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INTERNET TELEPHONY

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Vonage Readies Its IPO

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The VoIP Authority Since 1998™

Fixed/Mobile Convergence

... on the wings of IMS

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- The Network Neutrality Debate
- Has Video Peering Come of Age?
- Avoiding VoIP Fraud

INTERNET TELEPHONY®

Group Publisher and Editor-In-Chief, **Rich Tehrani**
(rtehrani@tmcnet.com)

EDITORIAL
Editorial Director, **Greg Galitzine**
(ggalitzine@tmcnet.com)
Associate Editor, **Erik Linask**
(elinask@tmcnet.com)
Contributing Editor, **Johanne Torres**

TMC LABS
Executive Technology Editor/CTO/VP, **Tom Keating**
(tkeating@tmcnet.com)

ART
Senior Art Director, **Lisa D. Morris**
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Editorial Offices: 203-852-6800
Customer Service: For all customer service matters, call 203-852-6800.

ADVERTISING SALES
Sales Office Phone: 203-852-6800
Sr. Advertising Director - Eastern U.S.; Canada; Israel
Anthony Graffeo, *ext. 174,* (agraffeo@tmcnet.com)
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John Ioli, *ext. 120,* (jioli@tmcnet.com)
Business Development Director - Western U.S.
Drew Thornley (dthornley@tmcnet.com)

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

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Circulation Director, **Shirley Russo,** *ext. 157*
(srusso@tmcnet.com)
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Sales Office Phone: 203-852-6800
Global Events Account Directors
Joe Fabiano (jfabiano@tmcnet.com)
Maureen Gambino (mgambino@tmcnet.com)
Chris Waechter (cwaechter@tmcnet.com)

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Fax: 203-853-2845 and

The VoIP Authority

By Greg Galitzine

A Walk Among The Bones

Change. Growth. Evolution.

My wife and I took our children to the American Museum of Natural History in NYC this past week, and it was an interesting study of human evolution. I'm not talking about the dioramas and displays, I'm talking about the mass of humanity and the complete lack of consideration for one another. Baby strollers used as battering rams, toddlers used as lead blockers to get through the crowd, I almost felt bad for the other museum-goers. I'm only half-kidding. I'm sure students of anthropology would have a field day just observing the behavior of people who go to museums on national holidays.

On a more relevant note, I could see parallels to the VoIP (define - news - alert) industry amongst the fossils and bones of the Saurischian dinosaur exhibit (think T. Rex). For me, a walk among the bones was akin to a stroll down memory lane of vendors, service providers, magazines, and trade shows that were once the cream of the telecom crop. Life changes and industries — much like species of flora and fauna — evolve.

Rich Tehrani addresses this issue somewhat in his Publisher's Outlook, how the industry has changed and how companies such as Vonage and Google have arrived at this point in their evolution and how that's reflected in the overall industry's development.

Indeed, a quick glance at the topics covered in this month's issue of *INTERNET TELEPHONY* serves as proof that we've evolved: we're most certainly "not in Kansas any more." Fixed/Mobile Convergence, Open Source development, VoIP Peering, Enterprise Wireless… I certainly wasn't writing about these things in our first issue, and now these technologies are critical to the future of our industry.

As the industry evolves, so do we and, in addition to launching new publications this quarter dedicated to SIP and IMS, our conference team has been hard at work building two new "in-person" offerings: IMS Expo and VoIP Demo. These two additions to our conference/exhibition stable address two very different constituencies, but still retain their roots in our desire to educate as many people as possible regarding the future of telecom.

IMS Expo, to be collocated with this fall's Internet Telephony Conference & EXPO West (San Diego, CA October 11–13, 2006) will feature a conference program designed to educate attendees as to the how and why of IMS, while the exhibit hall portion will showcase the industry's early leaders and the products and solutions they have on tap. For more information, please visit http://www.imsexpo.com.

VoIP Demo addresses a more pressing need: The desire for attendees to see live demonstrations of products and services so that they might make informed purchasing decisions when it comes to VoIP technology. This event is set to take place in Santa Clara, CA from August 8–10, 2006. For more information, please visit http://www.voip-demo.com.

We take our mission to educate our readers and conference attendees seriously. We also strive to be the single best source for providing information for decision makers looking to purchase VoIP products and services. We believe the launch of these two new conferences fulfills that mission. I hope to see you at one of these events.

I'd love to hear your thoughts on how the industry and this publication have evolved over the years. Drop me a line and share your thoughts at ggalitzine@tmcnet.com.

Greg

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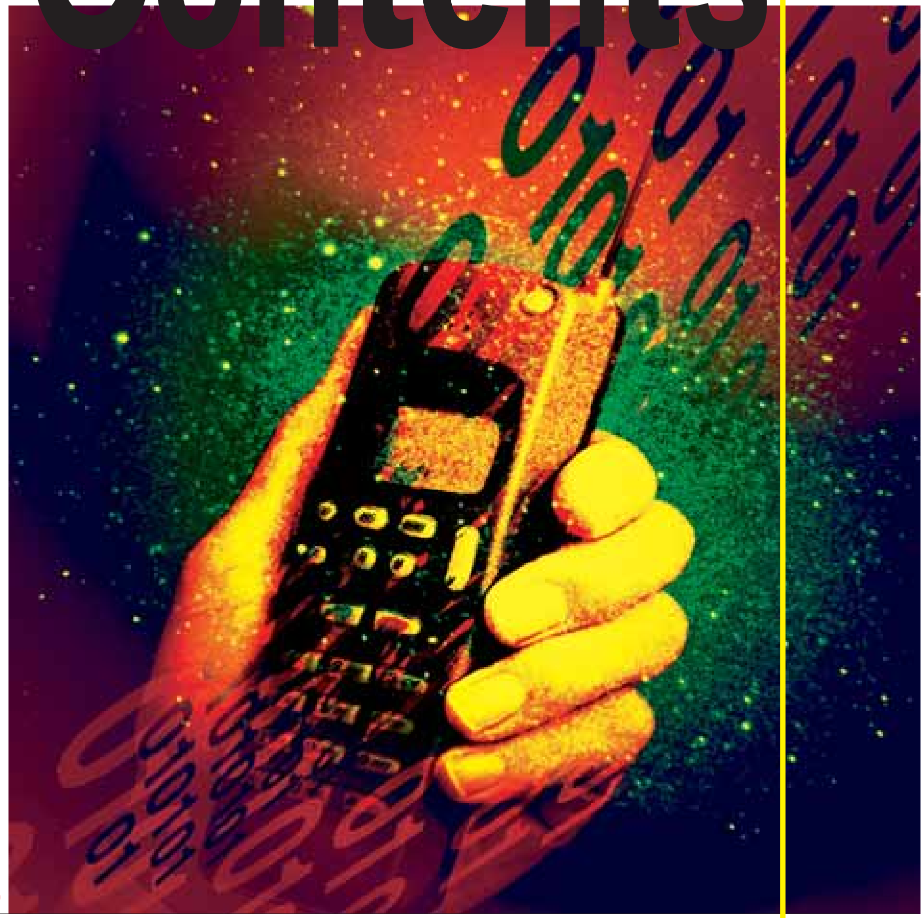

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10. San Jose, CA

QUOTE OF THE MONTH:

> “If a wireless solution is doing its job well, no one should know it’s there. For the mobile worker, this means real-time, continuous access to the right data and applications whenever and wherever they need it. For IT professionals, this means a mobile computing solution that offers users what they need and want. It also means a solution that has the least impact on IT resources in regards to deployment, administration, and management.”
>
> *– Andy Willett (page 98)*

WHAT’S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that’s happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With more than 5.9 million unique page views per month, translating into over 700,000 visitors, TMCnet.com is where you need to be if you want to know what’s happening in the world of VoIP.

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

Telecom Revenue to Grow an Average 9 Percent Annually Until 2009
The U.S. telecommunications industry saw an 8.9 percent increase in overall growth last year and will continue to grow an average of 9 percent annually through 2009, according to a report from the Telecommunications Industry Association. http://tmcnet.com/254.1

Customers Tell Call Center Companies: Fix the Core Issues
British consumers appear to rate the performance of call centers, at best, as “modest.” This was the finding of a spot survey of more than 300 customers in Greater London done during the month of December 2005. All the respondents had used the service of the call center they rated during a week before the interview date. http://tmcnet.com/255.1

Is Skype a wiretap killer?
Even as the U.S. government is embroiled in a debate over the legality of wiretapping, the fastest-growing technology for Internet calls appears to have the potential to make eavesdropping a thing of the past. http://tmcnet.com/256.1

U.S. tech giants defend actions in helping run China’s Internet
Four U.S. technology giants are defending themselves against charges they collaborated with China to crush dissent in return for access to a booming Internet market. http://tmcnet.com/257.1

Education Olympics
According to Cisco CEO John Chambers, America has fallen behind in math and science. Here is a direct quote from an article appearing on CNET: Education and research. It is no longer accurate to say America is falling behind on education. We have fallen behind. We are slipping on nearly every international metric on math and science. Congress must ensure that funding for K-12 math and science education is a national priority. http://tmcnet.com/258.1

TMC’s SIP Channel
The SIP Channel on TMCnet.com features the latest news, as well as original articles related to what many believe will be the cornerstone of our communications future. To visit TMCnet.com’s SIP channel, just point your browser to http://www.tmcnet.com/channels/sip/. Sponsored by AGN Networks.

TMC’s Triple Play Channel
The Triple Play Channel on TMCnet.com features the latest news, articles, and case studies in the booming Triple Play space. To visit TMCnet.com’s voice channel just point your browser to: http:// www.tmcnet.com/channels/triple-play/. Sponsored by NetCentrex.

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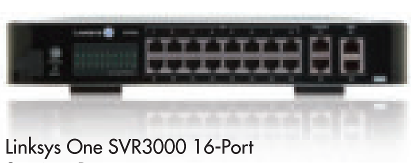

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CISCO SYSTEMS

By Rich Tehrani

What the Vonage IPO Means to VoIP

If you've been following *INTERNET TELEPHONY* for a while, you have probably read how challenging it was to start this publication. The challenge wasn't financial, mind you, but psychological. In 1997 when I attended Comdex and told many of the exhibitors about the new magazine title *INTERNET TELEPHONY*, I was more or less ridiculed. Of course, I believed in my heart that IP telephony was the future of telecom. It just seemed obvious to me at the time. Others, of course, couldn't shake the ham radio perception of the technology.

I have learned since that just because something will obviously happen, it doesn't mean it will happen the way you think it will or when you think it will. VoIP (define - news - alert) has absolutely revolutionized telecom and at the same time took a tired and old industry with small amounts of innovation and turned it upside down and inside out.

For example, just look at how Skype — (news - alert)a company that gives away free software — has become a telephone company overnight. Imagine having tens of millions of users so quickly and with miniscule acquisition cost per subscriber. Skype is a phone company. In 1997, if I had even suggested such a thing was possible, I would have been laughed out of Comdex for sure. And yet, it happened. More incredibly, other companies, such as VocalTec (news - alert), Dialpad (news - alert)and Microsoft (quote - news - alert), all had a business model similar enough to Skype that they could have been Skype. Just as many companies were dancing around Google's business model before they came onto the scene, Skype really showed us how you become a phone company serving tens of millions with not billions of investment, but instead, mere millions.

And yet, communications and VoIP, in particular, has not led the industry up a smooth and steady path. We are now seeing the third ramp in interest in IP telephony. This time, however, companies like Skype have sold for over two billion dollars and there is so much real purchasing taking place in the VoIP market that our future looks brighter than ever.

Perhaps having lived through "the bubble" has made us more sober. The post-bubble mentality implores us to look at things more carefully and to analyze them: Not just the positives but the negatives as well. Perhaps with some analysis we can guide the VoIP market ever upward ensuring the technology continues to help billions of people and make corporations more productive. And of course in the process, help lots of companies make money.

Vonage will continue to focus on market share first at the expense of profits.

Vonage — (news - alert)God bless them and Jeffrey Citron, in particular — was responsible for educating the market about VoIP and what it can do. They have managed one of the best marketing campaigns of any company in recent memory and, in so doing, they woke up business customers and their competition as well. Vonage, however, has been spending on a pace not seen for a startup since Pets.com. Many bloggers and people at industry conferences have told me Vonage is spending too much money and can't have a sustainable business model unless they change their strategy.

Others told me that that Vonage was going through customers at a rapid rate and their churn was in double digits. Negative rumor after rumor circulated throughout the Internet as well.

But recently Vonage put an end to many of the rumors by filing to go public. The company is growing so rapidly and has taken in so much capital that the success (or failure) of its IPO is important for the entire industry.

In reality, it shouldn't be this way, as Vonage is serving but a single sector of the market — consumers. The service provider and enterprise markets will spend far more on VoIP technology and services in the short term. Still, Vonage has attracted lots of attention and, indeed, its performance will affect the industry.

Here is an overview and some analysis on the S-1 documents filed by the company when announcing they will go public. I have done my best to not bore you here but

rather instruct you on the potential of Vonage to do well and the hurdles the company faces.

Vonage has 95% of their lines in the U.S., leaving tremendous growth opportunities overseas. If you have been waiting to hear how much Vonage spends on marketing, here is that number: $232.4 million was spent in 2004 and the first nine months of 2005 combined.

Of course, this spending ate into profits and revenues — although growing spectacularly — cannot mask the fact that losses are growing as well. Revenues were $18.7 million in 2003, $79.7 million in 2004, and $174.0 million for the nine months ended September 30, 2005. From the period of inception through September 30, 2005, the cumulative net loss was $310.0 million. The company's net loss for the nine months ended September 30, 2005 was $189.6 million. During the same nine-month period, marketing expenses were $176.3 million.

Vonage goes on to talk about strengths such as a number one market position in broadband telephony. They even point to market research stating that Vonage is synonymous with the word VoIP in the minds of many consumers.

Another important point the company makes is that their churn has only been 2.11 percent for the first nine months of 2005. More amazingly, 13 percent of overall net subscriber line additions resulted from customer referrals!

Future Strategy

In order to grow, the company will roll out new services and work with chip and equipment providers to come out with unique products and services. In addition, the company will improve customer service, expand distribution, and enter new geographic markets.

Risk Factors

Vonage points out that it will continue to focus on market share first at the expense of profits. This could continue racking up losses for the foreseeable future. Other risks are from competitors that could come up with dual-mode phones using VoIP and, thus, make Vonage unnecessary.

Still other risks come from competing with cable companies and LECs that can bundle services and even offer telephony at a loss. In addition, these other companies might offer superior service levels or some other features that Vonage does not offer.

There also is competition from Internet companies such as AOL, Google, and Yahoo!, who, as the company says, could push prices lower.

The rest of the VoIP industry will be judged by what Vonage does or doesn't do.

E-911

There is also talk of potential problems with E-911 service and how the company may not be able to deliver emergency calls for some reason. They may also fail to deliver 911 calls to the right PSAP in the case of someone calling from a portable client such as a softphone or WiFi telephony device. In addition they may not be able to comply with the FCC and will subsequently be fined or potentially not be allowed to offer service.

Here is a paragraph on this topic lifted directly from the document itself:

> *Our ability to provide our service is dependent upon third-party facilities and equipment, the failure of which could cause delays or interruptions of our service, damage our reputation, cause us to lose customers and limit our growth.*
>
> *Our success depends on our ability to provide quality and reliable service, which is in part dependent upon the proper functioning of facilities and equipment owned and operated by third parties and is, therefore, beyond our control. Unlike traditional wireline telephone service or wireless service, our service requires our customers to have an operative broadband Internet connection and an electrical power supply, which are provided by the customer's Internet service provider and electric utility company, respectively, and not by us. The quality of some broadband Internet connections may be too poor for customers to use our services properly. In addition, if there is any interruption to a customer's broadband Internet service or electrical power supply, that customer will be unable to make or receive calls, including emergency calls, using our service. We also outsource several of our network functions to third-party providers. For example, we outsource the maintenance of our regional data connection points, which are the facilities at which our network interconnects with the public switched telephone network. If our third-party service providers fail to maintain these facilities properly, or fail to respond quickly to problems, our customers may experience service interruptions. Our customers have experienced such interruptions in the past and will experience interruptions in the future. In addition, our new E-911 service is currently dependent upon several third-party providers. Interruptions in service from these vendors could cause failures in our customers' access to E-911 services. Interruptions in our service caused by third-party facilities have in the past caused and may in the future cause us to lose customers, or cause us to offer substantial customer credits, which could adversely affect our revenue and profitability. If interruptions adversely affect the perceived reliability of our service, we may have difficulty attracting new customers and our brand, reputation and growth will be negatively impacted.*
>
> *Local number portability too was cited as a potential problem as porting a customer's telephone number can take up to 20 days or more. Other potential challenges are having access to more capital or having access to capital at favorable rates.*

In addition, the company cites the potential of regulation as a risk to the company, as they may be forced to leave markets or raise prices as a result of new regulations. The company also mentions that it may be forced to contribute to the Universal Service Fund, which, of course, would increase costs.

Where is the Money Going?

As you guessed, the money raised from this IPO will be used for marketing and further expansion. The company mentions it may use some of the proceeds for an acquisition as well.

Other notable facts are that revenue per subscriber has decreased over the past two years due to price decreases and is now $26.63 per month.

The Vonage IPO was thought to be a multi-billion dollar event and it turns out now that it will only be worth $250 millions dollars. The reason for this lower amount is primarily due to the market share at all costs attitude. I am concerned that this attitude will generate many impatient investors who will push the stock to very low levels. Of course, if Vonage can't pull off financing their operations for some reason, they can simply cut spending on marketing and make instant profit. On the other hand, if they can get the financing they need, at some point they could become a worldwide telecom giant.

Still, I grow concerned that the rest of the VoIP industry will be judged by what Vonage does or doesn't do. The company certainly represents a sector of the VoIP market — but not all of it. Furthermore their business model sucks up much more money than that of, say Packet8, who is forming partnerships with companies around the world and allowing others to share marketing costs as well as rewards.

