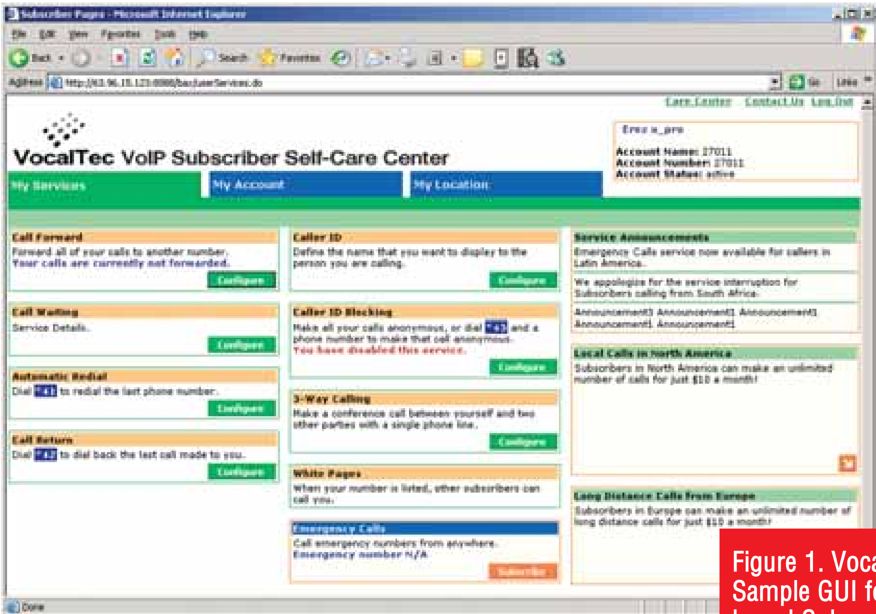

Figure 1. VocalTec: Sample GUI for Web-based Subscriber Self-Provisioning.

languages. This platform supports emergency dialing by allowing you to map an emergency access number (such as 911) to a real E164 number.

Essentra BAX scales up to 20,000 subscribers per server and is virtually a "turnkey" solution that allows service providers to simply connect a single BAX server in the network facility, while end users simply connect through a SIP-based IAD or SIP phone which hooks into their cable or DSL modem.

In fact, we tested the Essentra BAX with three different SIP-based end-user devices, including a Cisco ATA-186, a Grandstream BT101, and a Sipura SPA-100. We had these devices hooked up to our internal LAN with mapped IP addresses to the outside to get around any NAT issues. These devices registered with the Essentra BAX server with a specific SIP address. We then made several test calls to each of these devices and all of the devices rang when dialed. The voice quality was pretty good — we had no issues at all with latency or voice quality. All of our dialing tests passed with flying colors.

We should point out that while we mapped the devices outside the firewall with external IP addresses, the Essentra BAX supports STUN for VoIP NAT traversal and has tested/verified STUN support on the Grandstream and Sipura devices, as well as the X-Pro softclient.

The platform supports a fully functional media server for voice prompts as well as voice mail recording capabilities. Other features include load balancing, clustering, and the ability to integrate with RADIUS billing solutions. In fact, Essentra BAX supports postpaid and prepaid billing models. Prepaid services can be supported through the addition of a third-party real-time billing system. The prepaid solution includes IVR, balance notifications and termination of calls in real time when the balance is exceeded.

Essentra BAX supports industry standards when it comes to security. For example, secure communications between Essentra BAX and the SIP end-points is possible using the standard SIP DIGEST method security. Communication with the management interface for both system administrators and subscribers is based on secure HTTPS. One final feature of note is that the Essentra BAX can support H.323 when used in conjunction with VocalTec's Essentra EX Peering Manager. Essentra BAX can be deployed with other Essentra softswitch products to create the basis for future Class 5 alternative solutions.

Conclusion

All in all TMC Labs was very pleased with the Essentra BAX; from its scalability and reliability to its strong SIP support and other features, there was nothing we could point to in this product and say it could be improved. IT

PROS and	CONS
Good voice quality	None at this time
Scales well	
Strong SIP support	

INTERNET TELEPHONY® EDITORS' CHOICE AWARD 2005

Shunra Virtual Enterprise

Shunra
555 Eighth Ave., Suite 1102
New York, NY 10018
Tel: 877-474-8672;
Fx: 212-279-9561
Web: http://www.shunra.com

Price: Shunra Virtual Enterprise pricing starts at $40,000.

We already know that networks are critical in most enterprises, but what you may not know is that as networks become much more complex and distributed, the tools to monitor and test these complex networks also have to become more complex. We should also point out that applications often fail when deployed over a production network because IT staff ignore or are incapable of measuring the impact of the production network on the end-user quality of experience, before deployment. Many traditional testing tools do not accurately replicate geographically dispersed applications and remote end-users. Factors such as geographic distances and latency, the bandwidth of remote offices, and the number and behavior of remote end-users, must be incorporated into the development process for the application to perform optimally in production.

Fortunately, Shunra's Virtual Enterprise removes all of this complexity by providing an easy to use WYSIWYG design interface using Visio (See Figure 2) that provides a way to test your distributed applications in the real world environment, throughout the development lifecycle. In fact, one of the biggest selling points is that the diagrams within Shunra are designed via the popular Visio diagramming application and tested within Visio. You simply define your "clouds," which are your networks with various parameters and then connect the dots. Simply by drawing your network within Visio (drag-and-drop icons and connect them) and then assigning various parameters to the network entities (See Figure 3), it provides an exact replica of the production environment — including remote offices, end-users, and the network parameters, such as latency, upstream and downstream bandwidth, and so on. Shunra Virtual Enterprise helps you understand how your distributed applications will function, perform, and scale for remote end-users before deployment, and ensure on-time, on-budget rollout while meeting end-user expectations.

Any IT department looking to achieve consistently high service levels across the enterprise must be able to accurately model the distributed production application and network infrastructure environment. According to Shunra, "Accurate modeling is the key to effective testing, capacity planning, diagnostics, and service-level assurance. If you can't accurately model the production environment, your ability to analyze current application behaviors and predict future application behaviors will remain limited at best — especially when it comes to understanding the quality of the end-user's experience at all of your geographically dispersed locations."

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

2+2 = 5?

Shunra told us that unfortunately many IT organizations rely on tools that use mathematical calculations alone to model the production environment. These tools require engineers to input a complex set of parameters into the software, which then uses those parameters to build a theoretical model of the environment. Unfortunately mathematical models are prone to errors for several reasons. For one, it assumes that the behavior of IT environments can be accurately predicted with parameters; however networks are unpredictable and therefore mathemati-

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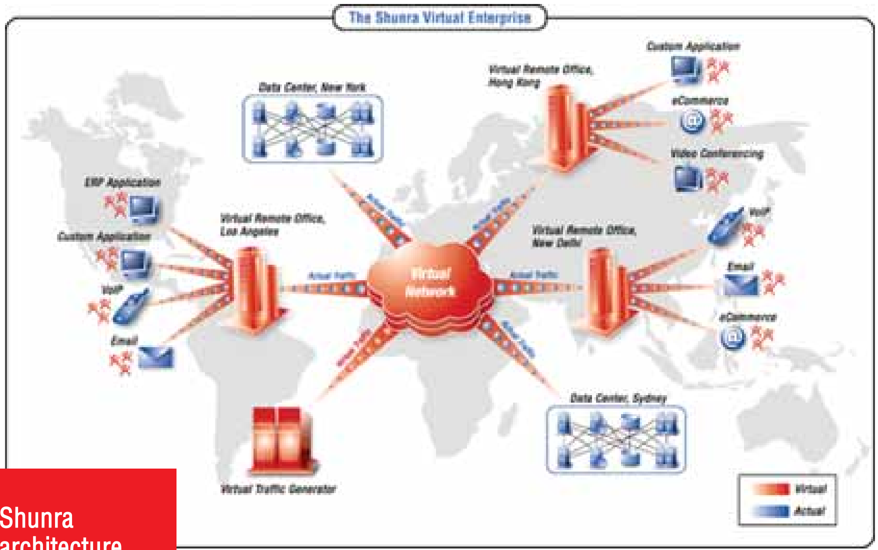

Shunra architecture

cal "assumptions" can never fully accurately define the network's behavior. Secondly, using mathematical models relies on the skill, precision, and completeness with which an engineer determines the appropriate values for the parameters provided, which also introduces potential sources of error. Lastly, the network is constantly changing, so the parameters would have to be constantly updated to keep in line with the network's behavior.

Shunra's solution to the mathematical modeling problem is to take an empirical approach rather than a purely mathematical approach to enterprise modeling, which is why they named their product "Virtual Enterprise." Shunra's empirical approach creates a "virtual enterprise" that reflects conditions in current and projected production environments with superb accuracy. Through this approach, users run the actual applications and send real traffic over the real infrastructure, through Shunra's emulation technology, which is configured by importing the network conditions directly from the production environment.

Testing — Have It Your Way…

Shunra gives you two ways of creating test scenarios. One way is to manually enter parameters including latency, packet loss, link faults, congestions, etc., (Figure 4).