For those of you that remember when the VoIP industry was full of doom and gloom (circa summer of 2003), who would have thought then that we would hear the words VoIP IPO again? Who would have thought that the business would be where it is today? These are indeed exciting times and I wish Vonage a spectacular IPO that exceeds expectations. IT

Multi-core Processing to Dominate Embedded Devices in 12-18 Months

News Analysis By Robert Liu
TMCnet Wireless and Technology Columnist

The challenges the firewall poses to voice over IP have been well documented. Those challenges of routing VoIP (define - news - alert) traffic to traverse the network address translation (NAT) firewall securely, yet quickly, are greatly magnified when scaled out to the carrier-class or Central Office level. A number of software-based solutions, like Interactive Connectivity Establishment (ICE) or Simple Traversal of UDP over NATs (STUN), have been developed to address the issue. On the hardware side, session border controllers (SBCs) certainly help with scalability. But very little has been done outside of the general purpose computing environment — until now.

With the advent of dual-core and 64-bit technologies, the brains on an integrated circuit board that processes thread after thread of control or data commands are also multiplying at a phenomenal rate. Within the next year or so, you will likely see multi-core becoming the dominant platform in the embedded devices market.

"It's going to start coming on strong in the next 12 to 18 months," said Romain Saha, Netcom (news - alert) Segment Manager for QNX Software Systems, a Canadian subsidiary of consumer electronics giant Harman International. Multi-core processing "is well-suited for embedded applications so you will see that driving adoption."

QNX (often pronounced "Q-Nix") makes the real-time operating system that runs inside of the CPU, whether that's on an x86-, PowerPC- or ARM-based processor. By March, QNX will be making its QNX Momentics development suite Multi-Core Edition, first announced last October, available to its customers to port legacy embedded designs or build new ones that can take advantage of multiple threads. With the release of the technology developer kit, the new Multi-Core Edition of the OS will be able to harness any variation of multi-core architecture, whether Symmetric or Asymmetric.

"This is a version of our OS targeted at helping OEMs [original equipment manufacturers] to make the migration from the uni-processor world to the multi-processor world," said Sebastien Marineau-Mes, Chief OS Architect for QNX Software Systems.

While a large portion of the embedded market targets applications specifically used for the military, automotive and industrial sectors, the networking realm could also greatly benefit from the increasing presence of multi-core technologies. Take, for example, Cisco Systems. At the heart of the networking giant's CRS-1 Carrier Routing System is the Cisco IOS XR software, a self-healing, self-defending operating system based on QNX.

Based on a configuration that includes an ASIC-based solution handling data classification, parsing and other data plane management, symmetric multiprocessing architecture could increase the network control management to yield performance gains that close in on the 2x threshold, according to Toby Foster, a product marketing manager at Freescale Semiconductor.

Foster, however, cautions that many factors influence the performance gain that is seen when moving from a single core solution to a dual core symmetric multiprocessor solution. The inherent parallelism in the application itself is the most important factor, and the contention for shared resources can be a factor as well. Still, performance gains are undeniable when deploying symmetric multiprocessing because more processing resources are applied to solve the problem.

For its part, Freescale argues that it has marketed asymmetric multiprocessing in one form or another for more than 10 years now. With its line of PowerQUICC communication processors, Freescale has shipped more than 200 million units to more than 500 customers, said Mike Shoemake, Business Manager of the Digital Systems Division at Freescale Semiconductor.

Since its inception, the line has been designed with a single-core PowerPC processor and a second 32-bit RISC-based communications processor (CPM) to handle data and control management. In the past, Freescale has designed the CPM component, also known as its "QUICC Engine," with dual- or quad-cores. But with the production of its MPC8641D last October, Freescale now offers a dual-core PowerPC to complement its QUICC Engine.

Folks like QNX's Marineau-Mes believe that trend will continue for the foreseeable future. "It's really a paradigm shift. A challenge in the industry is either you adapt to the new world or your competitors will. It's not a question of if, but when and who will be able to adapt and who will not," the Chief OS Architect explained. IT

Robert Liu is Executive Editor at TMCnet. Previously, he was Executive Editor at Jupitermedia and has also written for CNN, A&E, Dow Jones and Bloomberg. For more articles, please visit Robert Liu's columnist page.

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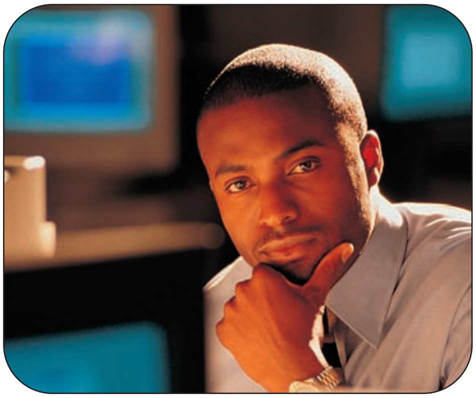

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Intel & IMS: Driving a Vision of Modularity

By Greg Galitzine
Internet Telephony Magazine

IP Multimedia Subsystem (IMS) is a standardized next-generation networking architecture designed to allow the rapid creation and deployment of IP-based multimedia services across disparate networks (wireline, wireless, cable) for a seamless user experience.

As carriers look to deploy IMS solutions, many network equipment manufacturers are looking toward standard-based platforms, such as AdvancedTCA®-based chassis. AdvancedTCA (Advanced Telecom Computing Architecture) is an industry standard for designing next generation carrier grade telecom equipment. AdvancedTCA incorporates the latest high-speed interconnect technologies and next generation processors for improved reliability, manageability, and serviceability, resulting in a new blade (board) and chassis (shelf) form factor optimized for communications — in essence, the next generation of building blocks.

It is this building block approach that is driving Intel's (quote - news - alert) vision of modularity for telecom equipment manufacturers. What modularity does, according to this vision, is simply this: It enables quick deployment, easier scalability, reusability, investment protection… in short, everything equipment manufacturers need to be successful in today's fast-changing market.

Driving this vision to become reality, Intel® announced two new offerings at the recent 3GSM World Congress 2006 in Barcelona, Spain this past February.

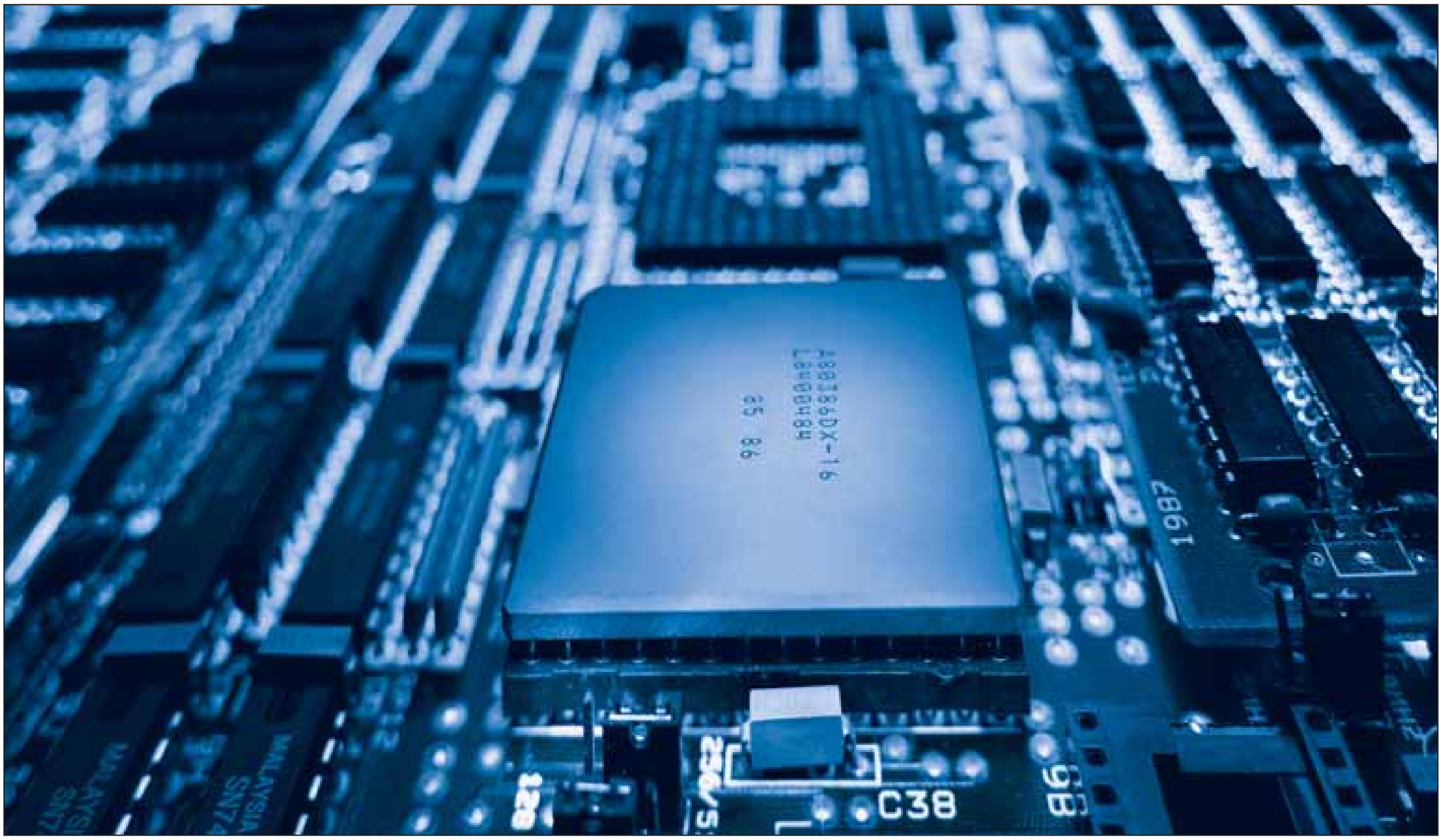
Dual-Core Intel® Xeon® Processor LV 2.0 GHz

First, Intel unveiled their first dual-core processor designed for communications applications. The Dual-Core Intel Xeon Processor LV 2.0 GHz combines the benefits of dual core with dual processor capabilities providing four high-performance cores per platform. These dual core/dual processor capabilities enable a significant performance and performance per watt increase over previous single core processors, yielding up to a 4x improvement in performance per watt. Using this new processor can provide benefits for a wide array of communications and embedded applications such as storage area networks (SAN), network attached storage (NAS), routers, virtual private networks (VPN), firewalls, intrusion detection systems, and telecommunications (wireless and wireline) servers, particularly in AdvancedTCA form-factor designs.

Some of the features of this new offering include:

- Two complete execution cores in one processor package provide advancements in simultaneous computing such as multi-threaded applications and multi-tasking environments. Dual-core processing efficiently delivers performance while balancing power requirements.

- High-performance front-side bus (FSB) provides dual processor support for



demanding high-performance, volume applications. Combined with dual-core processing, this supports up to four simultaneous threads on the system.

- Enhanced Intel SpeedStep® technology allows a system to dynamically adjust processor voltage and core frequency, decreasing average power consumption and average heat production.

- Streaming SIMD Extensions 3 (SSE3) provides significant performance enhancement for multimedia applications. Additional instructions designed to improve thread synchronization, complex arithmetic, graphics, and video encoding.

Intel NetStructure® MPCBL0040 Single Board Computer

The second announcement at 3GSM centered on a newly released single board computer — the Intel NetStructure MPCBL0040, which is based on the aforementioned Dual-Core technology. The MPCBL0040 is the latest in a series of high-compute boards intended to interoperate with AdvancedTCA equipment from Intel. This new board features two Dual-Core Intel Xeon Processors LV 2.0 GHz, which are designed to increase performance, offering greater processing power for applications such as IMS, IPTV, and more.

So what, then, is the benefit of this new single board computer? According to Anthony Ambrose, General Manager, Modular Communications Platform Division at Intel, "With more than twice the performance of previous AdvancedTCA single board computers from Intel, far more transactions and subscribers can be serviced per system."

"Multiple network elements can be integrated in a single chassis — moving one step closer to 'network in a box' technology for improved network modularity — or existing system capacity can be scaled to support a much larger number of transactions and subscribers," Ambrose continued. "In either case, cost per unit and total cost of ownership are both significantly reduced."

Some features of the MPCBL0040 include:

- Dual processor support, including an enhanced bus arbitration protocol, power-optimized 667 MHz front-side bus (FSB), and a 2 MB shared L2 cache per processor enabling up to four high-performance cores per platform.
- 36-bit memory addressing using Physical Address Extension (PAE) that provides support for more than 4 GB of memory.

- FSB address, data parity, and an enhanced error reporting mechanism through Machine Check Architecture (MCA) that ensures reliability and data integrity.

MCP Solution Center

To validate their vision of modularity Intel has deployed several Modular Communications Platforms (MCP) Solution Centers globally, with locations in Parsippany, NJ, USA, Beijing, China, and Kontich, Belgium. These testing facilities were designed to validate the interoperability of Intel communications building blocks, including the new Dual-Core Intel Xeon Processors LV 2.0 GHz and the Intel NetStructure MPCBL0040 single board computer with products from third parties. Just as critical, the MCP Solution Centers enable members of the Intel Communications Alliance to test their hardware and software solutions and see how they interoperate with technology from Intel.

> "With more than twice the performance of previous AdvancedTCA single board computers from Intel, far more transactions and subscribers can be serviced per system."

To fully establish modular solutions built from Intel components it is necessary to validate their capabilities. The Modular Solutions Platform Group (MSP) at Intel will provide an environment where ecosystem members can integrate, validate, and promote their solutions running on the Intel Telco Server Platform.

Today, solution architects and software/hardware engineers located at these MCP Solution Centers are hard at work testing and validating solutions that will be deployed under the Intel IMS solutions umbrella. The labs are equipped with a wide range of telco-grade server platforms and commercial telecom software, including the latest ATCA blade and rack mount servers.

The results of interoperability testing between Intel and Intel Communications Alliance members were on display in the form of live demos at 3GSM as well.

Fujitsu Siemens

Fujitsu Siemens showcased a Tele-Voting application based on AdvancedTCA. This AdvancedTCA platform simulates network traffic coming in from users that are voting over the telephone (wireless or wireline). This demonstration presented a variety of building blocks from the communications ecosystem, using high-availability, carrier-grade Linux and middleware.

Hewlett Packard

In a live demo titled *HP IMS with Handhelds on AdvancedTCA*, attendees were able to see an IMS framework based on AdvancedTCA platforms enabling robust interoperable gaming, voice, and data services. This IMS-ready solution uses OCMP video and BEA Systems SIP Application Server, allowing multiple SIP-based endpoints, like a PDA-based SIP Client, to be connected via audio and video communication. Data aware clients can also do data collaboration work such as document sharing, joint Web browsing and instant messaging.

Conclusion

Intel continues to make strides in IMS. By offering cutting-edge platforms based on multi-core technology and AdvancedTCA, Intel stays at the forefront of the industry when it comes to developing standards-based platforms for vendors to deploy their advanced IMS applications on. And it all comes back to their vision of modularity. By developing applications using modular building blocks, such as those provided by Intel, equipment manufacturers and carriers alike can realize the full benefits of IMS, specifically, the rapid creation and deployment of next-generation, revenue-generating services. IT

Greg Galitzine is editorial director of Internet Telephony magazine and the newly launched IMS Magazine.

The following companies are sponsors of the ipcommunications.com website and are part of the Intel® Communications Alliance, a community of communications and embedded developers and solutions providers committed to the development of modular, standards-based solutions on Intel technologies.

Site Sponsors — March 2006

Industry NEWS

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Any last regrets?
VOICE CARRIER PRESIDENT
Yes, I should have used SIPTalk-Pro from Global Touch

TalkSwitch Adds the TS-600 to its Line of Desktop Telephone Sets

TalkSwitch, (news - alert) a company specializing in the design and manufacture of innovative telephone systems for small and multi-location businesses today announced the release of the TS-600, its premium desktop telephone set for the North American market.

The TS-600 is a stylish single-line analog telephone set that incorporates a rich mix of telephony features optimized for integration with TalkSwitch telephone systems. It features an easy-to-read 7.4 square inch LCD display, a superior business quality speakerphone and single-cord connectivity meaning the phone can be powered through the phone cable.

"The TS-600 is our most sophisticated telephone set," said Jan Scheeren, President and CEO, TalkSwitch. "It's the ideal set for small business owners looking to add a little extra functionality and style to their desktops. Like all our phones, the TS-600 is a perfect companion to the TalkSwitch system. Its upscale styling and rich feature set make it the ideal addition to our product line."

http://www.talkswitch.com

Netcentrex Extends IVR Features to Video with MyCall Media

By Erik Linask, Associate Editor

Netcentrex, (news - alert) which develops next generation network (NGN) voice and video solutions to enable delivery of converged voice-video-data and fixed-mobile services, introduced MyCall Media, an interactive video solution for 3G mobile and broadband IP users that will enable video telephony, video conferencing, video blogs, remote monitoring, and entertainment applications.

Interactive voice response (IVR) has long been used by operators. Interactive video applies the same key features that have made IVR successful — short numbers, self-service approach, telephony billing — and adds benefits such as user-friendly video menus, real-time person-to-person live video connectivity and content delivery.

MyCall is designed to create and run high-margin, interactive video services with lowered time-to-market and simplified customer adoption. The solution includes application development modules and is based on a carrier-grade media platform used by more than 50 operators worldwide.

"MyCall Media is a complete solution that enables developers and service providers to create these interactive video services. It includes application building blocks and development assistance to ensure quality and fast time to market," said Erik Larsson, VP of Marketing at Netcentrex.

http://www.netcentrex.net

Juniper Intros Secure Services Gateway 500 Series

By Johanne Torres

Juniper Networks Inc. (news - alert) introduced the Secure Services Gateway (SSG) 500 Series on Monday. SSG is a new line of high-performance firewall/virtual private networking (VPN) platforms with integrated local-area network (LAN) and wide area network (WAN) interfaces. The new Secure Services Gateway 520 and 550 platforms will enable enterprises to secure branch and medium-sized offices.

SSG is based on Juniper's ScreenOS operating system. The platform delivers up to one gigabit per second (Gbps) firewall and 500 megabits per second (Mbps) VPN throughput, and it provides optional intrusion prevention, Web-based filtering, antivirus, and antispam capabilities to protect customers from internal LAN and external WAN-based security threats. The SSG provide concurrent firewall/VPN and content security processing for broadband-connected branch offices. The system's security zones also allow customers to divide their wired and wireless networks into secure segments, each with their own security policy, allowing for better support of various user groups that could include employees, contractors and customers.

"Broadband connected branch offices require high-performance and advanced security processing," said Hitesh Sheth, vice president of security products for Juniper Networks. "Juniper Networks is continuing to answer the needs of our customers with the new Secure Services Gateway product line, which provides best-in-class security and connectivity options for branch, regional and medium-sized offices."

http://www.juniper.net

Panasonic Has 40 IPv6-Compatible Office Products

By Patrick Barnard

Even though IPv6 is still being developed, Panasonic (news - alert) already has more than 40 IPv6-compatible products available. The company said it is making its office devices IPv6 capable ahead of time because it wants to stay ahead the game — but also because of the types of business customers it serves.

"One of the reasons why Panasonic got a jump on IPv6 is that it markets business products used in environments requiring the highest level of innovation and security, from MFPs in the White House to notebook computers used by U.S. military personnel in combat," the company said.

Among the IPv6-ready products Panasonic demonstrated were its C3 color MFP series and DP-CL22 color laser printer.

"The concept behind the secure IPv6 networked office is to allow both our communication and imaging office automation products to communicate over the same physical network," Hogan told the audience. "The technology strategy is directed to specific product targets like IP-PBX, web meeting systems, secure plain-paper copiers or multifunction devices, using common technology platforms. In all cases, IPv6 is central to the strategy."

http://www.panasonic.com

8x8 Rolls out New Features and VoIP Phone Adapter

By Johanne Torres

VoIP and videophone service provider 8x8 Inc. (news - alert) unveiled new calling features for its Packet8 voice and video residential phone service and the Packet8 BPG510, a new terminal adapter with an integrated router.

The new features include voicemail to e-mail notification, find me, follow me, simultaneous singing on multiple phone numbers, 7-digit local dialing, network unavailable forwarding and call waiting disable.

8x8 also announced the release of the Packet8 BPG510, a new broadband phone adapter that eliminates the need for a separate router in a subscriber's home network. The BPG510 is equipped with two Ethernet ports (LAN and WAN) and one RJ-11 phone port for connecting a standard analog telephone. The built-in router prioritizes voice packets over data and manages available bandwidth. The LAN Ethernet port can be used to connect a PC or other networked device without adding any additional equipment.

http://www.packet8.com

Auvi Adapter Transforms Analog Phones Into VoIP Receivers

By Mae Kowalke

Consumers who want to use their existing traditional or cordless telephone to receive calls using their VoIP service now have a new option — the WIP20 adapter, made by Auvi Technologies. (news - alert)

The WIP20 connects to the consumer's VoIP service using the audio jacks on any Windows XP desktop computer. An included setup wizard guides users through the installation process.

The WIP20 is 4 inches by 2 inches in size, and comes in two versions: WIP20S for Skype subscribers, and WIP20K for SIPTalk subscribers. It works with both broadband and dial-up Internet connections.

"The WIP20 enables users to make any telephone VoIP-enabled with as little hardware investment as possible," said Auvi CEO Santosh Patel, in a press release.

Auvi's adapter comes packed with two 3.5mm-to-3.5mm line-in cords, an AC adapter, software CD, and quick-start guide.

http://www.auviworld.com

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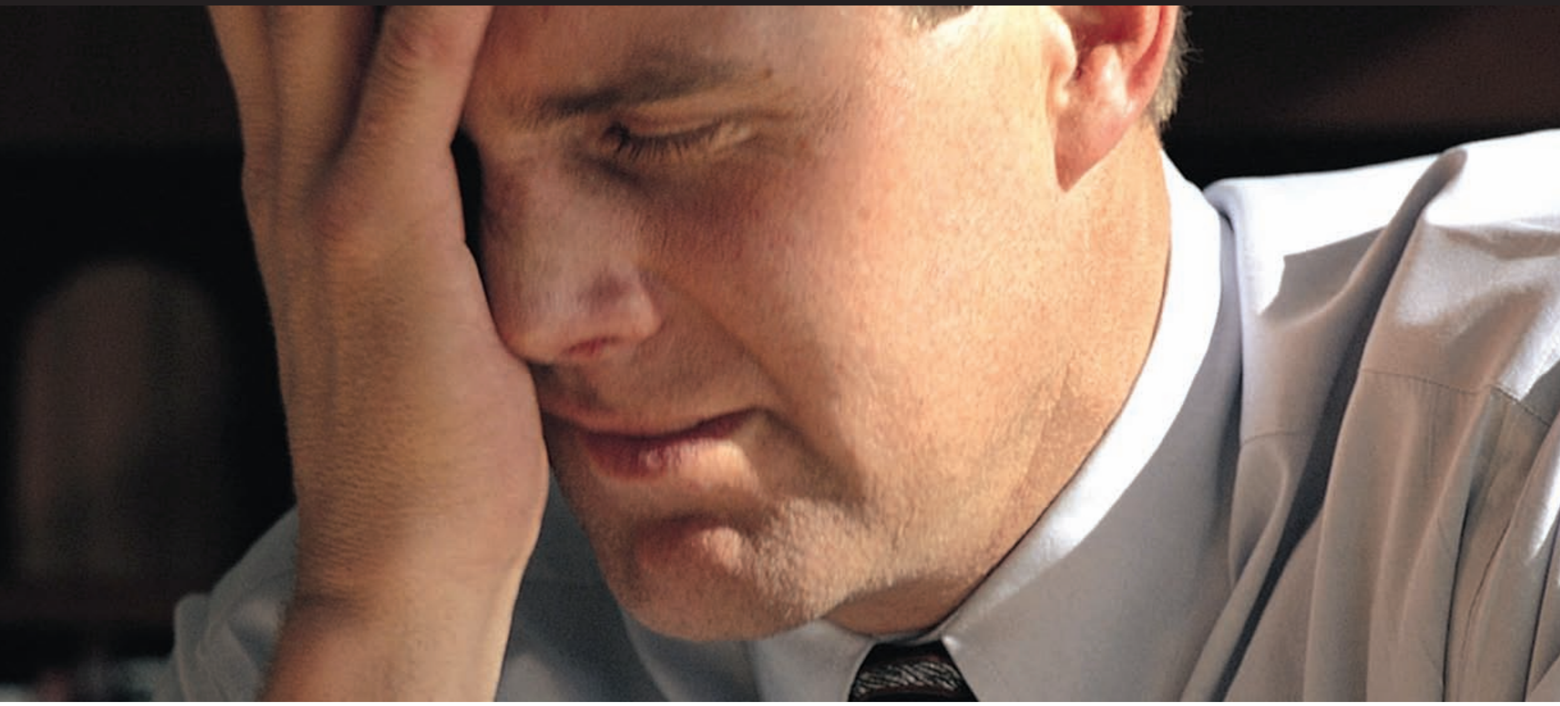
Don't Make a Million Dollar Voice Messaging Mistake



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

Why?

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

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

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

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

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

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

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

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

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

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

CS

COMMUNITECH SERVICES, INC.

www.communitechservices.com

Vodavi's New Telenium IP System Delivers Cutting-Edge Mobility

Vodavi Technology, Inc., (news - alert) a provider of traditional and next generation business telecommunications solutions, announced the availability of the third generation of software for its TeleniumIP communication system for business customers requiring increased mobile communications that network multiple offices, home offices and remote buildings together to create a cohesive communications solution.

Enhanced capabilities available on the TeleniumIP include a new family of IP desktop telephones and Soft Phone IP Endpoints that improve mobility and deliver voice and video capability on a laptop PC or Personal Digital Assistant (PDA). Along with the new generation of IP Endpoints supported with Generation III software, enhanced calling features, integrated Automatic Call Distribution (ACD) and enhanced system administration features are included. A wireless IP (Wi-Fi) handset that operates via wireless access points will be commercially available during the first half of 2006.

Available to suit the communication needs of businesses ranging in size from 10 to 250 station users, the TeleniumIP is based on distributed architecture that is completely LAN-based with Web-based system administration and maintenance. Relocating phones and handling moves, adds and changes are easy and can be handled by internal support staff.

http://www.vodavi.com

Intrado and XTEND Tackle Enterprise E911 Market

By Johanne Torres

VoIP E911 provider Intrado Inc. (news - alert) announced an agreement with XTEND Communications Corp., (news - alert) a provider of Computer Telephony Integration (CTI) and Public Safety Answering Point (PSAP) systems, to market VoIP E911 services. Under terms of the agreement, XTEND Communications will market Intrado V9-1-1 for Enterprise to large businesses that operate and support their own Internet Protocol Private Branch Exchange (IP-PBX) based networks. Intrado had just launched the enterprise market version of its popular V9-1-1 service yesterday.

With the integration of Intrado V9-1-1 for Enterprise and XTEND's Enterprise Alert system, critical E911 support will be available to remote employees who use a softphone to access their company's private network, remote employees who use company assigned IP phone equipment, or others who are networked to their employer's main site.

"Until recently, we were able to provide only a limited level of 9-1-1 services that are required for complex enterprise IP-PBX environments," said Bill Schwartz, XTEND's president. "By teaming up with Intrado, we can deploy a superior 9-1-1 solution to our customers, enabling them to leverage the full value of their IP-based equipment."

http://www.intrado.com
http://www.xtend.com

Aculab Maintains Prosody Media Processing Product Alignment with the Envox 6.2

Aculab (news - alert) has announced the integration of its range of Prosody media processing resources with Envox 6.2, (news - alert) the latest version of Envox Worldwide's award-winning communications development platform. Envox 6.2 reduces the time, cost, and complexity of developing and deploying voice solutions.

Aculab's latest telephony driver enables Envox 6.2 users to immediately integrate Prosody PCI and CompactPCI hardware variants. Prosody S is also available as a media processing resource for use with Envox 6.2. The Aculab telephony driver implements Envox's telephony driver API and provides access to Prosody resources including Group 3 fax and multi-party conferencing. Prosody can also be used with any Envox-supported text-to-speech (TTS) and automated speech recognition (ASR) engine.

Envox 6.2 is an open, standards-based communications development platform with integrated application development and management components that reduce the time, cost and complexity of creating voice solutions. Envox 6 allows customers to continue to leverage their investments in legacy hardware, software and solution development, while providing a smooth migration path to new standards and emerging technology, including speech, VoiceXML, VoIP and Web services.

http://www.envox.com
http://www.aculab.com

NEC Unified Delivers Enhanced IP to Small and Mid-sized Businesses

NEC Unified Solutions, Inc. (NEC), (news - alert) a leader in converged voice and data communications, announced the availability of its Electra Elite IPK II, NEC's new converged IP communications platform for the SMB market.

The IPK II is a cost-effective solution that is designed to help customers adopt next-generation IP communications, as their business demands it. The IPK II is an attractive solution for small businesses looking for the functionality of a traditional key system, as well as for medium-sized organizations with hybrid IP/PBX operations or remote users.

"The market, in general, agrees that small to medium businesses are leading the way in IP telephony adoption," said Rob Arnold, enterprise telephony analyst, Current Analysis. "With the IPK II, NEC is at the forefront of this movement by offering enterprise functionality at a lower cost for a product line that has already been selling well."

The IPK II mirrors NEC's strategy to facilitate easy adoption and migration of new technologies without requiring a total system replacement. It can be easily updated through remote software upgrades and integrates with NEC's complete product line to support higher port capacities. By networking between NEC's Electra Elite IPK, IPK II and its UNIVERGE NEAX product lines, users at remote sites can enjoy all of the benefits of the main telephone system and features.

http://www.necunified.com

Lucent Technologies Enhances Vivo's 3G CDMA2000 1xEV-DO Network

Lucent Technologies (quote - news - alert) announced a contract with VIVO, (news - alert) to expand and enhance the operator's third-generation 3G CDMA2000 1xEV-DO network. Lucent will provide additional infrastructure equipment and services and also work with its suppliers Riverstone Networks and Juniper Networks to deliver a core IP/MPLS network that will enable VIVO to efficiently manage large increases in network traffic as it expands its network to serve additional customers in São Paulo and Rio de Janeiro.

"The solution presented by our partners — Lucent, Riverstone and Juniper — will provide us with the ideal features to keep the network's performance highly reliable, thus ensuring the high availability of the advanced services Vivo Play 3G delivers to our subscribers," said Roberto Lima, VIVO's President.

Riverstone Networks is providing the Ethernet infrastructure, supporting the backhaul of data traffic from VIVO's 3G base stations to expand the delivery of IP/Ethernet services over metropolitan access networks

Juniper Networks is supplying M-series multiservice routing platforms, which will aggregate traffic from multiple base stations, combining IP/MPLS capabilities with unmatched reliability, performance, and scalability.

http://www.lucent.com

http://www.vivo.com

West to Purchase Intrado for $465 million

By Stefania Viscusi

West Corporation, (news - alert) a provider of communications solutions announced it has agreed to purchase Intrado Inc., (news - alert) provider of integrated data and telecommunications solutions, for $465 million.

West Corporation provides communications solutions, such as effective customer communications and relations, as well as tactics for driving greater revenue, to a number of the world's largest companies.

Providing the basis of safety for the nation's 911 networks applications, Intrado's services are relied upon by public safety and government organizations.

George Heinrichs, CEO of Intrado comments, "Being part of West Corporation represents the best future for all Intrado stakeholders — shareholders, customers, employees, partners, and suppliers — the general public — which our business ultimately serves every day — and the communities within which we operate. We anticipate a smooth integration process once the transaction is finalized."

"Its strong position in a growing market, its industry-leading technology, and its experienced management team will improve our ability to meet our customer's demands. Further, Intrado complements the existing offerings of our Communications Services segment, providing a highly visible revenue stream and additional cross-selling and margin expansion opportunities," said Thomas B. Barker, CEO of West Corporation .

http://www.west.com

http://www.intrado.com

RUN WITH IT

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

James Ballard Smith
Chief Operating Officer
InZon Corporation

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

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

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

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

"Recommend Emergent to others? I do that all the time."

Emergent offers solutions for what you need today, and where you're going tomorrow.

Build on it, grow with it, run with it.

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Sawtel To Deploy Meru's WLAN System for World's Largest Virtual Cell

By Erik Linask

Meru Networks, (news - alert) a leader in wireless VoIP infrastructure, has been chosen by Hartford, CT-based Sawtel Inc., (news - alert) a Satellite and Wireless Telecommunications company, for an island-wide WiFi deployment in the Bahamas. With the Meru solution, Sawtel will offer converged voice, data, and video service to both business and residential users across the island of Nassau. The voice services will give subscribers access to hosted IP PBX, PBX Enhanced Services, and IP Centrex applications.

"We have always believed that wireless technology is the ideal vehicle for delivering cost-effective services with QoS to customers who would otherwise not have access," said Sawtel Inc.'s Chief Technology Officer, Eric Asare. "Meru's WLAN System is the only solution that could deliver toll-quality, carrier-grade voice capabilities and the mobility and RF features required for our deployment."

Sawtel has made a name for itself by providing fast and reliable access to the world at the lowest possible cost via a blend of leading-edge technologies. For the 1000-node, island-wide deployment, Sawtel will use Meru's WLAN System, both outdoors and in, to create the world's largest virtual cell — it will cover 140 square miles.

http://www.sawtel.com
http://www.meru.com

Internet Taxation Battle Will Be Fought Again Soon

By David Sims

The Internet Tax Moratorium expires in 2007, and state, federal and international regulators and legislators are already targeting the Internet as a lush source of new revenue, says a new report released by the Institute for Policy Innovation.

George Pieler, an IPI research (news - alert) fellow claims that, "Absent a sweeping federal intervention to secure the Internet's freedom, it will be an increasingly rich target for revenues and regulatory interference from all directions."

VoIP's experience is not hopeful for the Internet. One indication of states' eagerness to collect Internet taxes, Pieler says, is how quickly they began taxing VoIP: "Because VoIP competes with traditional telecom services, the 2004 moratorium did not consistently block its taxation. If states are so quick to take this tax advantage, what is to stop them from taking even more Internet revenue?"

"Before the Internet Tax Moratorium expires in 2007, Congress and the executive branch should seriously review Internet taxation from the local, state, national and international perspective, and determine how best to sustain the largely tax-free Internet, that has done more good for the world than any bureaucracy ever could," concludes Pieler.

http://www.ipi.org

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XO Upgrades Concentric Hosting Plans

By Johanne Torres

Telecom service provider XO Communications Inc. (news - alert) announced it has upgraded its Concentric hosting plans to provide increased bandwidth, disk space, and e-mail addresses with its Email Hosting, Starter, and Clustered Hosting plans.

In addition to the hosting plan upgrades, Concentric plans also include one domain name registration free of charge for one year with Basic, Professional, and Enterprise plans. XO is also offering Private Domain Registration for customers who register their domain name through Concentric. The cost for Private Domain Registration is $8.95 per year.