The second way is capturing actual data, recording it, and then importing that data into the Virtual Enterprise. The Shunra Virtual Enterprise records up to 30 days of network conditions on a 24x7 basis, and uses that data to define parameters such as latency, error rates, and bandwidth utilization. For example if a packet is lost, the recording mechanism will capture that and play it back at exactly that point in time. This approach eliminates guesswork and ensures the accuracy of the model. It also allows "what if…?" scenarios to be created by making incremental adjustments to an accurate baseline model — rather than by speculating about possible future changes in environmental behavioral parameters.

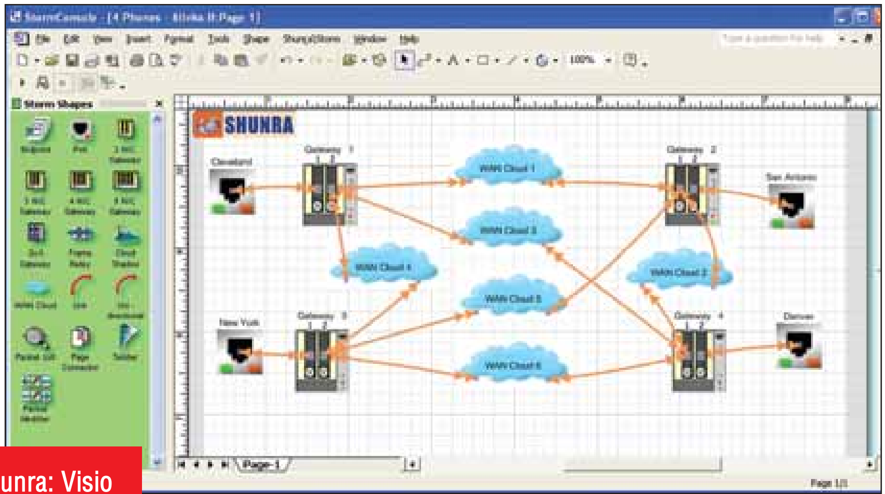

Figure 2. Shunra: Visio interface with Shunra integrated.

This platform has some obvious benefits to VoIP vendors or implementers of VoIP that include:

• Select the right VoIP infrastructure for their networks; Compare

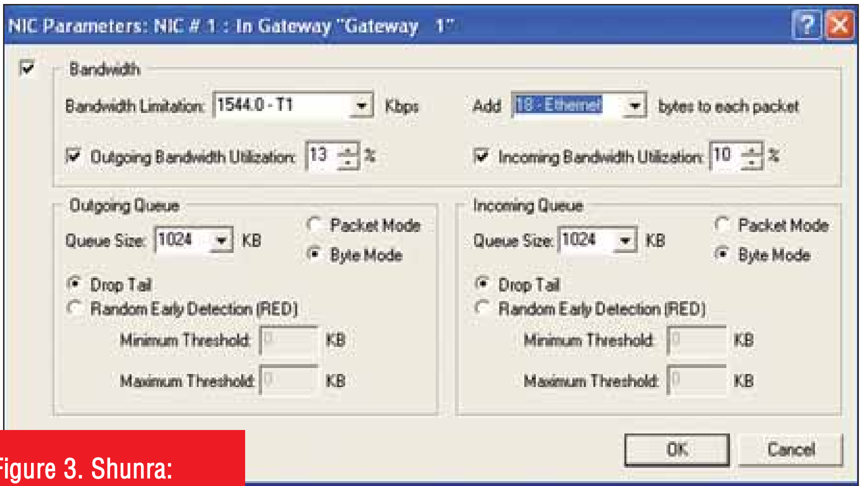

Figure 3. Shunra: Setting the bandwidth parameters for a gateway.

VoIP vendors over the exact same network conditions.

• Know that the network can support a VoIP system — before start the project, not three-quarters of the way into it.

• Know how the VoIP system will perform in production; Hear the End-User experience over any bandwidth.

• See how VoIP may affect other applications that share the same network, prior to rollout.

• Test the VoIP solution under worst case/extreme conditions.

• Ensure that changes to the network infrastructure do not adversely affect VoIP quality.

• Make sure that adding new applications, updating existing applications or adding remote users does not adversely affect VoIP quality.

• Troubleshoot and pinpoint problems in a repeatable environment.

• Eliminate the need for testing on the production network.

Features

One of the interesting parameters within Shunra's Virtual Enterprise is that you can check an option that allows or disallows packet reordering, i.e., allow packets to reorder (due to jitter). For example, if packet A is set to randomly delay 70ms and Packet B is set to randomly delay for 30ms, this will result in Packet B arriving before Packet A. If you disallow packet reordering, you will force Packet B to wait until Packet A has been received. However, for testing VoIP equipment it is vitally important to allow packet reordering. Many times there are bugs in VoIP equipment that are sensitive to packet reordering. One other interesting feature of note

PROS	and	CONS
Easy to use design interface Accurate production-environment modeling Choice in creating test scenarios		None at this time

is that you can change the IP header or the payload itself.

Other features include:

- Support for IPv6, Multicast.
- Real-Time Changes. Allows real-time changes to latency and data packet loss parameters, without having to stop the test.
- Extended Support for third-party applications.

This feature allows users to control any third-party application that supports automation, such as remote PCs running batch scripts, wireless emulation tools, ftp clients, home-grown testing tools, and Ixia's Chariot.

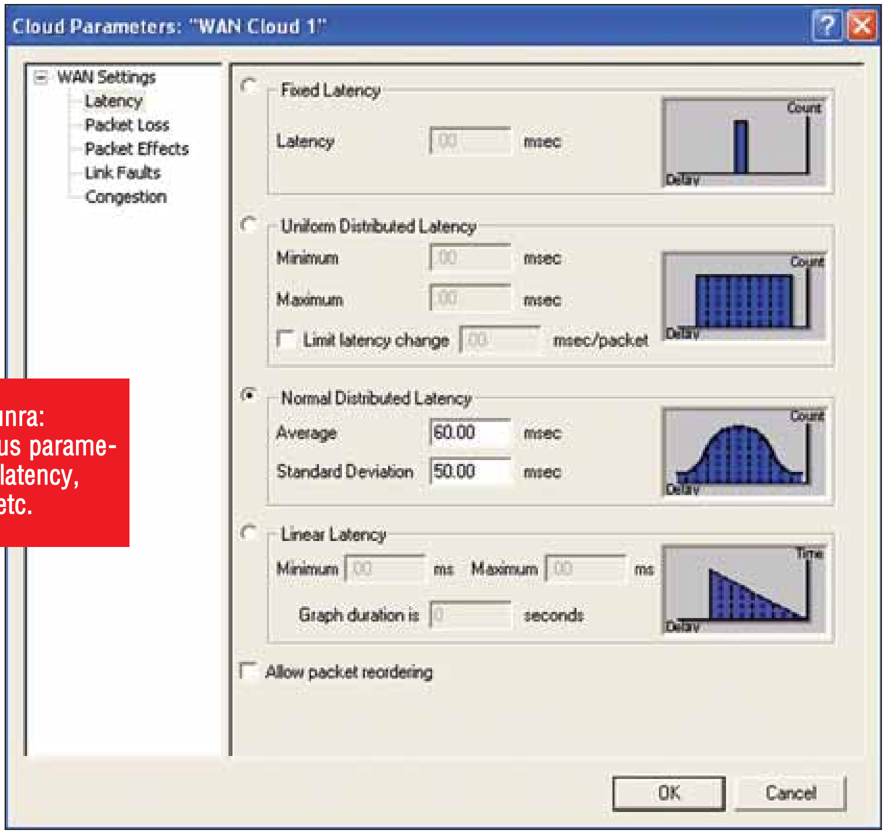

Figure 4. Shunra: Setting various parameters such as latency, packet loss, etc.

Conclusion

Shunra's Virtual Enterprise provides an exact replica of the production environment, including the WAN, remote offices, end-users, and traffic, which allows network managers to safely simulate various scenarios before and after deployment. TMC Labs loved the user interface within Visio and its ease of use which camouflages some very powerful functionality under the hood — we would recommend it to any network manager, VoIP vendor, or someone looking to deploy VoIP. IT

If you are interested in purchasing reprints of this article (in either print or HTML format), please visit Reprint Management Services online at http://www.reprintbuyer.com or contact a representative via e-mail at reprints@tmcnet.com or by phone at 800-290-5460.

Beyond The Contact Center With Presence-Enabled Enterprise VoIP

It is ironic that as companies have adopted new contact center technologies to become more customer-centric, they may systematically reduce the actual contact between customers and enterprise employees. However, this narrow view is quickly changing as organizations adopt voice over IP (VoIP (define - news - alert)) solutions and presence-enhanced collaboration tools that dramatically extend real-time customer touch points beyond the confines of the conventional contact center paradigm.

With VoIP and presence, enterprises are learning to create more dynamic views of the contact center model. By applying new communication technologies to route customer dialogues throughout the extended enterprise, organizations are discovering unique opportunities to differentiate on real-time customer service. And, ironically, they are finding that the more they involve a wider swath of enterprise employees in the customer care process, first-contact resolutions of customer queries improves.