"We are committed to continuously improving our service and ensuring that we have a competitive offer for our customers," said Barbara Branaman, vice president and general manager of XO's hosting business unit. "Businesses know Concentric's reputation for reliability, security, and excellent customer support. With these new upgrades, we are quite confident that small and medium sized business will know Concentric continues to offer the best value in web hosting backed by service guarantees critical to operating a business online."'

http://www.concentric.com

PAXIO to Deploy Triple Play Using PacketFront's FTTH Technology

By Patrick Barnard

PAXIO Inc., (news - alert) a registered CLEC providing Internet and VoIP services, has chosen PacketFront's (news - alert) BECS control and provisioning system for the deployment of advanced IP services in new residential developments in California.

PacketFront, which specializes in open access fiber-to-the-home (FTTH) technology and next generation broadband management, will work with PAXIO in offering FTTH to the rapidly growing technology hub of California's Bay Area.

The new network will offer homeowners "unlimited bandwidth" and Internet connection speeds of up to one Gigabit per second. This will allow homeowners to download movies from the Internet; conduct multiple downloads at one time; and use VoIP services. According to the news release, PAXIO will offer its IP services starting at $26.50 per month.

In order to deliver quality next generation services over fiber, PAXIO replaced its existing access layer with PacketFront ASR routers, managed by BECS, to achieve a Layer 3 design. This gives PAXIO control to the network edge, while allowing the rest of its existing network infrastructure to be retained. The new network will enable PAXIO to manage all end users from a centralized location, thus improving customer service and reducing the need for subscriber services.

http://www.paxio.com
http://www.packetfront.com

Verizon Continues to Roll Out FiOS TV

By Patrick Barnard

Verizon (quote - news - alert) announced that its all-fiber FiOS TV service is now available in the Long Island, N.Y., communities of Bethpage, Hicksville, Lynbrook, Malverne, Rockville Centre, Valley Stream and West Hempstead.

FiOS TV is run over Verizon's fiber-to-the-premises (FTTP) network, which the company currently is building in more than half of the states where it offers landline services. FTTP technology utilizes fiber-optic connections, instead of copper wire, and thus provides true "end-to-end" digital connectivity. The all-fiber network will enable Verizon to offer advanced voice, data, and video applications for residential and business customers (including VoIP, IPTV, and Video-on-Demand).

"FiOS is setting the new standard for consumer and small-business broadband services in New York," said Thomas A. Dunne, vice president of Verizon New York/Connecticut. "Customers who already subscribe to FiOS services are astounded at what they now can do with their online experience. Video chats and conferencing, purchased digital movie downloads and interactive multi-player games have become a part of their daily lives."

http://www.verizon.com

NMS Communications and LogicaCMG Launch Hosted Ringback Service

NMS Communications (news - alert) and LogicaCMG (news - alert) announced the launch of the hosted ringback tone service for Finnish telecom operator Elisa, a partner operator of the Vodafone Group.

Elisa's Tunnari ringback tone service is based on NMS's MyCaller personalized mobile entertainment platform and LogicaCMG's global delivery, systems integration and support capabilities. The combined solution by LogicaCMG and NMS is hosted using LogicaCMG's own data centers, enabling Elisa to reduce the cost and time-to-market for ringback tone services, while gaining the ability to quickly scale the service to meet growing demand.

Ringback tones complement other popular mobile applications such as ringtones and wallpapers, which let users personalize their mobile experience. Ringback tone services allow mobile users to replace the usual "ring" that callers normally hear with music and other audio clips chosen by the subscriber. Industry analyst firm Ovum estimates the worldwide market for ringback tones will reach $2.7 billion by 2009.

"Mobile subscribers today are looking for services that are not only entertaining but that let them express their personalities," said Mikko Uosukainen, project manager at Elisa.

http://www.nmscommunications.com
http://www.logiccmg.com

Ascom Launches New VoWiFi Communications

By Johanne Torres

Thanks to a new VoWiFi communications system launched by Ascom Wireless Solutions, (news - alert) staff at hospitals and other healthcare establishments will be able to roam around free of wires and cables associated with legacy telephone equipment that usually strap them to their desk stations.

The new system is based on the IEEE 802.11(b/g) standard and uses VoIP technology and wireless VoIP handsets. The system also offers an integrated personal alarm function, alert messaging, paging, and voice telephony, with access to a range of other communications services. It integrates with existing voice and data networks in the medical facilities and can interconnect with public networks.

"VoIP and WiFi have become standardized and mature technologies, enabling us to offer solutions that meet the healthcare sector's requirements for reliability, robustness, security and flexibility," said Ascom's Chief Technology Officer, Stefan Bramberg.

http://www.ascom.com

Skype, Google, VCs Invest in FON Hotspot Network

By Robert Liu

Skype Technologies (news - alert), Google (news - alert), and two high-profile venture capital firms, Sequoia Capital and Index Ventures, have invested 18 million ($21.5 million) in a tiny Spanish company called FON Technology (news - alert) SL of Madrid with the goal of building up a global network of wireless broadband hotspots.

"FON is trying to make it easy for anyone who has already invested in broadband and a wireless access point to securely share their WiFi, so together we can build a unified global wireless network which is easy to access, safe to share, and reasonably priced," Skype's co-founders Niklas Zennström and Janus Friis said in its main News Web log.

FON's ability to attract big-name investors is attributed to the notably reputation of its founder and CEO Martin Varsavsky. Before starting up FON three months ago, Varsavsky also founded Ya.com, Spain's second largest Internet company, and Jazztel, the second largest publicly traded telecom company in Spain.

http://www.fon.com
http://www.skype.com
http://www.google.com

Woize Brings its VoIP Service to Handhelds Running Windows Mobile 5.0

By Patrick Barnard

VoIP service provider Woize International Ltd. (news - alert) announced the availability of Woize for Smartphone, or SmartphoneWoize, an added feature to the Woize portfolio which brings the company's proprietary VoIP service to mobile phones.

SmartphoneWoize allows users to make free calls to other Woize, PocketWoize, or SmartphoneWoize users, as well as calls billed per second to users outside of the Woize community.

SmartphoneWoize works on mobile phones running Windows Mobile 5.0 and can be used whenever the phone is connected to a WiFi network. It has similar functionality to the Woize PC client, providing users with instant messaging, voice mail, caller ID, call forwarding, SMS and other services.

http://www.woize.com

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Wi-fi phone market to double in 2006

The worldwide WiFi phone market jumped 76 percent between 2004 and 2005 to 85 million, and is projected to more than double in 2006, as enterprises continue to deploy voice over wireless LANs (VoWLAN), according to a report from market analysis firm Infonetics Research. (news - alert)

Healthy VoWLAN growth is projected through 2009, when WiFi phone revenue will hit almost 1.6 billion. Initially an enterprise application, VoWLAN will eventually become popular with consumers, too, and has potential for enormous growth as part of a VoIP service bundled with broadband connections.

WiFi phone units grew 112 percent between 2004 and 2005 and are expected to grow 158 per cent in 2006. Roughly two-thirds of WiFi phone revenue came from single-mode WiFi VoIP handsets in 2005, about a third from dual-mode handsets. By 2009, dual-mode handsets will make up three quarters of total revenue. Some 52 percent of dual-mode WiFi/mobile handset revenue came from Asia Pacific in 2005, 25 percent from North America, 21 percent from EMEA and 2 percent from CALA.
http://www.infonetics.com

Dual-Mode Cellular/WiFi Device Will Drive Enterprise VoWLAN Growth

Enterprise Voice over Wireless LAN (VoWLAN) usage is expected to grow dramatically during the next three years, according to a new study issued by InfoTech as part of the InfoTrack for Enterprise Mobility program. In its Mobile Communications in the U.S. Workplace report, InfoTech (news - alert) predicts annual U.S. enterprise VoWLAN revenues will reach $1.1 billion by 2010, with dual-mode cellular/WiFi devices contributing the bulk of these revenues.

"Cellular/WiFi devices are driving this growth — not the WiFi-only phone," says Jeanine Sterling, VP and program director of InfoTech's IEM program. "Not only does the number of companies purchasing dual-mode devices increase dramatically, but adoption of this handheld within these companies is expected to ripple quickly through once a critical mass of models is available."

She continues, "For vendors and providers, the implications are clear. Expect the single-mode, WiFi-only phone to remain a niche product. The main growth area is the dual-mode device and the infrastructure to support it. In light of the enterprise's desire for streamlined, converged communication, dual-mode cellular/WiFi handhelds are perceived by customers as the next logical step."
http://www.infotech.com

Calypso Wireless to Demonstrate the C1250i WiFi-GSM

Calypso Wireless, Inc. (news - alert)announced the Calypso ASNAP C1250i WiFi-GSM Dual Mode VoIP Smart cellular phone, which runs on a Intel PXA series application processor and Microsoft WinCE 5.0 operating system, interconnecting with Skype for its long distance calls.

Calypso' ASNAP patented technology is the solution to deliver seamless connectivity to mobile phones, Laptops, PDA and mobile devices.

"We feel that customer loyalty for mobile phone operators will increase with Calypso C1250i WiFi-GSM Dual Mode VoIP Smart cellular phones; mobile users will be able to seamlessly connect between cellular mobile GSM networks and WiFi networks and enjoy new added services, such as broadband connectivity and wireless VoIP services and real time video-calling, and will be able to save money on local or long distance call costs when connecting to any WiFi access point using Skype," says Alfredo Sarrazin, Vice President of Sales of Calypso Wireless Inc.

http://www.calypsowireless.com

Cingular adds WiFi-capable Windows phone

Cingular Wireless (news - alert) is offering a new 8100 series phone that is equipped with Windows Mobile and integrated with WiFi.

The new phone, along with the recently introduced 2125 Smartphone, forms the base of Cingular's move into the Windows Mobile 5.0 arena.

"Cingular now has a Windows Mobile 5.0 device offering that appeals to a broad swath of wireless users," said Cingular's Michael Woodward. "These are just the first two of what will be an array of Windows Mobile 5.0-based devices that will be available from Cingular by the end of this year."

Cingular said the 8100 series Pocket PCs come either without a camera or with a 1.3 megapixel digital and video camera. The Bluetooth WiFi capability works off Cingular's national EDGE wireless data network.

http://www.cingular.com

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ADLINK Intros PXI IEEE 488 Interface Controller Card

ADLINK Technology Inc., (news - alert) a leading provider of test and measurement products, released its first PXI IEEE 488 controller card. The PXI-3488 provides interface between GPIB instruments and PXI-equipped systems that is compliant with IEEE488.1 and IEEE488.2 standards.

The PXI-3488 is PXI bus compatible and represents ADLINK's innovations in PXI hardware design. It is suitable for most PXI platform and provides the connectivity between PXI platforms and GPIB instruments. In addition, the PXI-3488's 1KB on-board FIFO and high-speed bus accelerated by the on-board CPLD gives it a 1.5MB/s maximum data transfer rate to satisfy high-volume data transfer requirements.

The PXI-3488 has been hardware and software verified with a wide range of products and applications. It supports popular application development environments such as VB, VC++, Delphi, LabVIEW and TestExec and features "drop-in" system configuration to be compatible with existing test and measurement applications. The PXI-3488 supports Windows 98/NT/2000/XP and its driver library is compatible with industry standard VISA and instrumentation protocols.

http://www.adlinktech.com

Corrent Announces VoIP Security Gateway for Service Providers

Corrent Corporation, (news - alert) a leading supplier of purpose-built network security appliances, announced the Corrent® SR770, a carrier-class security gateway for VoIP service providers that employs VPN tunnels to protect VoIP calls from all known security vulnerabilities. In a major price/performance breakthrough, the SR770 is the first IPsec-based security gateway with the speed and capacity to service the high call-volume needs of consumer VoIP service providers.

Since it is based on existing IP network standards, the Corrent SR770 can be deployed as a "bump in the wire" behind the Internet access point of a VoIP service provider's operations center and will operate transparently to other VoIP networking equipment and software. The SR770 benefits from the extensive learning curve and public review process that helped to refine those standards and guarantee their interoperability.

With its tremendous speed, a single SR770 can handle up to 80,000 concurrent calls at an aggregate bandwidth of 8Gbps, and can be provisioned for as many as 400,000 subscribers. Since latency is both low (< 120 µs) and deterministic, the SR770 is able to add security without affecting voice quality. The SR770 also solves the so-called NAT transversal problem, where the VoIP phone is located behind a residential router or firewall, and will avoid consumer installation headaches.

http://www.corrent.com

Proactive Voice Quality Management: The VoIP Management Solution by Nortel and NetIQ

Successful VoIP Management

Business Challenge

The convergence of voice and data networks brings a tremendous amount of long-term cost savings but may also present unique challenges. Deploying and maintaining IP telephony applications can be difficult because of the critical role they play within your network.

In fact, solutions for monitoring and trouble-shooting VoIP service quality are among the highest ranked concerns of IT, telecom and network managers. These leaders are asking a variety of questions to determine how to best manage a VoIP infrastructure. They include:

- How do you know that your network is ready to handle the unique requirements these applications demand?
- How can you monitor the end-user experience versus the network performance?
- How will you troubleshoot problems in real time?
- How do you achieve these goals in a cost-effective and timely manner?

The Solution

Nortel's Proactive Voice Quality Management (PVQM) solution with partner NetIQ ensures you have the proper tools in place to maintain the highest VoIP service quality. Once you establish Service Level Agreements (SLAs), the solution will monitor and report on the necessary information to ensure you fulfill your SLAs.

PVQM provides a life-cycle approach that offers the necessary management tools to support each phase of an IP telephony project: assessment, pre-deployment and ongoing monitoring and reporting. It then provides a set of inter-related technologies that enable an effective Service Level Management solution for IP telephony. PVQM focuses on the end-user Quality of Experience (QoE) using standards-based technologies to ensure your service quality needs.

The PVQM Life-Cycle

Network Assessment and Pre-Deployment

Before deploying VoIP, you should have the proper equipment and identify if network upgrades are necessary. Fully supporting Nortel Network Heath Checks, NetIQ Vivinet Assessor determines quickly and easily how well VoIP will work on the network prior to deployment. Before you invest in costly training and pilot deployments, the product predicts the overall call quality you can expect from the network and generates polished, customizable reports detailing the network's VoIP readiness.

Vivinet Assessor focuses on five key components: network inventory, utilization assessment, configuration assessment, bandwidth modeling and VoIP quality assessment—all designed to ensure your network is ready to handle voice traffic.

Monitoring and Reporting

Once you deploy VoIP, you must monitor your environment to maximize uptime, reliability and quality. NetIQ's AppManager Suite is a robust platform that provides comprehensive monitoring, management and reporting for IP telephony solutions.

NetIQ AppManager for Nortel extends the suite to ensure the health and availability of Nortel's VoIP platforms, monitoring call quality in real time from an end-user perspective. Nortel's RTCP XR standards technology found in the phones provides the necessary end-user metrics and information to AppManager. Then, AppManager combines platform, system and call quality metrics for Nortel Call Servers with information about the availability and health of network devices and overall network performance for VoIP.

If a problem occurs in your voice network, NetIQ Vivinet Diagnostics quickly and automatically pinpoints issues and explains why you are experiencing poor call quality. Vivinet Diagnostics reduces the time and skills needed to resolve issues in both pre- and post-deployments. Features, such as root-cause analysis and diagnosis of problems between two Nortel phones, allow customers to minimize downtime and increase incident response time.

PVQM Summary

Successful implementation of VoIP applications requires integrated management solutions that allow you to take control of your entire voice network and server infrastructure. Understanding how data traffic will affect voice applications before deployment—and then continually monitoring and diagnosing the status of IP telephony devices in your environment—helps ensure success.

NetIQ provides the most comprehensive solution available on the market. Its assessment, monitoring and diagnostic products help you ensure a successful implementation and accelerate your overall return on VoIP investment. Count on Nortel and NetIQ to meet your VoIP Service Level Management needs—a crucial component of a successful VoIP deployment and management.

Randy Rosenbaum is Product Marketing Manager at NetIQ.

Shunra VE 4.0 Predicts Service Level Compliance for Applications and Infrastructure

Shunra Software Ltd., (news - alert) a leading provider of predictive application and network performance assurance solutions, announced version 4.0 of its Shunra Virtual Enterprise (Shunra VE) solution. Shunra VE simulates any production network environment in a pre-production setting.

This latest version of Shunra VE is focused on delivering detailed service level compliance analysis that enables users to make informed "go"/"no-go" application rollout decisions, easily assess and validate alternative solutions or technologies, and determine which modifications are needed to improve performance and ensure service level compliance — before deployment.

Shunra VE 4.0 is a powerful network simulation solution that creates an exact model of any production environment, including the network, remote locations, and the number and distribution of local and remote end-users. With Shunra VE 4.0 users can automatically test the functionality, performance, scalability and robustness of applications and network infrastructure under current and future real-world conditions — before deployment in production. Shunra VE 4.0 is extremely flexible and easy to install, configure and use. It seamlessly integrates with an existing lab environment, tools and testing methodologies.

http://www.shunra.com

BEA Systems Thinks Telecom Needs More Java

By Robert Liu

A year after first charging into the telecom space, middleware vendor BEA Systems (news - alert) has gained control of two key Java specifications outlining the application programming interfaces (API) used in call control protocols.

BEA Systems announced that it has taken over as Spec Lead for Java Specification Request (JSR) 289, entitled Session Initiation Protocol (SIP) Servlet v1.1, and SIP 32, entitled Java API for Integrated Networks (JAIN) SIP API Specification. JSR 289 is an enhancement to JSR 116, first developed by SIP solutions vendor Dynamicsoft to define a high-level extension API for SIP servers.

But after pushing through the final release of JSR 116, Dynamicsoft was acquired by Cisco Systems for its expertise in SIP-based networking, and abandoned development of its own applications infrastructure platform, opening the door for someone to step in as Spec Lead.

In addition to the Java community, BEA Systems also has joined the board of the Parlay Group and is helping to define a telecom networking Web services interface called Parley X. The Parlay Group is a telecom consortium focused on developing industry standard APIs to facilitate call control and messaging capabilities.

http://www.bea.com

Netrake Announces Revolutionary Fixed-Mobile Security Solution

By Erik Linask

Netrake (news - alert) announced the 3GX, its next-generation security platform — and the first complete solution addressing dynamic security of voice, video, data, and mobile applications. The 3GX supports next-generation, converged fixed-mobile architectures being defined by 3GPP IMS, 3GPP2, ETSI TISPAN, PacketCable, and UMA to support workers, whether on the road, in the office, or at home.

Building on the successes of nCite Session Border Controller (SBC) and nCite Security Gateway (SG) platforms, the 3GX integrates the two and incorporates new functionality and interfaces required to operate in both fixed and mobile networks. Netrake has developed unique networking hardware and software technology that solves the security and service problems with voice, video, and data distribution over FMC networks.

IMS architecture will require that service providers support a range of protocols. By integrating Netrake's SBC and SG technology into the new 3GX platform, service providers will be able to realize economies of scale. This, then, is converted into an advantage for carriers as they locate the security device at the very edge of the network to provide a single point of access to the core network.

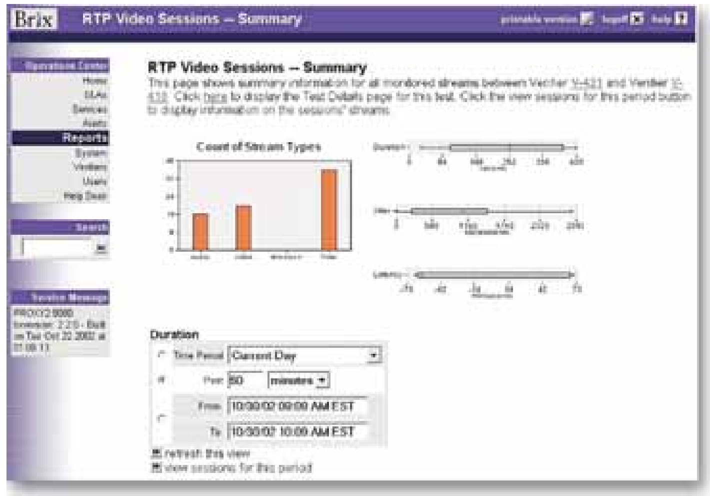
The 3GX addresses security issues with a carrier-grade, network edge solution, which is available in a range of configurations. It resides at the edge of the provider's IP network, bridging the divide between network convergence and service convergence, representing a sizable step forward in security and real-time user management. The 3GX allows service providers to manage a single secure connection to the subscriber that is capable of concurrently supporting all IP services such as VoIP, IPTV, and data services.

http://www.netrake.com

BrixVision Will Allow Providers to Monitor IPTV Quality

By Erik Linask

Brix Networks, (news - alert) provider of some of the world's most popular converged service assurance solutions, has unveiled BrixVision, it's IPTV service assurance portfolio. BrixVision gives service providers complete visibility into the quality of video content, the underlying delivery infrastructure, and the overall customer experience.



BrixVision builds on the company's expertise in VoIP and IP video service assurance, and has been developed specifically for IPTV deployments. It delivers the unequaled service visibility that providers have been requesting, yet has, until now, been unmet by legacy test and measurement tool suppliers.

BrixVision will enable service providers to perform root-cause analyses to identify IP transmission versus video quality impairments, monitor end-to-end video quality, and proactively monitor and manage their subscribers' overall and aggregate experiences.

In conjunction, Brix Networks has also introduced the IPTV Infrastructure Verification Package, the initial offering within the BrixVision portfolio that allows providers to perform continuous, proactive testing of their multicast infrastructure quality. By simulating user behavior and monitoring subsequent channel change response times, the Infrastructure Verification Package provides valuable insight into quality of experience (QoE).

http://www.brixnet.com

NACT Announces XL-MAX, Supports Excel CSP 2090

By Michelle Pasquerello

Telecom applications provider NACT (news - alert) announced the availability of XL-MAX, a suite of telephony applications designed with up-and-coming service providers in mind supporting the Excel CSP 2090 platform.

The CSP 2090's multi-functional platform design allows customers to build distinguished solutions to meet the needs of service providers. From unified messaging and automatic speech recognition to Web-initiated voice services and voice portal solutions, the CSP 2090 is an ideal solution for carrier-class communications services.

By merging XL-MAX software with Excel's hardware, global service providers will have the ability to deploy market proven revenue applications using TDM and IP delivery networks.

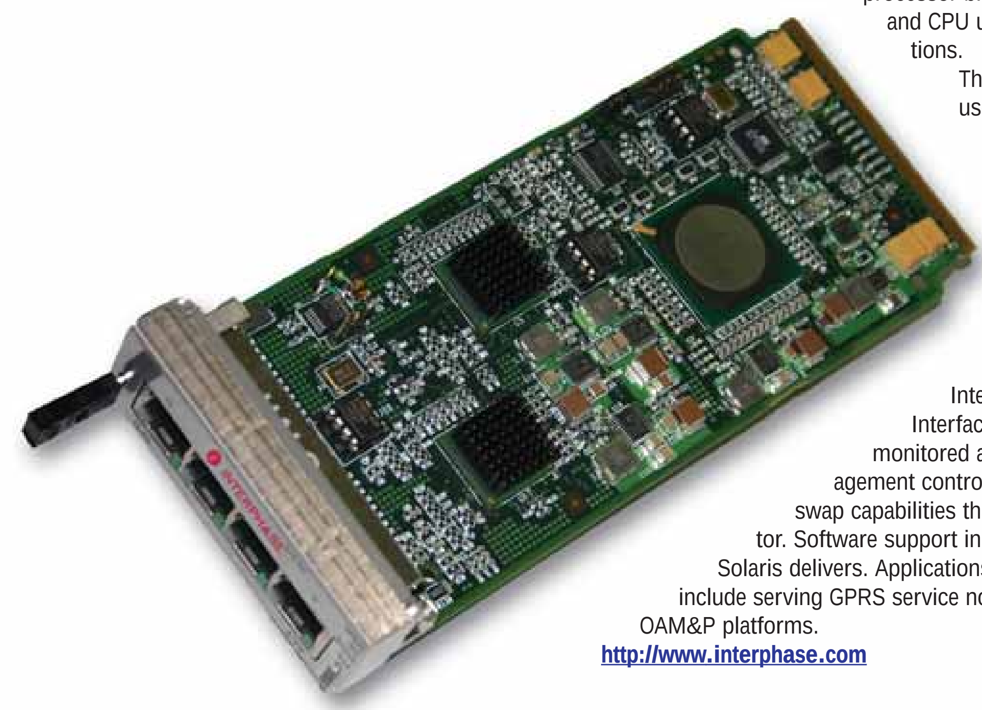
"NACT and Excel share a mutual focus on providing the most effective platforms for customers to deliver next-generation services quickly, easily and profitably," said J.C. Murphy, President, Excel Switching Corporation and Brooktrout Technology. "The powerful combination of NACT's application expertise and Excel's performance leading platforms enables service providers to confidently deliver cost-effective high-value VoIP applications."

http://www.nact.com

Interphase: Get the Most From Your Slot!

By Laura Stotler

Interphase (news - alert) has announced its SlotOptimizer 364G AdvancedMC Quad Gigabit Ethernet Card for delivery of advanced services in telecommunications networks. The new adapter is designed for next-generation and wireless network applications requiring multiple high-speed, high-bandwidth Ethernet ports directly connected to the processor blade. If offers optimum performance and CPU utilization for demanding applications.



The Interphase 366G is designed for use in AdvancedTCA and MicroTCA platforms. It features four independent 10/100/1000 Gigabit Ethernet ports, Gigabit Gigabit Ethernet controllers, up to x4 PCI Express 1.0a links, auto-negotiation and compatibility with PCI Express-compliant host systems.

The adapter also offers an Intelligent Platform Management Interface (IPMI), enabling the card to be monitored and controlled by remote shelf management controllers. The product also features hot-swap capabilities thanks to the AdvancedMC form factor. Software support includes Carrier Grade Linux and Sun Solaris delivers. Applications enabled by the SlotOptimizer 364G include serving GPRS service nodes, softswitches, MSC servers and OAM&P platforms.

http://www.interphase.com

Linksys Launches Affordable Family of Switches for Small Business Networks

By Erik Linask

Linksys, (news - alert) a Division of Cisco Systems, Inc. has announced a new line of Gigabit and Fast Ethernet switches for small businesses. The seven new switches, part of Linksys WebView switch family, help meet the needs of businesses looking for a cost effective way to provide high speed networking to their users or to easily migrate their networks from Fast Ethernet to Gigabit.

All ports include automatic MDI/MDI-X crossover detection, meaning that administrators don't have to worry about the cable type, and polarity detection, which will automatically correct wiring errors. Each port independently and automatically negotiates for best speed in either half- or full-duplex mode. Head-of-line blocking prevention keeps high-speed clients from bogging down in lower-speed traffic. Fast store-and-forward switching prevents damaged packets from being passed on into the network.

These new rack mountable switches include Linksys' WebView, which lets administrators control their networks remotely through a secure Web browser connection using HTTPS for simple configuration and monitoring. There is also a console port for those users who prefer to oversee their networks directly from a PC.

All the WebView switches are available now via the Linksys distribution and partner network, except for the SRW2024P and SRW248P which are scheduled to ship this summer. MSRP on the switches range from $530 to $740 (£300 to £415). Prices for the 2024 and 248 models will be determined once they become available.

http://www.linksys.com

SA Forum, PICMG, Mountain View Alliance Move Telecom Industry Closer to Open Standard Ecosystem

The Service Availability Forum (SA Forum) (news - alert) and PICMG have released the HPI-to-AdvancedTCA Mapping Specification. The mapping specification details how the SA Forum's Hardware Platform Interface (HPI) maps to PICMG's AdvancedTCA specification. This enables developers to use a standard method for implementing both specifications, thereby saving companies time, money and resources.

"The HPI specification represents a mature data model and programmatic interface to describe generic hardware platforms," said Dick Somes, PICMG's technical officer. "AdvancedTCA is generally perceived to be the underlying hardware platform for many HPI implementations. Thus a mapping between these two specifications is a natural outcome from these standards efforts."

The HPI-to-AdvancedTCA Mapping Specification exposes AdvancedTCA Shelf Management functionality and data in a standard, vendor-independent manner via the SA Forum's HPI.

By using open standards such as these, companies no longer have to develop proprietary solutions or devote resources to custom mapping between specs. This allows them to focus on their unique value-add, reduce their time-to-market and reduce lifecycle costs.

http://www.saforum.com

World's First AMD Opteron™ Based PICMG® 2.16 Single Board Computer

Performance Technologies, (news - alert) a leading developer of systems, platforms, components and software, introduced the CPC5564 Single Board Computer. The CPC5564, the world's first CompactPCI 2.16 compute blade based on 64-bit, single- and dual-core AMD Opteron processors, is part of Performance Technologies' next-generation compute products built for high-end telecommunications, defense and homeland security, and commercial applications.

The AMD Opteron processor provides a highly scalable x86 architecture that delivers next-generation performance as well as a flexible upgrade path from 32- to 64-bit computing. Its true multi-core architecture offers immediate, cost-effective technology to address today's processor design challenges–reducing heat and power consumption, by-products that occur at continually advancing single core processor clock speeds.

Designed to run Linux, Solaris and Windows® operating systems, the CPC5564 is the ideal computer for high-end packet processing or multi-threaded software applications found in wireless, softswitch, defense and other compute-intensive applications.

http://www.pt.com

Skype and Intel Collaborate to Create New and Improved Internet Communication

Skype, (news - alert) the global Internet communications company, and Intel Corporation (quote - news - alert) are collaborating to further integrate Internet communication into the fabric of everyday personal computer usage. The companies are working together to optimize Skype for Intel dual-core processor-based PCs and are planning to further enhance voice and video over the Internet so that it delivers the richest possible communications experience for personal and professional users.

The first result of the companies' joint technical efforts is the availability of free, ten-way voice conference calling for up to ten people in Skype 2.0 — an industry first for peer-to-peer Internet calling. The new feature is available exclusively for home and business users with Intel Centrino Duo mobile technology-based laptop PCs, and desktop PCs based on Intel Pentium D processors, Pentium Extreme Edition processors, and the recently introduced Intel Viiv™ technology.

The two companies plan ongoing, additional feature extensions and optimization of Skype for Intel's dual-core processors to take further advantage of the high throughput and simultaneous computing capabilities of Intel's dual-core processor architecture. Later this year, Skype will release video calling optimized for Intel dual-core technology, boosting performance and bringing free, high-quality video calling to millions of users with Intel processors. Beyond laptops and desktops, Intel and Skype share a common vision to enable Skype to function seamlessly across a wide variety of Intel-based computing platforms and network environments, including handheld computers as well as WiFi and WiMAX wireless networks.

http://www.skype.com
http://www.intel.com

ECI Telecom Launches Next-Generation SAN Extension Solution for Disaster Recovery and Business Continuity

ECI Telecom (news - alert) announced its next-generation SAN extension solution for transport networks to meet the growing market demand for mission-critical services. ECI's new BroadGate SAN XT enables disaster recovery and business continuity with high throughput of up to 2.5 Gbps for high-end storage applications over long distances — up to 1,600 km.

Carriers now have an end-to-end solution for connecting SAN clients and introducing new storage transport services to enterprise customers over their existing networks.

With ECI's SAN XT, traffic won't be interrupted in the event of hardware or line problems, such as a fiber cut or power failure. Mission-critical data is protected 24 hours a day, 7 days a week.

The BroadGate SAN XT meets the growing demand for disaster recovery and business continuity services by integrating with ECI's XDM multi-service provisioning platform (MSPP) family. Carriers now have an end-to-end solution for connecting SAN clients and introducing new storage transport services to enterprise customers over their existing networks. Being fully interoperable with carriers' SDH/SONET infrastructure, it is cost-efficient, and easy to install and use.

http://www.ecitele.com

New IBM Blade Computers Speed Business Data up to Ten Times Faster

IBM (quote - news - alert) introduced powerful new blade computing systems that enable data to travel up to 10 times faster than previously possible across corporate networks.

The revolutionary new high-performance systems, called IBM BladeCenter H, increase the bandwidth of tiny blade computers, providing businesses up to 10 times the capacity to move data across their networks. The processing breakthrough, made possible by IBM Research, increases the internal capability of the new system by delivering more than 40 Gigabits (Gb) of I/O bandwidth to every blade server.

The BladeCenter H systems introduced today provide a new way to deliver blade technology, by collapsing servers, storage devices, networking infrastructure and security appliances into a single location in the datacenter. They can be used by businesses to run high-performance analytics software and data warehousing applications in industries including finance, retail, healthcare, life sciences and telecommunications.

The new BladeCenter H systems provide crucial investment protection for IBM customers. They are compatible with existing BladeCenter systems and IBM's current blades and switches can be deployed in the new BladeCenter H system.

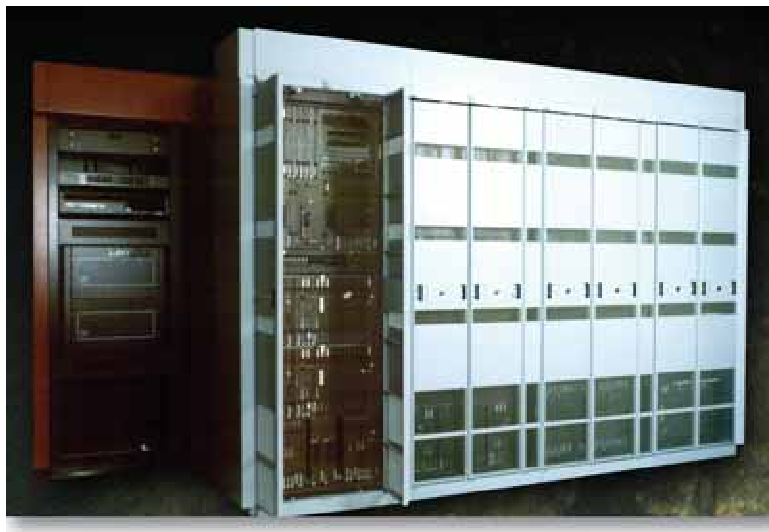
IBM's BladeCenter is the world's most popular blade computing system, with more than forty percent share of the market and revenue growth of 2,600 percent in less than three years.* Since its introduction in 2002, IBM has installed more than 350,000 blades for customers, more than any other computer maker.

http://www.ibm.com

Nortel Enables New Rural Market SIP Access Systems

By Johanne Torres

Carriers looking into offering triple play services now have the option of tapping telecom technology provider Nortel (quote - news - alert) to enable new SIP fiber to the home (FTTH) and converged IP access systems that will help them deliver broadband data, video and a set of voice features to residential and business customers in rural areas.



Nortel's new systems are enabled by the company's DMS-10 softswitch interoperating with products from Allied Telesyn and Pannaway Technologies. The bundle gives rural carriers access to a new portfolio of services such as VoIP with E911 lifeline calling, enhanced IP video, and video on demand.