"It's simple, really. Presence-management tools enable faster call handling and help to eliminate blind transfers," said Rick Tillotson, the telecommunications manager for the Texas Association of School Boards (TASB), an organization that recently deployed a contact center collaboration system. "These tools are helping us get customers, without delay, connected with the people who can help them the most."

First-contact or first-call resolution remains a major customer care concern. According to the Center for Customer Driven Quality at Purdue, the customer's inability to reach the right person with the right information drives 60 percent of customer dissatisfaction today. Letting calls linger, unresolved, is also an organizational burden. The lack of first-contact resolution accounts for at least 30 percent of a call center's operational costs due to callbacks, escalations, and multiple repeat calls, according to The Call Center Manager's Forum.

VoIP and presence change the equation. At TASB, for example, the organization's abandoned call rate fell by 61 percent only six months after getting its contact center collaboration system up and running. TASB said it also expects to benefit from six-digit annual productivity savings.

"Our presence-enhanced contact center system enabled us to immediately solve our most important contact center challenges: addressing increasing call volumes while meeting the ever-rising expectations of our members for timely and informative responses," Tillotson said.

Breaking Down Barriers

Many enterprise contact centers still carry a design legacy based on a telephony model of wired desktops and phones, mainframe or client/server computer systems, and a relatively static communications environment. Beyond a single point of entry to the organization, contact centers continue to be seen as "buffer zones" that essentially insulate the vast majority of a company's employees from incoming customer interactions.

Even with the adoption of VoIP and presence-enhanced collaboration tools, many organizations have maintained previous contact center management paradigms and used the new communi-

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By Ross Sedgewick

cation tools to simply expand the traditional functions of the contact center to include more remote or home-based agents. But first-call resolution is left unsolved.

Why? Quite often it's because agents lack the knowledge, expertise, authority, or insight to close a customer interaction on first contact. It's understandable. Products and services have become highly complex. Competitive pressure for increased speed to market only complicates the customer communication dilemma. Shortened life cycles, quicker sales cycles, and rapid product time lines only magnify the challenge for call handlers.

Experts and specialists are needed more than ever to clarify product and service information and address issues in a context that each consumer can understand. At the same time, the speed of the Internet has caused customers to demand immediate responses to more questions, on top of greater access and

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more self-service options.

Organizations must rethink how the contact center solution fits into the broader enterprise customer interaction environment. The handwriting is on the wall: enterprises need more organic dialogues and value-added interactions — to the benefit of both consumers and the companies they depend on.

The Power Of Presence

VoIP and presence tools have unique features that can efficiently extend the knowledge boundaries of an organization and better leverage the expertise and specialties that exist throughout the ranks of employees.

In the context of contact centers, presence-enhanced collaboration tools give agents the power to "see" who, within an organization, is available. And, more importantly, who is available with the right skills, knowledge or authority at a moment of customer need. The results include higher first-contact resolutions, more efficient internal operations and stronger customer relationships.

At TASB, for example, 40 percent of transferred calls ended up re-queued or transferred prior to the organization's deployment of a contact center management system. In a typical scenario, a worker's compensation claim would come in to TASB's contact center and, since contact center agents could not see if a specific adjuster was available, callers were often blindly transferred to voice mail. This led to frustrated callers and more repeat calls.

With a presence-enhanced collaboration tool, agents know ahead of time if specific adjusters are available for a call and, if not, whether they are only away for a few minutes or if they're out at a meeting or out for the day. Meanwhile, another currently available adjuster — with the required skills or knowledge — could be identified to help out.

First-contact resolutions also set the stage for pursuing higher-value customer opportunities. Customers who are immediately provided with informa-

Myths And Realities Of Migrating To IP-Based Contact Centers

By Ross Daniels & Sean O'Connell

To provide the best-possible service levels to customers, an automobile manufacturer with over 300 dealerships across the United States maintains four regional technical assistance centers (TACs). These TACs are staffed with dozens of expert technicians who are on call in the event that local service technicians can't resolve a problem quickly enough for customers. With an IP-based customer contact center solution, the company transformed these four regional centers from isolated operating departments into a "virtual TAC" that, among other things, intelligently routes calls to the next available technician, or most knowledgeable, or even the agent who previously took a call pertaining to the same car. As a result, the company can disseminate troubleshooting and repair knowledge more readily, resolve problems more quickly, improve technician productivity, and reduce dealership costs for loaner cars.

Using its IP-based solution, the manufacturer also saves money on airfare and hotels by providing a home tech support setup for any TAC technician who lives more than an hour away from a TAC hub. The company saves on long-distance calls by using toll bypass between dealerships and TACs. The IP-based call processing system cost is just 25 percent of what the company would have spent for a PBX, and the maintenance is about 20 percent of traditional TDM costs. And this does not include the savings associated with maintaining one IP-based system instead of four TDM systems.

Contrary to common belief, you don't have to be a big company to take advantage of IP-based customer contact center solutions to deliver better service, improve productivity, or reduce costs. Companies of all sizes in every industry can benefit from IP-based contact center solutions. This article addresses this and other myths about deploying IP-based solutions.

Myth #1: Migrating to IP-based contact centers requires a complete equipment upgrade.

One of the greatest myths about IP-based contact centers is that you have to throw out everything you have in order to start realizing the benefits of IP. The truth is that a massive upgrade is entirely unnecessary. Companies can migrate easily and comfortably to IP-based contact centers, preserve existing investments as they go along, and realize compelling benefits as they evolve.

For example, if you still have a year or two left on your TDM-based switch lease, you can deploy IP-based contact center software that works across both IP and TDM networks and multiple TDM switch types from different vendors. By having a real-time view across different networks and switches, you can add load balancing across two or more contact centers, roll up reports across multiple centers, and begin to integrate capabilities like advanced speech applications. As the TDM-based switch goes off lease or reaches the end of its life, you can replace it with an IP-based system, or simply consolidate its functionality with IP-based systems previously deployed. When you complete your migration to IP, you have the added benefits of a single network to manage as well as simpler integration of voice, video, and data.

With VoIP and presence, enterprises are learning to create more dynamic views of the contact center model.

Myth #2: An IP-based system is not as reliable as a TDM-based system.

One of the persistent myths about IP-based systems is that IP-based systems are not as reliable as traditional TDM-based phone systems. In reality, IP-based solutions can actually provide better protection against outages. Why is this? In the traditional telephony world, you get "five nines," or 99.999 percent, reliability by adding a second processor to the TDM switch in case the primary processor fails. The "box" is highly reliable, but continuity of operations is certainly not assured. If the whole building is taken offline by a cut T1 trunk or a hurricane, your contact center is down even if the PBX is up and running. In fact, any contact center model that requires agents to be within a certain proximity of a PBX or call processing system limits the ability to reroute around problems.

The distributed nature of IP allows you to move beyond the traditional box-based model to network-based reliability. An IP-based network is designed to route around failures, so if one location goes down you can route calls to other agents in locations that are not affected by the outage. An IP-based contact center model gives you true geographical redundancy.

Myth #3: IP- and TDM-based contact centers offer similar scalability.

It is true that traditional TDM switches come in all shapes and sizes, and some are capable of handling enormous transaction and call volumes. But compared to IP-based tive answers and warm transfers do not have to start over from the beginning to explain their situation since, across VoIP systems, calls are linked with important information about the customer needs. And finally, by efficiently handling the customer's needs, more time is available for possible agent-initiated cross-selling and up-selling opportunities.

Rethinking The Contact Center Model

Customers evaluate an enterprise's service quality based on their total experience with an organization. Companies are recognizing that processes and job roles spanning the enterprise can impact or support customers. In fact, the largest percentage of knowledge, experience, and problem-solving capability resides outside the boundaries of the contact center.

An opportunity exists to alleviate the information barriers between customer-facing departments through increased sharing of customer data among the various enterprise applications and communication systems. Truly customer-centric organizations will use presence-enhanced VoIP solutions to expand customer interactions much beyond the confines of the dedicated contact center paradigm.

Step 1: Leverage Knowledge and Expertise Throughout the Organization

Each knowledge worker in the organization must be identified by a skills resume linked to a centralized communications routing system. A selective contact escalation process will speed up the first contact resolution process and provide a good feedback channel for senior level managers who otherwise may not be routinely exposed to customer situations. Universal interaction routing and queuing across the extended enterprise, driven by business rules and tracked with the same closed-loop discipline of the contact center, is the foundation of an efficient process.

Presence-enhanced instant messaging and chat applications can also allow agents to easily consult with extended enterprise employees in a more accessible, business-friendly manner. IP-convergence will enable skills-based routing across multiple and distributed locations — whether a remote agent, mobile supervisor, home office worker, temporary or mobile agent. User profiles allow high-value knowledge workers to determine how, when, and by whom they can be reached.