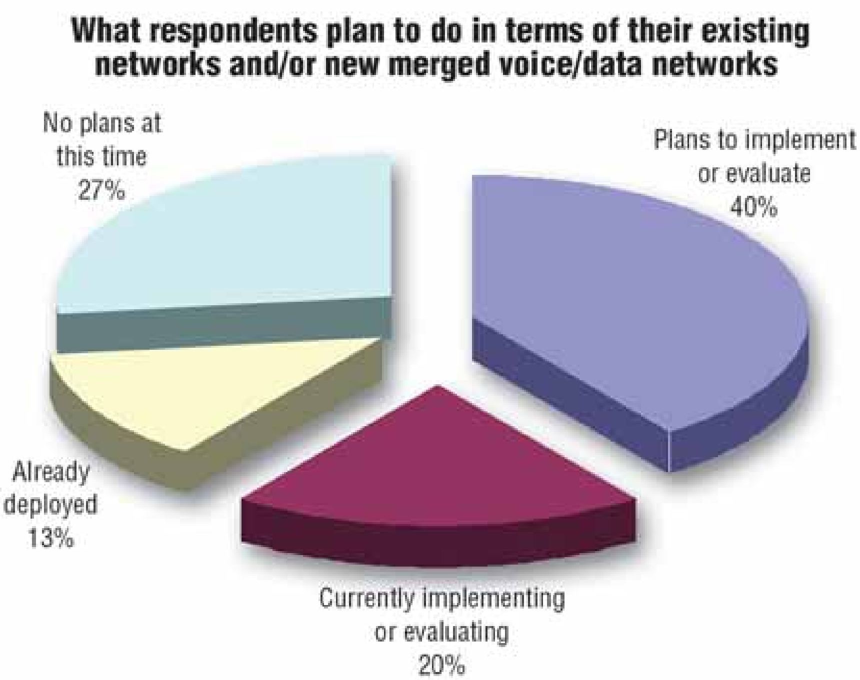
"As we evolve to VoIP, it is absolutely critical that we have the ability to provide advanced calling and safety features such as E911 lifeline support as well as new applications we can run on an application server. Interoperability between Nortel's DMS-10 and Pannaway's SCN will allow us to offer premier video and data services along with support for emerging applications such as HDTV and video on demand," Joseph Gottwald, CO engineer and ISP manager, Empire Telephone, which will implement the system.

http://www.nortel.com

SIP Trunking Simplifies Conversion to VoIP, Saves Money

By Mae Kowalke

A recent survey by Computer Technology Industry Association (news - alert) found that 60 percent of small and mid-sized businesses (SMBs) plan to increase their use of converged voice and data communication solutions during the next 18 months.



Survey respondents were technology decision makers at companies with 20-500 employees.

One reason for the growth in converged communications is the advent of SIP-trunking — a way for SMBs to cut down considerably on the startup and management costs of switching to VoIP phone service.

SIP-trunking eliminates the need for gateways by using software to manage a company's VoIP service, and by utilizing the carrier's already-existing network of gateways.

When both caller and receiver are set up with internet telephony systems, SIP-trunking is especially efficient, because it allows a call to travel the entire way as a digital signal.

For many SMBs, using a software-based, SIP-trunking platform means the payback for switching to VoIP can occur in as little as 3-12 months, Landry said.

http://www.comptia.org

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Stinghorn Offers NAT Traversal for VoIP and SIP Traffic

With Stinghorn's (news - alert) VPN solution any company can utilize inexpensive Internet telephony to support their own business operations. Oy Stinghorn Ltd is a Finnish-based software solution vendor specializing in secure interconnections over IP networks. Using VPN connections provided by Stinghorn Secure Business Suite solution, companies can route SIP-based communication securely into their own internal network and back to the remote terminal. With Stinghorn's solution, VoIP connections can also be created through Network Address Translation (NAT).

Stinghorn's solution is not bound to any particular protocol nor to any particular Internet telephony vendor. On the contrary, it can be utilized to enable the use of practically any vendor's VoIP/SIP solution from anywhere in the world. With Stinghorn's solution, it is also possible to create a secure Internet phone call between two users residing in the same or separate remote networks.

Stinghorn's security and interconnection solutions are available from Stinghorn and selected partners worldwide both as complete shipments and as turnkey services with monthly fee.

http://www.stinghorn.com

MAILVISION Announces its APPLYNX SIP Application Server for IMS Enabled Networks

The APPLYNX (news - alert)is already deployed with several NextGen service providers enabling service delivery of a number of applications such as prepaid, voice mail, conferencing, Recording and a variety of IVR based services. The APPLYNX platform has proven interoperability with all SIP network elements; Softswitch, Media Server, Media Gateways, and IMS network elements.

"MailVision's APPLYNX is at the core of a rich service delivery eco-system designed for the IMS architecture providing Fixed Mobile Convergence solutions. It can easily support any SIP based application," says Benjamin Bar-Ness, VP of Business Development for MailVision.

The MailVision APPLYNX SIP Application Platform is that enables service providers to build scalable, reliable packet-voice services. MailVision's unique IP-based architecture directly integrates into an IMS network infrastructure and delivering cost-effective, revenue-producing enhanced communications services.

http://www.mailvision.com

Avaya IP Telephony Takes Flight with U.S. Fighter Jet

Avaya Inc. (quote - news - alert) announced that IP Telephony has taken to the skies, linking an F-15E fighter jet in flight with a remote government command center in California and a three-star general at the Pentagon via a multi-party, IP conference call.

"The call lasted for more than 20 minutes, and the audio quality was excellent," said Ryan Greene, a Boeing engineer who worked on the test with Avaya and participated in the call.

"Being able to call a tactical aircraft from anywhere in the world and vice versa is a critical combat capability unprecedented with legacy communications," said Lt. Col. Stephen Waller, USAF, DARPA program manager for the Tactical Targeting Network Technology program.

Both tests used Avaya's industry-leading Communication Manager IP telephony software hosted on an Avaya Media Server. Avaya SIP Enablement Services were used to ensure connectivity with standards-based endpoints for telephony, instant messaging, conferencing and collaboration. The multi-party conference call involved both an Avaya IP phone and an Avaya SIP IP telephone at China Lake and a traditional desk set at the Pentagon. A third-party SIP softphone running on a Windows-based auxiliary computer was used in the F-15E cockpit, connecting the pilot to the conference via a secure wireless link.

http://www.avaya.com

Abbeynet Introduces Abbeyphone Deskbar VoIP Application

Abbeynet, (news - alert) a European leader in the development of cutting-edge VoIP technologies, presents Abbeyphone Deskbar, the easiest way to call from the Internet, intended for all Abbeyphone users.

Abbeyphone Deskbar is a VoIP application that installs onto the Windows taskbar and provides immediate access to Abbeyphone communication services.

With Abbeyphone Deskbar, you can call fixed-line and mobile phones throughout the world, make and receive calls to and from other Deskbars and other IP terminals, both hardware and software, over the Internet free of charge using Session Initiation Protocol (SIP).

Abbeyphone Deskbar also includes a plug-in for Microsoft Outlook, which brings all of these functions directly to your e-mail client, so that you can interact quickly and efficiently with your list of contacts.

http://www.abbeyphone.com

Covergence Integrates McAfee to Deliver First SIP-Based Anti-Virus Solution

Covergence, (news - alert)a leading provider of unified security and management solutions for applications and services based on SIP, announced that it has integrated technologies from McAfee, Inc. to ship the first complete anti-virus solution for SIP-based applications and services. McAfee's award-winning anti-virus solution is now tightly integrated into the Covergence Eclipse family of SIP application management products enabling enterprises to benefit from multi-modal, real-time collaboration while protecting them from the dangers of SIP-borne malware.

The proliferation of applications and services based on SIP and its extensions for instant messaging and presence (SIMPLE) represents a new challenge for enterprises concerned with protecting themselves against dangerous malware. One of SIP's greatest strengths is that it supports multi-modal collaboration applications that can support several different forms of real-time communication simultaneously. While this offers tremendous business benefit, it also creates serious security vulnerabilities.

"The explosion of SIP-based applications and services has created a whole new set of security concerns for enterprises," said Bob O'Neil, president and CEO, Covergence. "With McAfee Anti-Virus for Eclipse, organizations can finally have the peace of mind that comes with knowing their SIP-based solutions are protected."

http://www.covergence.com

NICE and Avaya to Deliver IP Contact Centers

By Johanne Torres

NICE Systems (news - alert) announced that it joined forces with Avaya (quote - news - alert) to deliver IP-based contact centers to businesses. The newly extended alliance between both companies will include developing a joint offering of a suite of applications and working on strategic marketing initiatives.

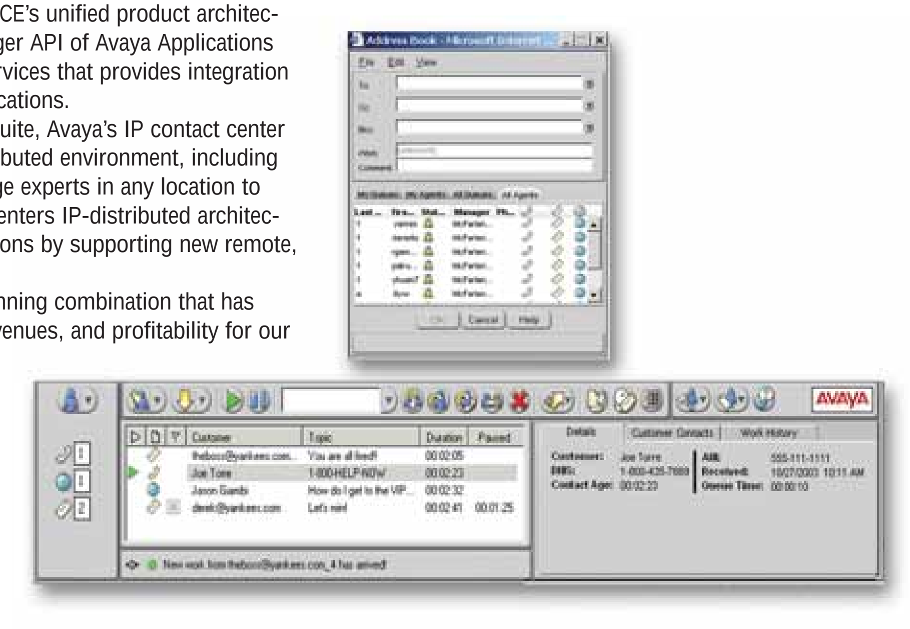
The joint system offers monitoring, branch recording, centralized storage, liability, and interaction analytics. It is an integral part of NICE's unified product architecture, and integrates the Communication Manager API of Avaya Applications Enablement Services (AES), a suite of Web services that provides integration with Avaya's converged communications applications.

As part of the Avaya Customer Interaction Suite, Avaya's IP contact center solutions help businesses link together a distributed environment, including agents, branch office employees and knowledge experts in any location to serve customers faster. Avaya offers contact centers IP-distributed architectures to flatten, consolidate and extend operations by supporting new remote, branch, satellite or global agents.

"The Avaya-NICE partnership has been a winning combination that has been successfully driving customer loyalty, revenues, and profitability for our mutual customers," said Zvi Baum, NICE president of Enterprise Interactions Solutions. "We are very excited about extending our long standing relationship with Avaya with this new agreement. By combining our capabilities, especially for complex high-end, mission-critical VoIP environments, we are spearheading the IP-based contact center of the future."

http://www.nice.com

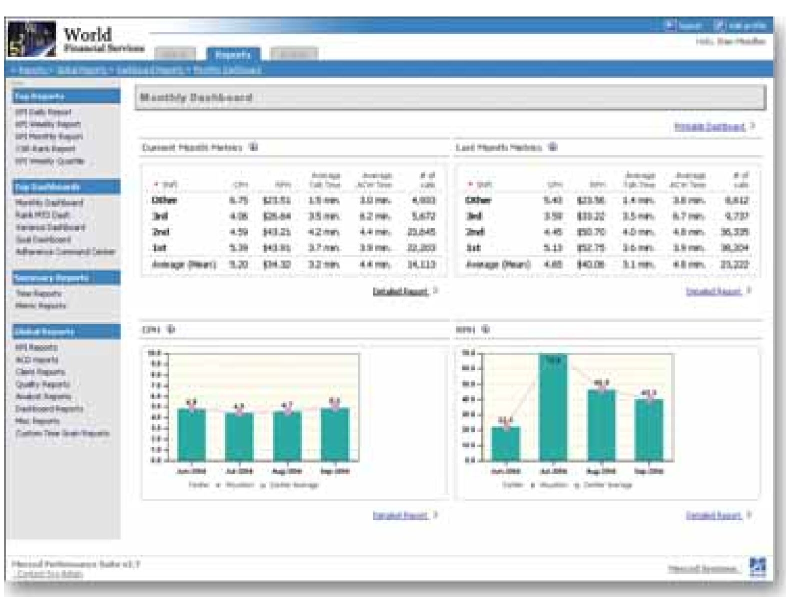
http://www.avaya.com



Merced's Suite 2.7 Hits The Market

By David Sims

Merced Systems Inc., (news - alert) a vendor of contact center and operational performance management applications, announced the release of Merced Performance Suite 2.7.



The Merced Performance Suite is designed to increase "the efficiency of contact centers by consolidating data from disparate sources, delivering advanced analytics, providing personalized dashboards, and integrating workflow and process improvement tools," according to the company.

The suite's data management capabilities claim to reduce application administration and deliver "the lowest total cost of ownership on the market," according to Michael Schmier, Merced's vice president of products.

With the release of version 2.7, the Merced Performance Suite extends its workflow and data management capabilities by introducing additional pre-configured workflows for best practice processes for agent development and performance improvement plan tracking, and extended data management capabilities to "allow customers to more easily and rapidly extend their applications."

http://www.merced.com

Five9 Releases Virtual Call Center Inbound Solution

By Mae Kowalke

Five9, (news - alert) provider of customer interaction management solutions, announced the launch of its new Virtual Call Center Inbound product.

Virtual Call Center Inbound enables inside sales and customer service departments to efficiently manage high volumes of inbound calls.

The product automates call routing using a variety of functions including phone trees, “skills groups,” and prioritizing based on customer type.

Using Virtual Call Center Inbound, supervisors also can record calls for quality control, and provide agents with scripts and worksheets.

“Technologies like Five9’s can help companies create customer-centric inbound call centers that increase sales productivity and bottom-line profits, while offering excellent service which ultimately improves customer satisfaction and leads to repeat business and referrals,” said Sheryl Kingstone, program manager at Yankee Group, in a press release.

http://www.five9.com

Extensive Headset Usage Poses Greater Risk for Early Hearing Loss

By Susan J. Campbell

Recent news for call and contact centers covers technological advances that can improve performance and lower attrition for the agent base. But an area that is starting to receive more attention is hearing loss due to headset use.

In all fairness, call and contact center management ensures that the equipment provided to their agents is ergonomically safe and does not pose potential damage to hearing. The greater risk is unfortunately often caused outside of the call or contact center. According to the National Safety Council, (news - alert) agents can receive acoustic shock injury from noise that travels over telephone communication equipment due to electronic feedback, fax modems or even malicious callers who use devices such as whistles.

At this point, there is little research supporting the implication that certain styles of headphones or headsets are more dangerous than others; however sound-minimizing headsets for music players are becoming more popular.

So, to minimize potential damage, precautionary measures are essential. Call and contact center agents can help prevent hearing loss by following simple guidelines like taking breaks and maintaining volume levels. Most importantly, both types of users must take it upon themselves to ensure that their hearing is being protected. Don’t assume it is a priority for anyone else.

http://www.nsc.org

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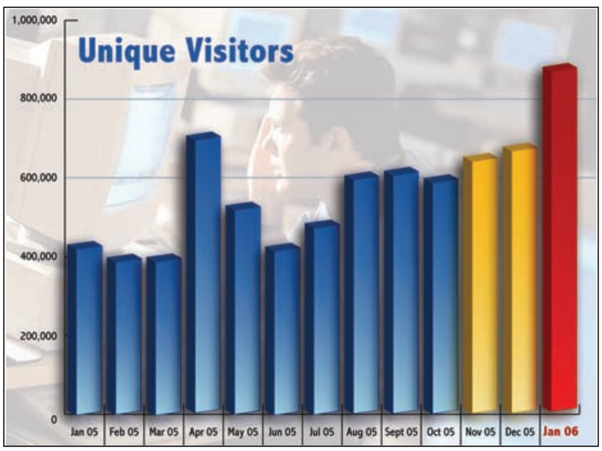


- **Over 13.8 Million Page Views***
- **Over 895,000 Unique Vistiors***

Source: Web Trends February, 2006

**Source: Alexa.com ranks Web sites by traffic. The number indicates a site's proximity to being the number one most visited Web site. Date: 2/10/06
Alexa is an Amazon.com Company. Neither Alexa.com nor Amazon.com endorse, or are affiliated with, TMCnet.com in any way.

TMCnet.com Traffic vs. Technology/IT Web Sites

Web Site	Alexa Site Rank
TMCnet.com	1,861
eWeek.com	2,043
Computerworld	3.274
InfoWorld	4,361
Network World	7,311
Light Reading	12,686
Wireless Week	42,378
Pulver.com	46,182
Destination CRM	58,411
Telephony Online	59,050
VoIP News	74,463
Telephony World	191,881
America's Network	221,623
Telecomweb	230,307
CommWeb	238,885
Call Center Magazine	254,302
Wireless Review	496,188
Communications News	635,124

TMCnet.com Traffic vs. Business Magazine Web Sites

Web Site	Alexa Site Rank
TMCnet.com	1,861
Smart Money	3,067
Fast Company	4,001
Red Herring	4,914
Inc. Magazine	6,048
Fortune Magazine	6,093
Barron's Online	7,960
Technology Review	8,267
Weekly Standard	8,954
CIO Magazine	13,231
BtoB Online	25,846
Worth Magazine Online	194,856

TMCnet.com Traffic vs. Fortune 500 Company Web Sites

Web Site	Alexa Site Rank
TMCnet.com	1,861
GE	2,830
Ford	3,406
General Motors	4,132
State Farm Insurance	6,681
Coca-Cola	8,301
DuPont	16,121
Kroger	21,778
AIG	27,670
Chevron Texaco	32,760
Exxon Mobil	34,341
Fannie Mae	39,669

TMCnet Traffic Analysis

Note: Alexa.com ranks Web sites to their proximity to being #1.
The lower the number, the higher the ranking and therefore the greater the traffic.
Yahoo!, the world's busiest Web site, is ranked #1 by Alexa.com

To Advertise Please Contact Dave Rodriguez at 203-852-6800 Ext.146 • drodriguez@tmcnet.com

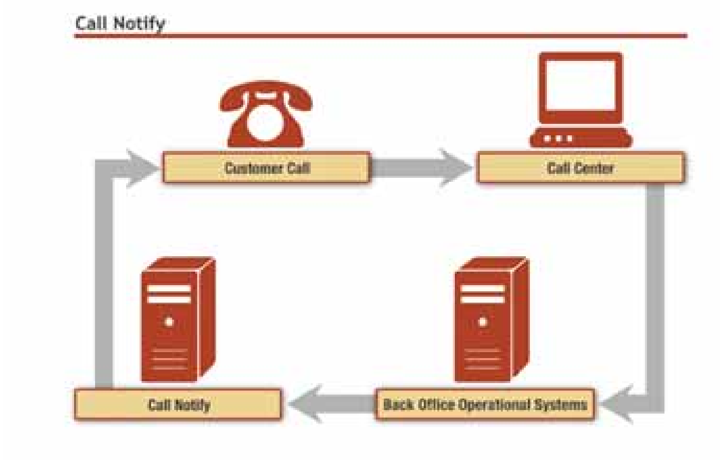
SkyCreek Launches Call Notify IVR Platform 5.2

By Johanne Torres

Customer contact systems provider SkyCreek Corporation (news - alert) recently launched version 5.2 of its Call Notify IVR Platform. This latest version includes an Expression Builder, File Feed Wizard, Enhanced Monitoring and Reporting.