Step 2: Create and Manage Structured and Automated Communications

With employees becoming easier to access, it is critical to apply controls around contact routing to contain interaction management costs. The knowledge worker's skills backgrounder will define roles that dictate a worker's accessibility and privileges to view and update customer information.

IP convergence enables the enterprise to apply routing and communication processes to all users, independent of location or media. Soft clients provide remote agents with the same contact center functionality as on-site agents, regardless of location and with consistent business rules, closed-loop tracking, and reporting capabilities.

Step 3: Providing a Single View of the Customer

As the enterprise moves toward a broader customer care framework that leverages the knowledge of the whole enterprise, it is imperative to provide unified access to relevant and timely customer information throughout the enterprise. The ability to present a complete view of a customer's multi-channel interaction history is greatly enhanced by IP convergence and deployment of ubiquitous CRM desktop tools.

Today's CRM application tools are typically confined to the contact center. However, to enable the whole enterprise to serve the customer there must be pervasive CRM tools with a common customer view. Ensure that the enterprise adopts an integrated channel approach versus the tedious and more costly addition of fragmented multimedia channels.

Bringing It All Together

The goal of first contact resolution is not only higher customer satisfaction (and, therefore, retention) but also reducing the cost of repeat calls, escalations, and cumbersome callbacks.

VoIP and presence applications, in addition to providing numerous efficiencies and advantages, can also serve as a major catalyst for re-aligning an entire workforce around customer-focused issues. While the contact center will continue to play a central role in enhancing customer service, enterprise-wide communications models must evolve to better leverage the knowledge an expertise of the whole enterprise to service customers effectively and efficiently. Often, for example, experts within an organization (and outside the call center) need approvals or help from their colleagues to resolve complex customer issues.

Highly effective organizations should extend presence and collaboration capabilities throughout the enterprise's community of workers. In addition to contact center solutions, presence-enabled collaboration portals also exist to help employees link more effectively with peers and across devices such as phones, wireless devices, e-mail, instant messaging, and other media. **IT**

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Companies of all sizes in every industry can benefit from IP-based contact center solutions.

contact center solutions, TDMs scale in a very rigid way that negatively affects both budgets and business models.

In the TDM world, you reach capacity when you run out of expansion slots. At that point, the only way to add more capacity is to buy another expensive, proprietary switch. With IP, you can start small and grow incrementally to match the current needs of your business. And IP-based call processing software runs on industry-standard servers that offer dramatically better economies of scale than proprietary TDM hardware.

Once you get your main site up and running, adding subsequent sites in an IP-based environment is exponentially easier than in the TDM environment. You no longer have to locate agents near the ACD — you can even have your call processing in one location and agents in another city, state, or country. This opens up the possibility for new business models, including:

Home-based contact center agents: Many companies would like the option of using home-based contact center agents in order to offer more flexible work hours for employees, expand their labor pool, extend service hours for customers, save on facilities costs, and more. The proprietary hardware of TDM-based solutions has always made it difficult and expensive to extend call centers to home-based agents. IP-based solutions, in contrast, take advantage of industry-standard hardware to make such configurations affordable, while ensuring that home-based agents have the same real-time access to customer information as their counterparts at the office-based contact center.

Branch-based service: In industries like retail and financial services, stores and branch offices have become an important place for customers to seek service and support. Companies want to be able to empower store or branch personnel to handle customer calls in order to take greater advantage of knowledgeable sales and service people in the field. IP-based contact center solutions employ sophisticated call-routing logic to route calls more intelligently — for example, by location, expertise, next-available service person, etc.

Outsourcing: Many companies today are using customer service agents in different geographies to reduce costs and increase flexibility. IP-based customer contact centers make it easier and more affordable for small and medium-sized companies to use an outsourcing model.

Tangible Business Benefits

IP-based customer contact centers offer even more benefits to companies. You can create virtual customer service operations that allow you to tap into agent skills and availability anywhere and at any time. You can respond to customers in any way they choose, including voice, Web chat, Web collaboration, or e-mail. You can create a more comprehensive knowledge base about customer interactions and agent productivity to help refine your service delivery, and even influence the development of products and services. In reality, IP-based contact centers are the best choice for any service-oriented company. **IT**

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Achieving Convergence Now —

The Opportunity For Mobile VoIP

Convergence between mobile and broadband networks creates a common access platform that will erase the distinctions between fixed and mobile phone numbers. Users will only need one phone identity regardless of where they are located, using the lowest cost network available at home, work, and in public places. Emerging phones will combine WiFi (define - news - alert) VoIP (define - news - alert) access and cellular capability for true seamless mobility, yet pure VoIP devices are already accelerating down the cost and volume curve.

Mobile and VoIP convergence is seen by many as the catalyst that will transform the telecommunications industry — in effect redrawing the competitive landscape for mobile operators, integrated operators who operate both wireline and wireless networks, cable multi-service operators (MSOs), ISPs, and Mobile Virtual Network Operators (MVNOs).

In this article, we will explore how carriers — both mobile and wireline, can build mobile VoIP services and take the lead in converged communications, and examine the technology and the ecosystem of partnerships that might be formed to bring such mobile VoIP services to fruition in North America and around the world.

Threat & Opportunity

To better illustrate both the threat and opportunity of the convergence of mobile and VoIP, here are some statistics to consider.

According to a recent survey by In-Stat MDR:

• 14.4 percent of U.S. consumers currently use a wireless phone as their primary phone and that number is growing as the younger generation, which grew up with the mobile phone, matures.

• The remaining are still using a landline as their primary phone, but according to IDC make 36 percent of their personal calls at home with their cell phone.

• 26.4 percent would consider replacing it with a wireless phone, demonstrating a significant potential for wireline displacement over the next five years.

In addition, incumbent fixed line providers are beginning to realize that they must embrace the lower-cost broadband VoIP technology or, thanks to Local Number Portability, risk losing their customers not only to wireless carriers, but also to new VoIP services entrants.

A second set of statistics shows the threat to mobile carriers. Mobile carriers are starting to see competition from broadband VoIP service providers for indoor communications. With over 40 percent of phone calls made indoors, such services have advantages over indoor mobile phone use in the areas of peak-time cost, coverage, and voice quality. VoIP providers like Vonage are beginning to offer near-mobile telephony using Wi-Fi cordless phones that can operate in hotspots at much lower cost than cellular.

To summarize, fixed line PSTN is already under attack from mobile and now both mobile and fixed line PSTN are in turn threatened by VoIP.

The Drive To Convergence

The solution to these threats lies in the convergence of mobile and VoIP. Convergence draws the competitive lines amongst carriers according to the network assets they own (see Figure 1).

• Integrated carriers that own both fixed and mobile networks, such as Bell

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By Sanjay Jhawar

Already, broadband carriers are adding mobile services to move beyond the triple play of voice, video, and data. The quad play, including mobile, is their new battleground.

Canada, Deutsche Telekom, and France Telecom are arguably in the best position, unless the regulator hampers their ability to bundle converged services at discounted prices (as has occurred in Korea, for example). They can upsell both their fixed and mobile customers to the convergence service.

• Affiliated wireline/wireless carriers, such as SBC (quote - news - alert) /BellSouth (quote - news - alert) /Cingular, Verizon Communications/ Verizon Wireless (quote - news - alert) need to cross organizational boundaries to put their convergence assets to work.

• Independent mobile carriers, such as Sprint/Nextel (, T-Mobile USA, Vodafone, and others, and fixed local and long-distance carriers such as BT, AT&T, Qwest, and cable MSOs, such as Comcast, Time Warner Cable, Cox Communications, etc., need to strike partnerships to put the convergence play together.

There are two partnering models open to independent carriers: the MVNO model and the roaming model.

In the MVNO model, a provider without a mobile network (the MVNO) resells a mobile carrier's service under their own brand, purchasing minutes at a wholesale rate. In the United States, Sprint has enabled Virgin Mobile and Qwest to do just this and has announced similar relationships with AT&T and even ESPN. The MVNO takes the financial risk of handset subsidies and subscriber acquisition but does not operate the network.

In the roaming model, a provider with a fixed line VoIP service could strike a roaming agreement with one or more mobile carriers. When calls made to the user's mobile number are terminated on the VoIP/broadband connection, a charge appears on the cellular bill. Conversely, when calls made to the fixed VoIP number are delivered instead to the mobile phone, the charge appears on the VoIP bill. In both cases a settlement must occur between the service

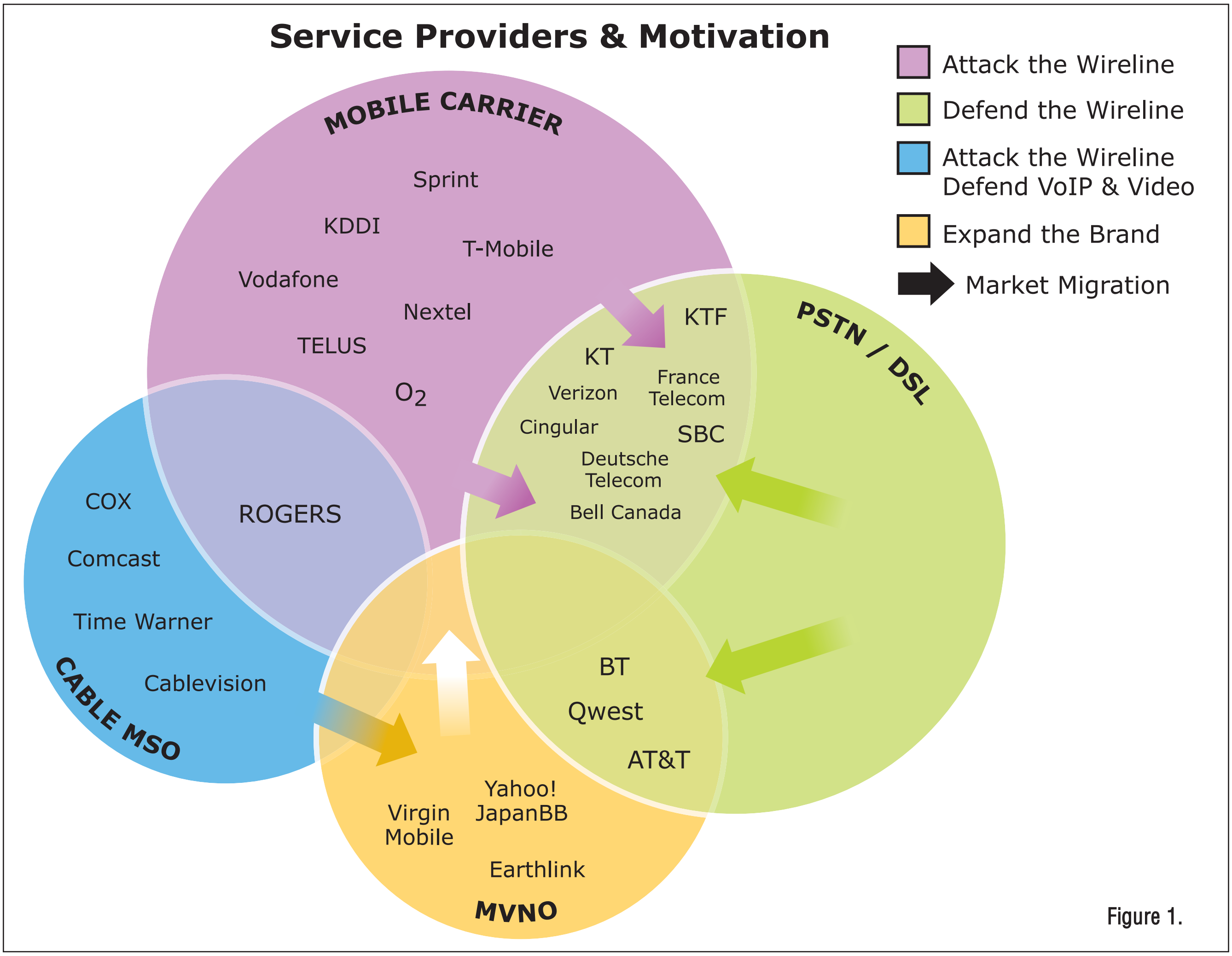

Figure 1.

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providers. This roaming model has existed between mobile providers for years, but can now be extended to work between VoIP and mobile providers, and can offer independent service providers a way to offer convergence services without the expense of handset subsidies and large scale marketing campaigns.

In November 2004, reports surfaced that several U.S. cable MSOs were negotiating with investment bankers to form a mobile communications joint venture — which would give the cable MSOs the leverage to strike MVNO or roaming agreements on more favorable terms — including the required access to the cellular core network (especially the Home Location Register or HLR) to make their subscribers' existing VoIP numbers mobile, instead of needing to issue their customers new numbers for mobile use.

Roaming Seamlessly From Mobile To VoIP

Mobile VoIP services can be built using both new converged devices which combine cellular and Wi-Fi VoIP capability, and existing cellular phones in combination with standard, fixed SIP compatible VoIP phones.

Converged devices offer the potential of seamless handover of an in-process call from cellular to Wi-Fi and vice-versa. Such capability is most valuable in an enterprise setting, ensuring coverage gaps within and between campus buildings do not impede an ongoing business phone conversation. However, these devices will have cost, size, and performance challenges for the next couple of years, which indeed limits their use to enterprise users who justify these tradeoffs on the basis of improved productivity.

Conversely, for residential consumer use, seamless handover is a nice-to-have — since houses are usually smaller than enterprise buildings and device cost is more of an issue. For these users, the ability to use low-cost and widely available existing cellular and VoIP phones

Trends In Messaging

By John Finch

Thanks to the recent proliferation of VoIP, service providers are able to develop and introduce new applications and options — particularly in the areas of messaging. In the months to come, companies that support VoIP communications will begin to roll out many of these new features. In fact, some companies are using a few of them already. So, what's out there, and what should businesses know about so they can make informed technology decisions before their competition beats them to the punch?

New capabilities such as visual voice mail and VoIP voice mail retrieval over broadband wireless promise to improve 2.5G and 3G return on investment. Along with "voice mail offload," these features will create new revenue streams and free-up capacity on the narrowband cellular network for adding subscribers. The industry also can expect "voice mail offload" to become for the wireless world what primary rate interface (PRI) offload has become for the wireline world.

Offloading Voice Mail

The demand for mobile services continues to grow rapidly, and voice mail is an important component of the wireless offering. Some experts suggest that voice mail is responsible for connecting up to 40 percent of mobile calls, and without voice mail, the revenue from all of those connected calls would be lost. The only downside is that typical voice mail retrieval occurs on the voice network, swallowing up voice bandwidth and driving up capacity requirements. As a solution to this, some mobile network operators (MNOs) are looking into voice mail offload, which will help them optimize their capacity while reducing capital expenditures.

With voice mail offload, voice mail retrieval traffic is removed, or "offloaded," from the digital voice network and re-directed to the packet data network. The process requires a flexible, advanced voice/data network architecture.

Wireless IP access on broadband networks, IP switching and sophisticated mobile devices have set the stage for the creation of voice mail application platforms capable of providing voice mail offload capabilities. Such a platform requires simultaneous support for TDM and IP connectivity. The TDM functionality provides a connection to the circuit-switched voice network. The IP functionality connects to the IP network. The combination of the two makes it possible to remove the voice mail retrieval traffic from the wireless digital voice network and place it on the wireless IP network.

Offloading the voice mail from the narrowband cellular network onto the wireless broadband network enables MNOs to enjoy the benefits of an IP core network. IP networks are less expensive than circuit-switched networks, and they are more efficient. By transporting voice mail retrieval traffic over the IP network, the MNO can increase the capacity in its voice network. That increased capacity makes room for additional subscribers. With such a setup, MNOs also will have the beginnings of an infrastructure that will support the transmission of voice and data simultaneously.

together, using a single phone number is a good solution until the converged devices are more widely available. This IP version of call forwarding can be made automatic utilizing presence technology, and realizes cost savings by avoiding the long-distance circuit-switched network.

The required network infrastructure depends on a capability to bridge cellular roaming into the VoIP (SIP (define - news - alert)) world — a Network Convergence Gateway (NCG). Such a gateway can send calls made to mobile phone numbers to external broadband connected VoIP platforms and SIP endpoints. It can also accept SIP calls from external VoIP platforms like softswitches (define - news - alert), IP Centrex (define - news - alert) platforms, and IP PBX (define - news - alert) platforms and send them over IP into the mobile network avoiding a circuit-switched call forward, thereby maintaining the superior cost economics of VoIP.

With these technology building blocks, a number of solutions for end users can be constructed (See Table 1).

Bundled Services, Integrated Billing

As converged networks provide subscribers the ability to roam on one phone number between mobile and fixed networks, new revenue opportunities will arise through the bundling of services. Mobile VoIP actually reduces subscriber churn by harnessing the very disruptive technologies that threatened to dramatically increase churn.

A Whole New Way To Check Messages

In addition to the service-provider trend of voice mail offload, retrieval of voice mail by the end-user also is going wireless. We're not talking about the voice mail messages left on a wireless phone. Wireless voice mail retrieval from the end-user perspective involves logging onto the Web via a mobile device and accessing a visual voice mail system on a desktop PC.

With this type of voice mail system, it would be possible to check a central mailbox that contains voice mail, e-mail, and any other messaging from a mobile phone, wirelessly enabled PDA, or various other mobile devices. The beauty of such a system is that, because it uses the broadband network to provide access to the voice mail information, it does not tie up the voice channel, and it enables providers to collect revenues for broadband network usage from mobile devices.

WiFi devices also fit into this category. With the visual voice mail setup, it would be possible to use a WiFi-enabled laptop or other mobile devices to access the broadband network and connect to the central mailbox. Once there, the user would have control over how to listen to, read and/or respond to messages left in the voice mailbox.

Of course, this type of technology is not widely adopted yet, but there are businesses using these kinds of visual voice mail and conferencing-on-demand features as part of their VoIP and mobile communications.