The Call Notify IVR Platform is a Web-based, next-generation intelligent IVR platform, that enables enterprises to proactively update, inform, up sell, advise, educate, train, and/or survey their customers, while delivering better customer service.

The platform includes a CallDesigner for use in designing any voice and/or IVR applications and campaigns; a CallManager for managing and scheduling voice and/or IVR applications and campaigns; and CallReports for generating reports for any voice and/or IVR applications and campaigns.



The Call Notify IVR Platform now includes an Expression Builder that enables IVR developers or SkyCreek developers to build logical expressions to facilitate the real-time customization of the call flow to be delivered via any inbound or outbound voice application. This feature will for example, allow an IVR developer to alter the greeting based on the time of day.

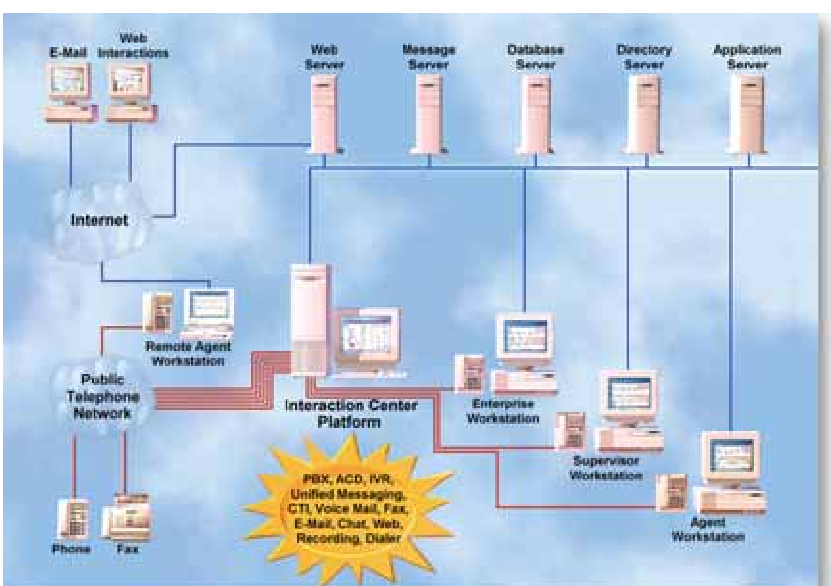
http://www.skycreek.com

Interactive Intelligence Automates Outbound Notifications

By Laura Stotler

Interactive Intelligence (news - alert) is now offering a hosted notification service to cut costs associated with outbound customer interactions. The icNotify service automates traditional manual processes and eliminates the need for on-premise customer equipment and network bandwidth.

The new service will be targeted at financial service institutions and physician offices initially. icNotify will offer outbound notification services for automated messaging, offering subscribers flexible, multi-channel contact options based on their unique needs. Financial institutions can use the service to send customers notification of loan status, fraud alerts, deposit confirmations and portfolio updates. Physician offices can send notifications like prescription reminders, appointment reminders, “agent-less” health education campaigns and patient satisfaction surveys.



icNotify is switch independent and can work with a number of third-party PBXs and IP-PBXs. The service will be delivered via landline and mobile phones, as well as through email, fax, pagers, PDAs and short message service-enabled devices. The service will use text-to-speech, speech recognition, pre-recorded messages, on-the-fly message recording and self-service options. The self service is designed to minimize inbound call volume in response to outbound notifications.

“For a typical financial services firm, the average outbound call costs $2.70 – a cost largely attributable to labor,” said Art Schoeller, senior analyst, Yankee Group. “Services such as icNotify can go a long way toward displacing a major portion of that expense through automation. In addition, customers are better served since these services are geared to deliver messages based on the customer’s preference, such as e-mail, text message, or recorded voice call.”

http://www.inin.com

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Wicom CSS 5.0 Software Allows Fully Mobile All-IP Customer Service

Wicom Communications, (news - alert) a leading European provider of all-IP contact center solutions and VoIP enterprise telephony software for fixed and wireless network environments, today announced the latest version of its Wicom Communications Server Suite (CSS) software.

Wicom CSS software version 5.0 adds a host of new features to the telephony and contact center solution, which integrates fixed-line and mobile communications into a cost-efficient, centrally managed system. In addition to enhanced user performance and improved business application integration, Wicom CSS now provides enterprises with true virtuality by integrating rich mobile phone functionality with the comprehensive contact center and telephony capabilities of the software.

With Wicom CSS 5.0, all incoming calls, e-mails, Web contacts, and SMS messages are handled quickly and efficiently through a single user interface. Employees may log onto any workstation on the company network and immediately access their personalized telephone functions. The Wicom softphone application also integrates with Microsoft Outlook software, further improving the availability of office-based employees. A workstation or IP phone can be paired with a mobile phone, allowing all calls to be answered on either phone.

With its industry-standard Web Services application interfaces, the new Wicom CSS 5.0 integrates seamlessly with core telephony and IT systems such as existing PBX, CRM and ERP solutions.

http://www.wicom.com

InPhonex Enhances their Industry-Leading VoIP Reseller Program through VarPhonex

InPhonex, (news - alert) a leading supplier of Internet telephony solutions, announced additions to VarPhonex, (news - alert) their popular reseller program. Developed in response to the need for a lower price point and more flexible reseller solutions, the VarPhonex Program supplies small businesses and entrepreneurs with a complete "VoIP-Reseller-in-A-Box" solution.

New options for resellers now include:

• Choice of a Free Linksys PAP2 Adapter or a 25% discount on new orders for the U.S. and Canada Unlimited Plan.

• A 25% discount on the reseller price for the InPhonex Brazil Unlimited Plan.

• Support of Incoming International Phone Numbers (DID numbers).

"We remain highly market-responsive to our resellers' needs in the fiercely competitive VoIP market," said Todd Hirshorn, InPhonex Chief Marketing Officer.

"Resellers need not only a robust network infrastructure, complete 'private label' services but extremely price-competitive offerings to compete with the larger players."

http://www.inphonex.com
http://www.varphonex.com

Nortel, Huawei Join Forces

By Cindy Waxer

Hoping to grab a sizable piece of today's global broadband access equipment market, Nortel (quote - news - alert) has entered into a memorandum of understanding with Huawei Technologies, (news - alert) a provider of next generation networks for telecommunications operators. The intent is to establish a joint venture for developing ultra broadband access solutions for markets around the world. The two companies have also entered into a supply agreement that allows Nortel to immediately begin engaging customers with Huawei's current broadband access portfolio.

The new company will combine Huawei's broadband access solutions with Nortel's voice and broadband networking technologies to create a new product portfolio. These products will give service providers the ability to converge the delivery of voice, video, data and wireless services to business and residential customers onto a common IP platform that supports copper, fiber and fixed wireless networks.

The joint venture will be majority-owned by Nortel and headquartered in Ottawa and will be focused on product enhancements for Huawei's current broadband access portfolio and the development of a new ultra broadband product portfolio.

http://www.huawei.com
http://www.nortel.com

Communication Server 1000

Communication Server 2000

Communication Server 2000 - Compact

AccessLine and Global Crossing Extend Contract

By Cindy Waxer

Marking an extension on an already existing five-year relationship, Global Crossing (news - alert) has agreed to provide AccessLine Communications Corporation, (news - alert) a provider of hosted communications and managed voice services, with a suite of high-performance VoIP services including VoIP Outbound, VoIP Toll-Free and VoIP DID.

"VoIP is critical to our continued business growth," said Doug Johnson, AccessLine's chief executive officer. "Our partnership with Global Crossing enables us to deliver our turnkey VoIP application suite to customers over Global Crossing's high quality VoIP network."

Transition to an IP-based network is a key enabler for AccessLine as VoIP transforms the world of voice by enabling users to access a range of new features more efficiently. Currently, Global Crossing runs more than two billion minutes per month of VoIP traffic over its private, global backbone. The company's VoIP services deliver carrier-class quality, backed by end-to-end IP service level agreements for jitter, packet loss, availability and latency.

http://www.accessline.com
http://www.globalcrossing.com

SIPquest Joins Symbian Platinum Program

By Laura Stotler

SIPquest (news - alert) announced that it has joined the Symbian (news - alert) Platinum Program. The SIPquest Mobile Assistant works with Symbian OS phones to deliver instant messaging, presence and corporate directory services to IP PBX users.

As a member of the Platinum Program, SIPquest will extend personalized control of communications services through fixed mobile convergence (FMC) and IP Multimedia Subsystem (IMS) solutions. These solutions will operate over WiFi, GSM and CDMA network interfaces to the Symbian OS ecosystem.

Symbian develops and licenses the Symbian OS for many popular smartphones. Nearly 48 million Symbian OS phones have shipped worldwide to date. The company's Platinum Program features companies that have a technology or strategic position that is key to the success of mobile computing technology surrounding Symbian OS. Platinum Partners may benefit from a range of commercial and technical services, including joint marketing opportunities, technical support along with privileged access to Symbian OS source code and a dedicated Partner Support team.

"With the rapid growth of the smartphone market, multimedia services have become a fundamental part of today's advanced open-standards based mobile phones," said Jerry Panagrossi, vice president of US operations, Symbian. "The addition of SIPquest to the Symbian Platinum Program underscores our joint commitment to bring mobile VoIP technology and compelling fixed mobile convergence (FMC) solutions to the ever expanding and increasingly sophisticated Symbian OS ecosystem."

www.sipquest.com
www.symbian.com

Telrex Call Recording Software Helps Covad Expand Its Market Share

By Mae Kowalke

Telrex, (news - alert) developer of IP PBX call recording and monitoring software, announced that it is partnering with voice and data service provider Covad Communications, (news - alert) adding call recording functionality to Covad's PBX service

Teaming up Telrex's CallRex software with Covad's vPBX voice service, the two companies are creating what Telrex claims is the first IP-based call recording software solution offered by a hosted PBX provider.

Using CallRex — a software-only IP-based call recording solution — Covad will be able to attract customers who require their PBX service to include call recording.

"CallRex expands our market to all of the small and medium-sized business that won't subscribe to a hosted IP PBX solution without having a call recording solution," said Scott Scherer, director of product development at Covad Communications.

http://www.telrex.com
http://www.covad.com

3Com Acquires Majority Ownership of Huawei Joint Venture

By Cindy Waxer

Only a day after Nortel announced entering into a memorandum of understanding with Huawei Technologies to establish a joint venture, 3Com Corporation (quote - news - alert) has announced that it has purchased from Huawei an additional two percent interest in Huawei-3Com Ltd.

Huawei-3Com is a China-based joint venture formed by 3Com and Huawei in November 2003. The sale was completed on January 27, 2006 following final approval from the government of the People's Republic of China, resulting in 3Com owning 51 percent of the joint venture and Huawei owning the remaining 49 percent.

On October 28, 2005, 3Com reached an agreement with Huawei to gain majority control of the joint venture, subject to Chinese government approval and usual closing conditions. 3Com agreed to pay Huawei $28 million in consideration for two percent of the outstanding shares of Huawei-3Com owned by Huawei.

"When the joint venture was formed in 2003, we had three key objectives: to establish a substantial presence in China, the world's fastest growing market; to create a resource capable of building enterprise-class, cutting-edge switching and routing products faster than we could deliver on our own; and to capitalize on a rapidly growing pool of engineering talent," said Scott Murray, president and CEO of 3Com.

http://www.3com.com

Voda One Takes Delta Roadmap FastTrack Training Series Across U.S.

Westcon Group, Inc. announced that its Voda One division (news - alert) has launched the third season of its Delta Roadmap FastTrack training series in selected cities across the United States. The in-person, classroom style training features in-depth product overviews for Avaya best- in-class solutions for enterprises and small and medium businesses (SMBs), as well as an overview of complementary vendor offerings from Westcon Group's affinity portfolio.

Also featured in the course will be several of Westcon Group's core affinity vendors which enhance Avaya solutions: Meru Networks, SpectraLink, Extreme Networks, NICE Systems and Metropolis Technologies. Avaya Financial Services will also be on hand for any resellers interested in learning more about credit, leasing or financing options.

Rob Linder, vice president of sales for Voda One, commented: "Our Delta Roadmap Reseller Training helps our Avaya Business Partners become more familiar with the latest Avaya releases while providing them with increased insight on specific business applications. At the end of the day, Avaya BusinessPartners walk away with a clear understanding of how Avaya solutions, when combined with other solutions from our affinity portfolio, can impact their sales and increase the competitiveness of their own customers."

http://www.vodaone.com

By Tony Rybczynski

Getting Users Turned Onto IP Telephony: The Converged Desktop

The industry is abuzz with IP Telephony, but what's in it for you and other end users in your business unit? IP Telephony can deliver the same feature richness, reliability and performance that you have grown accustomed to… and much more. What you may not realize is that IP Telephony can also deliver new desktop capabilities that can make you much more effective.

Traditionally, a phone and a PC on the same desk operated totally independently of each other. Today, the Converged Desktop creates a totally new workspace. The phone itself can now provide enriched communications; can display corporate alerts and have business content pushed to it; and can be augmented through USB-attached peripherals (e.g., a keyboard or wireless headset). Enriched communications includes corporate and departmental directories with click-to-call, conference managers that simplify chairman controls and enhance security, push-to-talk capabilities and zone paging that speed up communications, and visual voicemail that accelerates voicemail handling. Alerts include security alerts, weather alerts, IT alerts, travel advisories, and company announcements. Content that can be pushed to IP phones can stretch the imagination. It can include general purpose information such as headline news and stock prices, and work-related applications such as management dashboards providing daily sales figures, inventory levels, manufacturing efficiency, and call center status. In addition, vertical applications can be provided, for example: restaurant reservations and room and maid status in hospitality; inventory lookup and price checks in retail; time clocks in manufacturing; and student attendance tracking in K–12.

While some vendors choose to put a browser in every phone and limit access to Web-based applications, the business need is better served with information from targeted applications being securely tailored for the best user experience, in a way that doesn't require any changes to the application. The converged desktop also leverages the power of the PC to deliver real-time converged communications including voice, multimedia conferencing, application sharing, instant messaging, and presence-enabled personal agents.

The converged desktop leverages the power of the PC to deliver real-time converged communications.

Desktop Phones: Hard or Soft Clients?

With IP Telephony, users have a choice of using an IP phone as their telephony instrument, or to install a soft client on their PC. The former has the advantage of exhibiting the familiar look and feel of a phone, including use of a handset or hands-free operation, and the availability of a familiar numeric key pad with associated numerics (i.e., 2 and A, B, C). Convergence does not imply dependence: if the PC is being rebooted to install a patch, the phone keeps working, even in the Converged desktop scenario.

The big advantages of an IP Telephony soft client running on a PC (which can have a similar look and feel through a point-and-click key pad on the screen), are that your quality of experience is the same whether you are in your office or connected remotely over a broadband connection. It is also a lower cost solution and requires a smaller footprint on the desk, as long as the user is comfortable with using a headset. Hands-free operation, however, is problematic. Of course, power over Ethernet can provide continuous operation for IP sets in case of power failures, but is not applicable to PCs because they draw too much power.

The Value of Video for Business

The key technologies that have made video much more affordable are converged IP networking, low cost cameras, and digital processing and storage on general purpose PCs. One-way video streaming is central to video surveillance, training and employee communications. In some industries (e.g., brokerage), live TV news feeds can be streamed to desktops to keep users plugged into world and industry developments.

Many studies have been done to assess the value of two-way interactive video in real-time inter-human communications. R. B. Ochsman and A. Chapanis found that the value of interactive video follows a bi-modal distribution with a low value region between high value conflict resolution ("I want to see the whites of their eyes") and high value personal interactions ("Hi grandma"). They also found that "The most important determinant of a team's problem solving speed was the presence of a voice component."

In business settings, Ellen Isaacs and John C. Tang found that desktop video conferencing enhances the richness of the interaction. On the other hand, when compared with face-to-

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face, it can be difficult to control the floor, have side conversations and manipulate objects. "To fully enable rich interactions, video should be integrated with other distributed tools that increase the extent and type of shared space in such a way that it enables natural collaborative behaviors."

The implication is that, while consumer interactive video is growing to be a hot market, the value of business video is greatest when viewed as part of a real-time converged communications environment, leveraging the power of the PC and the converged desktop.