In addition, because VoIP operates on a system of media streams, it is less expensive for providers to offer advanced conferencing features. And, based on the cost savings and ease of use, this type of technology is likely to spread in the months to come.

Conclusion

Each of the aforementioned trends is beginning to take shape, and, in some cases, service providers are employing the use of these technologies today. Messaging applications will continue to play a major role in the advancement of VoIP technologies. As service providers find new methods of using messaging capabilities, the messaging platforms that support those applications will have to be flexible to accommodate the changes.

Regardless of whether all of these trends come to fruition tomorrow or a few months from now, they will provide service providers with new sources of revenue, and they will help end-users to expand they way they communicate. IT

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Table 1.

	Description
Roam-to-Home	Subscribers roam their *mobile* number into their homes via one to many SIP devices.
Roam-to-Mobile	Subscribers roam their *wireline* VoIP number into the mobile network via their current mobile device.
Roam-to-Work (IP Centrex integration)	SME subscribers roam their *mobile* number into their business environment via one to many SIP devices, and/or their wireline number to their mobile phone when out of the office. *Managed Service from a carrier.*
Roam-to-Work (IP PBX integration)	Enterprise subscribers roam their *mobile* number into their Enterprise environment via one to many SIP devices, and/or their wireline number to their mobile phone when out of the office. *Interworks with enterprise's own telecom infrastructure.*
Roam-to-Campus	Student subscribers roam their *mobile* number into their campus environment via a dual mode phone that supports MMS etc.
Roam-to-Hotzone	Public Access subscribers roam their *mobile* number into public access networks, including hotspots via one to many SIP devices, including a PC softphone and can bypass international cellular roaming charges.

The consumer can conceivably sign up for broadband service which includes VoIP calling, and also get mobile services on the same number. Furthermore, the services can be billed together by the broadband provider, saving the provider costs of separate bills and the user the headache of paying multiple bills. Alternatively mobile carriers can offer a cost-effective PSTN replacement service using the mobile number, with calls delivered on the subscriber's broadband connection when they are at home or work.

Already, broadband carriers are adding mobile services to move beyond the triple play of voice, video, and data. The quad play, including mobile, is their new battleground.

It Takes An Ecosystem

Providing for a smooth arrival for mobile VoIP will be the job of many. Not only will the SIP-based protocol and other open standards be a centerpiece of mobile VoIP's development, the circle of partners who work to make this happen will need to coordinate efforts.

Those who provide the infrastructure software to link networks to each other, allowing mobile to VoIP roaming to work will play a key role. VoIP and mobile infrastructure vendors will increasingly focus their offerings on this key market, which has the potential to remake vendor positions. Also key will be the systems integrators, handset makers, and other hardware providers who create VoIP convergence devices.

The convergence of mobile and VoIP services will also benefit intermediary service providers who will mediate inter-carrier relationships and provide the roaming, clearing, settlement, and managed security services.

RELATED LINKS

3GPP
www.3GPP.org

Wi-Fi Alliance
www.wi-fi.org

SIP Forum
www.sipforum.org

The Future — When VoIP Moves To IMS

While voice and messaging is still the largest driver of consumer behavior and service provider revenues — much attention has been paid to the transmission of video, data and larger files to and from mobile devices. The mobile network platform for these services is known as the IP Multimedia System (IMS)

IMS is heavily based on SIP technology in the core network, and is in the early stages of vendor trials today. The architecture is very flexible and has appealed not only to mobile operators, its original design point, but also to fixed operators who know that convergence is a necessary part of their future. However, much remains to do on interoperability and compliance to the complex standards set by 3GPP and 3GPP2, leading to a wide variance of opinions on when the system will be commercially deployable – from late 2005 to 2009. Also, the demand for mobile video telephony, TV, push-to-talk, etc., has proved hard to forecast, resulting in uncertainty on the timing of revenues. Thus, the business case for such a significant core network upgrade has remained difficult to make.

Summary

It is not a question of whether VoIP and mobile services will converge — but a question of the best path to take. While several approaches have emerged, the choice to use open standards, primarily SIP, which provide the call control flexibility to implement many different user experiences and service provider business models, will accelerate the development of the ecosystem of vendors and technologies that carriers will use to implement mobile VoIP.

Carriers must seize the mobile VoIP opportunity to meet the VoIP threat and simultaneously create the business case bridge for full next-generation network deployment based on IMS. IT

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The Fourth Age Of VoIP

Now that we've arrived, it may be time to pause and reflect. But when I say "we've arrived," where exactly are we? Where is "here?" VoIP (define - news - alert) is a viable, commercial reality. The audio quality is acceptable, there is a standard protocol that everyone supports, and successful deployments occur so often that they're not news any more. Vonage (news - alert) has announced its four hundred thousandth customer, and every major company is or wants to be a player in what has become, once again, a hot space.

First-generation VoIP basically didn't work. Good thing it was free!

I think "here" is second-generation VoIP. It does work. But it works at replicating all of the pomp and circumstance, all of the artifact, of the PSTN. We could call it "PoIP"—PSTN over IP. The same old 12 keys comprise the primary means to input commands into the system, the same old muddy monaural audio emerges from the speaker. The same blind handset lets us try, again and again, to call people who are patently unavailable (or unwilling) to answer.

So we've replaced RJ-11 with RJ-45, and traded some reliability for some cost savings. Is it any wonder that Vonage had to spend around a hundred million bucks to get four hundred thousand people to subscribe ($250/customer)? I don't think conventional PSTN service, at least domestically in the U.S., is expensive enough for most people to spend time and effort, much less risk, to replace it.

We're seeing the edges of third-generation VoIP now. There are products that center around the buddy list, rather than the dialpad. Some people are using software that lets them begin with text (where most business calls probably should start) and escalate to voice if appropriate. Ad-hoc conferencing is available, which lowers the barrier to multiparty meetings and speeds the flow of knowledge in an organization.

A recent article in this publication pointed out that there is a large improvement in market efficiency due to separating the transport from the application, and the media from the signaling. This is true. But I think the biggest value proposition for VoIP has yet to be fully realized.

I think we need to go back and look at the PSTN's architecture and business model to understand which of its properties are mere accident, and not essence. When we set out to build its replacement, we want form to follow function, and we want to preserve only the good parts of the PSTN.

An example of something which may not be one of the good parts is the concept of the line. To make a call, one used to have to lease a copper circuit from end to end, for the entire duration. The cost of this lease was based on the length of the copper and the time of the call. This encouraged a culture of not dialing until one had something really important to say, conducting business as quickly as possible, and hanging up. If there was no cost per minute, would this still be the behavior?

At the moment, yes, because even if call time were free, one would still have to free up the line. But what if virtual lines were unlimited?

One would still have to hang up, because if two people talk at the same time, there is no intelligibility with monaural audio. In a live face-to-face meeting, this isn't a problem because the ear can pick out any one voice and ignore the others as noise. I believe that stereo audio, and virtual 3D positioning of each voice in the sound field, is the single feature that enables the next generation of VoIP. Fourth generation VoIP offers value propositions other than cutting costs.

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By Keith Weiner

It is ironic that a new technology for communications is promising to reduce the amount we communicate.

IM is so successful because it allows multiple conversations at the same time. This doesn't necessarily mean frantically typing and rapidly switching windows like a music video switches camera shots. It means leaving channels open with one's contacts. The value of this is that the barrier to communicating is lower, and information can flow faster. How many times per day, in every enterprise, does it happen that someone looks at his watch, decides that it's late in the day and since he doesn't have the information finalized, will call tomorrow (or Monday)?

4GVoIP adds voice to the IM model. With a more informal environment, it is easier for people to communicate earlier. This heads off problems sooner, and enables companies to be more competitive.

We talk about convergence today, but often mean convergence of services (e.g., the triple play), or of devices (e.g., the smart phone), or of different phone numbers (e.g., find-me-follow-me). But it's also about different applications, specifically text, voice, and conferencing. This will lower the barrier to multiparty meetings.

It gets more interesting when combined with presence information. The problem with ad hoc conferencing is that often one or more members are not available at the moment. The problem with scheduled meetings is that they tend to be put off a few days. I think the opportunistic conference, created as soon as all members are available, will be the most common type of conference.

Beyond conferencing, multi-conferencing enables tighter teamwork between remote colleagues. Companies install open office plan furniture to enable "peripheral listening." This lets coworkers help one another and learn faster. This might be just the feature to enable a successful "virtual office" comprising employees in Birmingham and Bangalore. Challenges related to the time zones and language and culture do not need to be exacerbated by trying to sip communications through a thin straw.

It also provides a back channel. How many times does an engineer support a salesman on a customer call, and become frustrated listening to promises that he can't keep even if he bleeds? He doesn't want to embarrass the salesman, nor does he want to let the conversation continue past promising a cure for cancer to world peace and balancing the Federal budget. If he could type in a different window or talk privately to the salesman, he could save the day.

A great deal of the value of VoIP will come from presence information. I don't merely mean the "online/offline" indicator that's common today. Users need to switch from one device to another as they move about. Presence can tell potential callers which device the user is on. More importantly, it can automate find-me-follow-me. But the biggest benefit could be to make it simple and easy to control access. When I am in my office, I expect to be reachable by certain people, but not others. Out at lunch, I may want this list to change. At home, especially after 10pm, it's different still.

I say "could be" and "may" because I don't presume to dictate how people will turn out to use it. I know that the culture changes around the new capabilities. For example, voice mail and caller ID have allowed people to screen callers. Some people are consistent about calling back when they have the time. Other people develop reputations as being never available. Presence information will allow a new, more sophisticated way to lie, "I am not available (to you)." How people will use it, and what will be tolerated versus being perceived as unacceptable, is unknown.

If you think about it a certain way, it is ironic that this is the first time that a new technology for communications is promising to reduce the amount we communicate. But, actually, we are inundated with too much communications that we don't want, and sometimes this forces us to ignore or even delete important messages (e.g., with spam).

Conclusion

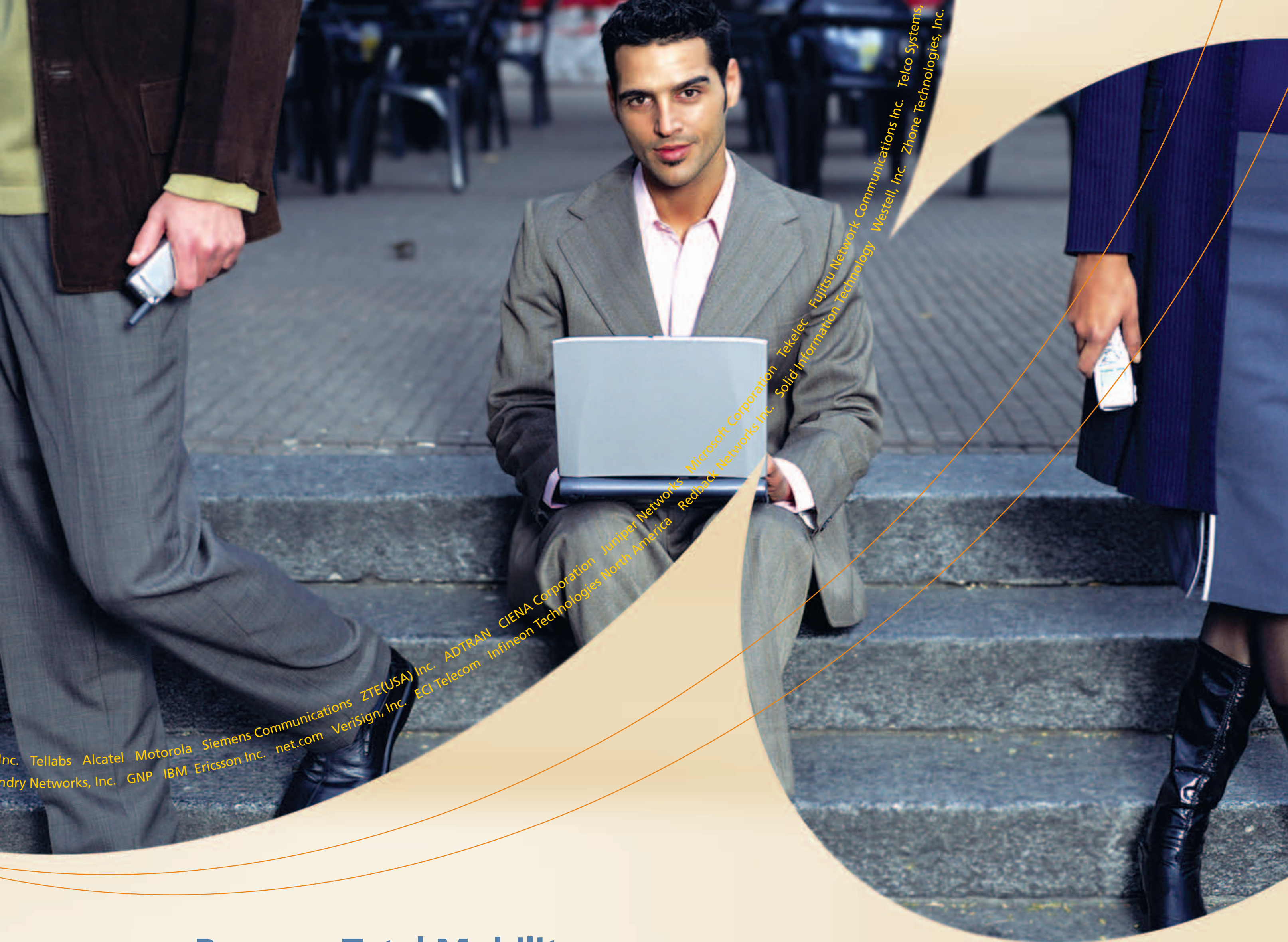
For the sake of sanity, if not productivity, we have to retake control of our communications lives. I wonder how many people turn off the phones in their houses at night, but then regret missing an emergency call from a hospital or loved one in need.

Of course, this cannot come at the expense of simplicity. The PSTN is not well designed from a user experience perspective, but it is pretty simple to use. You dial a number and hear a ring, busy signal, voice mail, or even the party to whom you wanted to talk. Few people would replace this with a system of endless cascading menus and submenus to control a million variables. On the other hand, the alternative may be to shut out the world. What will happen to the volume of unwanted calls when the cost per call is zero?

The purpose of this article was to be a starting point, not the "final answer." I think we all need to think about how VoIP can get away from the legacy of the PSTN. Even if you disagree with everything I else said, I have accomplished my goal if you see VoIP in a new light. Think about what you would design with a clean slate, if you had an always-on high-speed network with supercomputers as the edge devices…

Would it really be a cheaper PSTN service? IT

Keith Weiner is CEO of DiamondWare. For more information, please visit the company online at http://www.dw.com.



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Investing In The Future Of Convergence

Financial institutions know all about bottom-line incentives. Voice and data network convergence provides one such incentive for banks and credit unions to save sizable dollars while enhancing their call center functionality.

Voice system manufacturers such as Avaya (quote - news - alert), Nortel (quote - news - alert), NEC (quote - news - alert), and Mitel (news - alert) have now stopped all Research and Development on traditional telephone systems and are instead focusing exclusively on architecting converged platforms. And they will stop supporting traditional voice systems altogether by 2007. In other words, they are 'end-of-lifing' current systems to make way for the new Internet telephony convergence platforms. Leaders in the financial services arena are looking at these industry drivers and realizing there is a near-term urgency for business decision-makers such as themselves to ponder their tech convergence options and plan accordingly.

In 2004, the voice systems market traveled beyond the early-adapter phase of these new technologies and segued into the 'fast follower' stage. Knowing that this stage is the optimal time to employ best practice strategies, Affinity Plus Federal Credit Union (APFCU) recently created a significant ROI by embracing the convergence process.

"The primary goal of this project was to enhance the performance of our Member Relations by extending call center functionality to all APFCU locations," states Keith Malbrue, Chief Information Officer of Affinity Plus. "We realized that an enterprise-wide call center solution would provide better utilization of our employees as well as overall better service to Credit Union members. In addition, converging our voice and data networks would give us greater efficiencies and cost savings."

The largest credit union in Minnesota in terms of revenues, Affinity Plus recently converged its 20 locations into a centralized model with advance featuring that now accommodates 275 agents versus the previous 35. This extended skills-based contact center provides routing alternatives that have enhanced customer service and reduced hold times. Customer inquiries are not only answered quicker, call distribution is more evenly managed and voice mail capabilities are centralized. The model also utilizes subservient server gateways at remote locations for survivability and strengthened disaster recovery capabilities. The overall design of the new system enables Affinity to sustain higher employee retention rates and to better understand staffing needs.

"Our move from traditional TDM to IP PBX (define - news - alert) and contact center has benefited us by enhancing our agent productivity" says Brad Wampole, Affinity's Director of IT. "More streamlined workforce management has resulted from our skills-based call routing. Management of the system is easier since we've reduced our num-

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By Lynda Treanor

ber of vendors from six to two. There are fewer bills to review on the accounting side; service issues are more easily resolved; and maintenance costs are no longer a major factor. Overall, we feel that we now have a notable competitive advantage over the other major credit unions in our region."

The financial industry as a whole grapples with high employee turnover rates, redundant training issues, and long customer hold times. Affinity Plus had to deal with the same concerns. These were the fundamental reasons that prompted them to take action on convergence. Other factors included:

- hardware technology was out of date;
- upgrading was cost prohibitive compared to a new system;
- downtime during an upgrade would lessen service capabilities; and
- administrative efficiencies were needed to facilitate continued growth.

With their old system, 13 of the Affinity Plus offices had been using archaic analog systems. In the new model, high-end IP telephones have replaced the old analog phones in these outstate branches. In the metro Twin Cities, duplicate point-to-point T1s have been eliminated, with voice and data networks converged on a single T1. At headquarters, centralized call processing is the hub of the wheel, with redundancy features for additional survivability.

Utilizing existing staff at remote sites offers ideal flexibility for call center staffing. Skills-based routing allows calls to be directly routed to the correct resource to better handle customers' needs.

Moving away from Frame Relay allowed Affinity to be more confident in WAN routing time-sensitive traffic. Affinity was not required to change or add staff to support this new solution. The centralized and Web-based system administration gives their existing employees better accessibility and visibility into remote sites.

The Metro branches did not see many technology changes, although Unified Messaging and additional redundancy are key benefits to these branches.

Service capabilities at the remote branches have been measurably improved. The ability to have a higher speed connection for data heightens the productivity of existing users. Providing direct inward dial, multi-line phones with voice mail, unified messaging and branch-to-branch free calling — as well as other notable features — has enhanced the overall customer experience.

"Customers are demanding that converged systems have the same features and reliability of traditional systems.

Making a change to a converged solution is based on current business drivers, technology opportunities and cost savings," says Nortel's Director, Paul Thieken.

Multi-site financial service organizations dependent on customer contact centers at the heart of their business are implementing convergence strategies primarily because of this improved agent productivity.

"The efficiency of such call centers can be greatly improved on IP platforms," notes Scott Strand, a telecommunications consultant with N'compass, Inc. "Greater flexibility in servicing customer calls is a defining benefit of converged voice. This is compelling more and more companies to consider their convergence options. In fact, telecommunications experts expect over one million such call center agents will be using IP platforms in the U.S. by 2007."

Several primary organizational barriers set up blocks for many financial groups — as well as other businesses — from moving forward with convergence implementations. Voice and network work teams — sometimes known as 'the cats and dogs' — have until recently typically operated with silo mindsets in different departments. They lived in entirely different camps and communication was terse at best. Now they are being integrated and forced to work together — like it or not — and are often managed by one CIO. Networks originally built to provide access to common-area stored files now need to also provide support for voice, video, and security demands. Just a few years back, interruptions to Internet connectivity were commonplace, but now no outage is acceptable. The exhortation *Always On!* demonstrates how expectations have changed and why leading-edge financial institutions are moving past all such delay-causing barriers.

Many banks and credit unions postpone IP telephony implementations because of initial bottom-line concerns. The cost to upgrade a traditional voice system usually averaged 50-60 percent of the cost of the original platform. While it is true that the cost-to-build dollars for an initial IP installation are higher, the cost-to-provide-services figures are substantially lower.

Affinity Plus figured that the cost to upgrade their voice network would come to 120 percent of the cost of a new system. But after the cost of new technology was factored in, the credit union will save $1.5M over five years because of their system replacement. Connectivity via a wide-area network has eliminated long-distance fees between branches. And of course service contracts and maintenance fees are significantly lower.

"Affinity Plus had been looking for a solution to improve their customer experience… this convergence solution did exactly that and more. A virtual call center offers extreme flexibility in making sure that each customer call is answered as quickly as possible and by an agent who has the skills to help the customer independent of the agent's location. In addition, Affinity's overall disaster recovery capability was greatly enhanced," sums up Mike Cline of Cisco Systems.

Overall, the infrastructure savings at this statewide credit union are more than paying for the upgrade to their new IP PBX and phone systems.

"We did need to upgrade our network to be prepared for IP telephones," points out Keith Malbrue. "But the overhaul was well worth it when we saw that our Credit Union members benefited, our employees benefited and our organization benefited. Such a win-win-win equation is hard to beat."

According to Keith Meierhofer of N'compass, there are myriad business benefits to be had from IP convergence. Some primary questions to consider when assessing how soon to embrace the process include:

- How can you match your business drivers with technology options?
- How can you sort out hyperbole from fact in manufacturers' marketing claims and, with many vendors to choose from, how can you match the best product to your business requirements?
- How can you quantify and capture the best value of IP convergence?
- How can you leverage existing technologies when creating a common transport network and architect IT infrastructures to handle new demands?
- How can costs be reduced while improving user capability/functionality?
- How can IP convergence enhance reliability and increase disaster recovery capabilities?

Overall, the savings at this credit union are more than paying for the upgrade to their new IP PBX.

The decision by voice system manufacturers to no longer invest R&D dollars in their traditional systems isn't just a minor change in course for 2005. It's a major transition, even a tectonic shift. We are fast approaching a technology tipping point when video, voice, and data convergence will be the rule, no longer the exception.

This move towards convergence can in fact be compared to other notable industry accelerations such as the change-over from propeller engines to jets. Affinity Plus Federal Credit Union is one financial services example of why it makes bottom-line sense to invest in the future by analyzing your IP convergence options now. IT

Lynda Treanor is a senior technology consultant at N'compass, Inc., a consulting firm specializing in telecommunications and technology systems design, maintenance solution services, and optimization. For more information, please visit the company online at http://www.ncompass-inc.com.

VoIP MARKETPLACE

CEO Spotlight

Anders Gustafsson
CEO
Spirent Communications

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 Anders Gustafsson, CEO of Spirent Communications (news - alert).

GG: What is Spirent Communications' mission?

AG: We make it more economical and efficient for our customers to develop, deploy, and assure next-generation equipment and services on a worldwide basis. We do this with our performance analysis and service assurance systems as well as global testing services. Equipment manufacturers depend on us to help them quickly deliver solid, new equipment and systems. Service providers depend on us to validate equipment performance and interoperability during evaluation and trials and to assure the performance of their networks and services. Large enterprises with business-impacting infrastructures also depend on Spirent to make sure those infrastructures perform with 24/7 uptime.

Technology performance, whether in VoIP (define - news - alert) networks, cell phones, user authentication servers, or optical access nodes, can only be analyzed and assured by staying ahead of the technology curve. Our customers depend on us to do just that.

GG: What sets Spirent apart from your competition?

AG: People know Spirent for the breadth and depth of our portfolio and for our expertise. We offer one of the broadest portfolios of leading-edge products, technology, and services available today for lab and network testing, from the physical to application layers, and from legacy to next-generation networks. We employ highly skilled people who have deep experience in carrier operations, and we multiply our value by applying our areas of expertise across both the product development and network deployment lifecycles. Again and again we are told by our customers that when it comes to developing solutions to test and assure next-generation networks there is no better partner than Spirent. Our customers rely on our expertise and trust in us enough to bring us to the drawing board with them to create solutions that allow them to deploy next generation technology and services.

GG: What possible hurdles to VoIP momentum do you anticipate?

AG: The momentum is certainly for real. The business drivers are established, so now there is no going back. Regulations and other factors may slow things down, but in the long term, the business drivers will prevail. The hurdles that remain are really about the industry bringing the technology up to the level of robustness and reliability of circuit switched networks.

One major challenge lies in achieving robust interoperability. Protocol standards that are not yet mature can often result in varying implementations, causing interoperability challenges. Another potential challenge is a kind of growing pain — enthusiasm over new technology can sometimes lead to its introduction before complete testing and validation. We provide a safeguard against such errors by helping the industry apply the time-honored principle "measure twice and cut once." Another issue is a risk that the new network may not scale fast enough to reliably replace the legacy networks — an area where Spirent's technology and expertise can really help.

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

AG: IP telephony accelerated in 2004, and that will continue. We are helping customers assure five 9's in IP telephony, so they are already overcoming critical hurdles. Evolving business drivers such as Triple Play — voice, data and now video — will increase complexity and could temporarily slow down VoIP growth. Security and E911 issues will continue to present even more challenges.

More critically, legacy and next-generation technologies will coexist for a long time, so requirements to test across IP and legacy will be with us for years to come. IP infrastructures need to become much more robust for people to become as comfortable with them as they were with the PSTN. This will take time, but it will happen, and as converged networks are widely deployed, the notion of a distinct VoIP industry will likely melt into a multimedia reality.

GG: In what technology areas is Spirent Communications especially focused, and why?

AG: We have a major focus on IP networks and technologies, where we con-

We invest our greatest energies wherever we believe we can open up new markets and opportunities for our customers.

tinue to invest heavily and gain market share. Meanwhile, we are also applying our expertise and integration to video quality and Triple Play — areas that will be driving a lot of industry activity. Security is a major area and we are the leader in security testing for enterprises and equipment manufacturers. We also are emphasizing wireless and advanced access networks, including passive optical.

We invest our greatest energies wherever we believe we can open up new markets and opportunities for our customers. For example, many service providers may have the determination, but not the manpower or expertise, to roll out new networks and services. Consequently, a major growth area for us lies in providing testing services such as certification and training as well as professional services such as implementation, outsourced facilities, integration, engineering, and consulting.

GG: What is in store for the testing and service assurance industries?

AG: Many companies are creating waves of innovation in IP telephony, security, Triple Play, fiber, and wireless. Still, it is becoming harder for smaller players to sustain themselves as service providers carefully measure the risks of investing in small suppliers. That will accelerate the lifecycle of startups and create more industry consolidation. Because of the jobless recovery and ever increasing network complexity, service providers are having a difficult time maintaining in-house expertise in so many areas, so they will need to rely on third-party specialists like Spirent Communications. The testing industry will need to keep ahead of it all. IT

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