Vote for What You Need

While IT has the overall responsibility for establishing and evolving the enterprise IP telephony and real-time converged communications infrastructure, it's the end user who will ultimately reap the benefits of productivity enhancements through collaborative tools. Therefore, business units and end users are key stakeholders in the evolution of the business to leverage the benefits of IP Telephony and multimedia collaboration. The converged desktop is but one of a myriad of opportunities that can enhance your productivity, whether you are in the office or workspace, roaming the site (e.g., warehouse, store, and hospital) or campus, or on the road or working from home. IT

References:

R. B. Ochsman and A. Chapanis. "The effects of 10 communication modes on the behavior of teams during co-operative problem-solving." International Journal of Man-Machine Studies, 6(5):579—620, Sept. 1974.

Ellen Isaacs and John C. Tang, "What Video Can and Cannot Do for Collaboration", Published in 1994 in the Springer-Verlag journal Multimedia Systems, Volume 2, pp. 63-73.

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

By John Cimko

The Network Neutrality Debate

The phenomenal growth of the Internet is largely a product of how the Internet works. Web surfers can pretty much go anywhere and do whatever they want on the Internet. The "survival of the fittest" governs the fate of new Internet services and applications. The Internet's open and unrestricted platform enables these innovations to be tested, and either embraced or rejected, by Internet users.

This way of doing things has been supported by broadband Internet access providers. The unwritten rule generally has been that these providers operate like common carriers. Subscribers get to use their pipes in an unrestricted manner. Nobody's services or applications are blocked, no Web sites are off limits, and nobody gets any special preferences from the broadband providers.

But now all this could change. Debate is swirling around this question: Should "network neutrality" — the principle, as Vinton Cerf explains it, that there are "no gatekeepers over new content or services" — be codified, or should Internet ground rules move in a different direction?

Broadband providers argue that the current Internet model is not carved in stone, and that they are short changed by the existing scheme. They say they've made large investments in Internet infrastructure, and they should be allowed to implement business models that maximize their return on this investment. They claim to support network neutrality, but they qualify their endorsement with the mantra that network neutrality is "a solution in search of a problem" and does not need to be enshrined in federal laws or regulations. Kerry Knott, Comcast's vice president for government affairs, explained it this way in a recent interview: "It is better to let the market work and, if there are instances that pop up, you deal with them."

Some broadband providers want to supplement market forces with legislation of their own. As Michael Geist, a professor at the University of Ottawa, recently observed, BellSouth and AT&T are lobbying Congress "for the right to create a two-tiered Internet, where their own Internet services would be transmitted faster and more efficiently than those of their competitors."

Broadband providers say they've made large investments in Internet infrastructure, and they should be allowed to implement business models that maximize their return on this investment.

Network neutrality advocates are crossing swords with Internet access providers, arguing that moving away from a common carriage model would clear the path for closed broadband networks that stifle competition, impose costs on consumers, and dampen innovation for new Internet applications.

Professors Tim Wu and Lawrence Lessig cite two reasons to support a regulatory guarantee of network neutrality. First, it would eliminate the risk of future discrimination against services and applications, thus providing greater incentives for investment in broadband applications, content, and services. The *potential* of discrimination itself has a dampening effect on innovation. Second, network neutrality brings fair competition among applications and services, ensuring that those favored by biased network platform providers are not given an advantage. Competitive innovation — marked by the edge-to-edge design of the Internet — is promoted by network neutrality. Wu and Lessig argue that consumers, not network operators, should decide what applications, content, and services succeed or fail on the Internet.

Federal policymakers are grappling with the issues raised by this Internet debate. Last September, the FCC adopted a "policy statement" about how to "encourage broadband deployment and preserve and promote the open and interconnected nature of the public Internet." The non-binding policy statement included these principles: consumers are entitled to access the lawful Internet content of their choice; they're entitled to run applications and use services of their choice (subject to law enforcement needs); and they're entitled to competition among network providers, application and service providers, and content providers.

Meanwhile, network neutrality has become a centerpiece of the telecommunications reform efforts on Capitol Hill. Several different approaches are in the mix as Congress attempts to move forward with a telecom rewrite package. A House Commerce Committee staff counsel predicted in a January press interview that legislation will pass this year, and will be "market-based and market-driven."

What's a reasonable solution to the network neutrality debate? A touchstone for government policymakers should be that broadband providers must not be permitted to leverage their control over Internet platforms to gain advantage through discriminatory or anticompetitive practices. TMCnet Associate Editor Patrick

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Barnard recently pointed out, for example, that "[p]erhaps the most compelling arguments against a second [Internet] tier center on the issue of fairness in competition."

Restrictions on broadband providers' operations make sense because the Internet access pipes are bottlenecks. The phone companies and cable operators who control the pipes argue that their head-to-head competition is all that's needed to safeguard network neutrality. But how likely is it that a duopoly broadband market will produce true competition? A choice between only two providers is not a competitive choice. In some cases, enterprise and mass market users only have one choice for broadband service.

Consumers, not network operators, should decide what applications, content, and services succeed or fail on the Internet.

Internet access providers should be able to recoup their infrastructure investments, and they should be permitted to develop business models that seek new sources of revenue from their Internet infrastructure. But they should not have a blank check to impose restrictions that turn the Internet into a closed network driven by anti-competitive preferences. 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. (news - alert)

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By Hunter Newby

VoIP Peering 2.0 Is Video Peering 1.0

It is quickly becoming widely understood that voice as VoIP is just another application like e-mail. It's a no-brainer, yesterday's news, right? Not such a no-brainer a few years ago though. So, what makes video any different? If the differences in the protocols and compression, such as G.711 versus MPEG4 are set aside, nothing is different but the bandwidth requirements. Video is just another application as well. IP is peered, VoIP is peered, and video is next.

Video peering is the "Wild West" and, just as ENUM within VoIP (define - news - alert) Peering is disrupting the economics of the traditional voice business model, the mere capability of a globally distributed IP network openly carrying video feeds, clips, and feature-length films poses real threats to broadcast television as well as the cable companies. It's no wonder that the RBOCs are racing with lightning speed into IPTV. Don't be surprised when they all begin to peer with each other to deliver video content — not that they're moving anywhere near as fast with VoIP Peering and ENUM. Knowing the players, their vantage point, the revenue models they must protect, and the ones they must attack, sheds a whole lot of light on their motivation.

VoIP as just another application trivializes and compartmentalizes an entire industry that for, 100+ years, employed and fed hundreds of thousands of people. Such is life, such is evolution. VoIP is becoming a component of every type of IP interaction imaginable, starting with online reservations and heading straight in to the audio component of a full-duplex video session. Just when you thought the rate per minute couldn't go any lower, in came flat rate voice. It doesn't end there. Now voice as audio is just another component of video like color versus black and white. When you watch TV, you don't have to pay extra for sound.

With the exception of a few sparsely inhabited places on Earth and those with governments that inhibit development and new economic growth by nationalizing ENUM under a legacy revenue model, there will be no separate business model for the audio piece. It is embedded and a given. After all, most of what will be driving the voice piece in the IP world will be free, open source software. Since this is now becoming clearer, we must revisit the "bandwidth requirements" for video.

Video over IP does not mean video over the Internet in exactly the same way that VoIP doesn't mean voice over the Internet. This is very important and it frames the critical debate of IP Networks versus The Internet as the transport medium. With voice as the main character the debate for the carriers is split. Some want to further utilize their existing ISP investments, while others see the benefits of security and reliability outweighing any incremental costs associated with operating or using a private voice Internet. Many small to medium enterprises have no concept of private VoIP connections to other businesses and still see voice as a service delivered to them by a provider rather than the application that it is, such as e-mail, riding over an extended WAN. For large enterprises there is no VoIP over the Internet debate. The answer is no. Private wide-area network connections carrying the VoIP traffic is the only way to go. Cost is not the concern, whereas security, performance and reliability are. These are all very prudent concerns and each ultimately leads to its own respective, logical conclusions.

What's interesting about the public versus private VoIP debate is that from a capacity supply versus demand perspective it is actually possible to have it in the first place. This is because the most bandwidth conservative CODEC out there, G.711, consumes only a mere 64k for a full-duplex conversation. That means that the Internet can, and does, successfully carry high-quality voice calls. This is due to the underlying ISPs all having OC-192, or close, backbones. Millions of simultaneous G.711s don't amount to much strain within 10 Gig. If we look at the more aggressive CODECS such as G.729 the compression gets better and the number of calls can increase without impacting quality too much.

With video as the main character, the story changes considerably. Those old 192 backbones are no match for real-time video when it comes to servicing a multiplicity of networks and users simultaneously. One network alone could consume a Gig. That tops out the 192 at 10 networks. Simply put, the public Internet is not ready, nor are the ISPs in a position, to accept the several orders of magnitude increase in bandwidth required to support the IP enablement of live broadcast video. This is not to say that canned video in conjunction with a TIVO can't be supported — it can, but even those files sizes are already increasingly putting a strain on what's out there today.

This pending bottleneck limitation is not slowing the pace of development, though. So, how is the issue addressed? Well, peering, of course. Not peering via the public Internet, but rather peering as the ISPs do to CREATE the Internet itself. Video networks (content and plain transport) using dark fiber, waves and Ethernet will all find common physical layer points on the Earth to connect to a common Ethernet switch, establish VLANs between and amongst each other and peer. This is what the ISP's have been doing for a long time, what the VoIP networks are doing now, and this is what the video networks will be doing. It's all a matter of time. IT

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

By Bob Aldrich

Debates on "BITS" Bill Challenge Congress to Ensure Net Neutrality, Cable-Telco Parity

Sometime this year, the Energy and Commerce Committee of the U.S. House of Representatives is likely to report a bill that could reshape the marketplace for Internet Protocol (IP)-based products and services. Although not yet formally introduced when this article went to press, a draft bill circulated last Fall has already sparked debate on the ultimate shape of federal regulation (or non-regulation) of the IP market. By holding hearings on a draft bill, the Commerce Committee leadership retained flexibility to test the waters and make revisions before formally submitting a bill for "mark-up" by the committee. While chances are good for moving the bill through the committee, getting a bill passed by both House and Senate this year is a much tougher challenge.

Committee Chairman Joe Barton (D-TX) wants Congress to set a comprehensive framework for regulation (or non-regulation) of broadband and Internet services. As released last Fall, the draft bill had four parts, known as "titles." The first, Title I, establishes the framework for "broadband Internet transmission services" (BITS), which are essentially any packet-switched services and related facilities used to transmit the services. BITS includes but is not limited to broadband services used to access the Internet. Title II addresses voice-over-Internet-protocol (VoIP) services and related facilities. Title III addresses broadband video services (BVS), which are essentially cable TV-like services that integrate the provision of video programming with Internet access. The final part of the draft, Title IV, contains general provisions affecting all services.

The draft bill seeks to minimize regulation of broadband services and to equalize, more or less, the regulatory requirements affecting various providers of Internet access, BVS, and VoIP. The bill reflects major battles between the cable TV industry, the telephone companies, and other providers of these services to gain the advantages available to their competitors. For example, the Supreme Court recently upheld an FCC decision stating that Internet access facilities provided by cable TV operators are an "information service." As a result, cable TV companies are not subject to FCC regulation as "carriers" when their cable facilities are used to access the Internet. As a practical matter, this meant that cable TV operators offering Internet service do not have to open their access services or "last mile" facilities for use by other Internet service providers.

The role and authority of state and local governments to regulate broadband services would be substantially reduced.

Currently, those same Internet service providers are able to use telephone company facilities to compete with the telephone company's digital subscriber line ("DSL") service. Under the FCC's September 2005 Broadband Access decision, however, the FCC reclassified and deregulated DSL service as an "information service," subject to a one-year transition period. As drafted last Fall, the bill would largely institutionalize the Broadband Access decision, freeing the DSL Internet access vehicle, like cable modem service, from regulation as a common carrier service. Both would be classified as BITS and, except as provided in the new legislation, neither the FCC nor any state could regulate rates, terms, or conditions of service or entry into the BITS market.

Essentially the legislation is designed to ensure that telephone companies have the same freedom to market DSL as the cable companies already have to market Internet access through cable modem.

Similarly, in BVS, the bill attempts to equalize regulatory treatment of broadband video services that are integrated with Internet access. The idea here is to regulate other providers of BVS in a manner similar to cable TV. For example, in order to provide a BVS, a telephone company could not rely on its existing franchise to provide telephone services and simply upgrade the facilities to offer the BVS, as the telephone companies contend they are currently able to do. The draft legislation provided that a telephone company would have to go get a BVS franchise from the authorities in the same manner as a cable operator (although both cable and telephone companies would be subject to a much more streamlined franchising process than they are today).

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The draft legislation also tries to provide regulatory parity between providers of VoIP services and providers of traditional telephony. All VoIP service providers would be subject to the same regulatory regime. VoIP services could be subjected to universal service funding obligations. Subscribers to VoIP services would have many of the same rights as subscribers to conventional telephone service, such as number portability. But the provision of VoIP service would be free of traditional common carrier regulation.

VoIP (define - news - alert) service providers also would have to be able to provide 911 and E911 services and to access the E911 infrastructure in accordance with FCC requirements. In general, the E911 provisions of the draft bill track current FCC requirements. Another bill, specific to E911, however, has been introduced in the Senate and could move forward on a separate track.

Under the draft legislation, the role and authority of state and local governments to regulate broadband services would be substantially reduced. Virtually all IP-based services are declared to be in the interstate jurisdiction, where the authority of the FCC to regulate the services would be quite limited. The requirements for entry as a provider of any of the services would be made much simpler. In general, all that is required for entry to provide the services is the filing of a "registration" statement with the FCC and a few formalities.

The draft would have a significant impact on the enterprise market. While reducing the regulatory burden for new entrants to the market, the bill also would lighten the load for established dominant firms like the Bell companies. As a result, the draft legislation would institutionalize and perhaps extend the impact of recent FCC decisions that threaten to make it more difficult for competitive broadband service providers and manufacturers to enter or grow in the market.

A key area of concern is "network neutrality." The most recent draft of the bill does mandate open access to broadband facilities. Under an exception, however, Bell companies and other providers of broadband facilities could give routing priority and other forms of "enhanced quality of service" to their own services, including video services. Competitors are alarmed that these exceptions could pave the way for a two-tiered Internet in which the dominant firms are able to offer premium services with built-in advantages.

In contrast to the ways whereby regulation would be reduced, requirements for broadband products and services to be "accessible" to people with disabilities would seemingly be made stronger than the existing requirements for circuit-switched telecommunications. Rather than requiring equipment and services to be accessible if doing so is "readily achievable," they would have to be accessible unless the manufacturer or service provider "demonstrates that taking such steps would result in an undue burden." Although the exact legal impact of the language is unclear, it appears to shift to the manufacturer or provider the burden of justifying why it is not cost effective to provide access.

Conclusion

The legislative process is still only a few steps beyond the starting point. There are different approaches under discussion in the Senate and even some more narrowly targeted bills circulating in the House. A lot will happen before legislation is adopted, and it will be a difficult challenge for Congress to actually enact legislation in this election year. All participants in the IP market, however, should remember to keep an eye on the speed and direction of the federal legislative process. IT

Bob Aldrich is a telecommunications law practitioner at Dickstein Shapiro Morin & Oshinsky, Washington, D.C. Aldrich represents the Enterprise Communications Association (ECA) and other competitive telecommunications firms and organizations. For more information, please visit the organization online at http://www.encomm.org. (news - alert)

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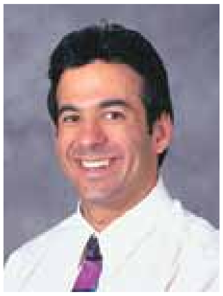



By Rich Tehrani & Max Schroeder

IP Communications Transforms Business Continuity Planning

How IP Communications Pays for Itself Today and Pays Again When Disaster Strikes

Recent and frequent headlines stress the need to better prepare for adverse events. Even so, the idea of investing in continuity planning can still be a hard sell. A key part of that challenge is dispelling the myth that it will be an expensive project that may never be needed. Many enterprises and government groups turn to their local resellers for guidance and the resellers, in turn, look to their vendors. The following information is a condensed version of a paper provided by Forsythe Technology, Inc., to their resellers and is a good example of vendor marketing support.

While a single technology is never, in itself, a business continuity solution, one technology that stands out is IP Communications (IPC) including VoIP telephony. IPC also drives short-term benefits by improving an organization's day-to-day business processes and operations. This is a key factor, since is allows a company to recoup its investment even if a disaster never transpires. In fact, the cost reduction benefits of IPC are driving the current wave of IP telephony deployment, which, in 2005, exceeded the traditional legacy phone (TDM) equipment being shipped. The primary result of this shift will be a fundamental change in the way business continuity works.

The initial capital outlay for the deployment of IP telephony is typically justified by a reduction in ongoing operating costs as compared to the cost of maintaining a traditional TDM (define - news - alert) phone system. However, the savings are magnified when factoring in business continuity. An IPC infrastructure is uniquely designed to be implemented as a geographically dispersed architecture. The benefit is to reduce the likelihood of a business interruption by eliminating single points of failure and enabling remote recovery with seamless fail-over between locations. A bonus is that recovering a single, converged network typically takes less time than recovering separate voice and data networks, further decreasing the impact of any potential interruption.

An implementation is only as successful as its systems management and user training.

Keeping the technology up and running is vital, but people and processes must also be maintained. Again, the same IPC technology features that enhance day-to-day business productivity also speed the return to productivity. For instance, when localized flooding temporarily closed an IP contact center for a major online retailer, the company was able to seamlessly reroute customer calls to an alternate contact center not affected by the flooding. The outage was transparent to callers, and the company was able to keep its major source of revenue operating.

Another IPC component — unified messaging — has been documented to increase employee productivity by 30 to 60 minutes per person per day under normal working conditions but also enables access to voice messages and faxes even if phone systems are down and the office is closed.

Integrated conferencing increases productivity by enabling real-time collaborative work via a combination of audio, video, or Web-conferencing, and instant messaging (IM) — across multiple locations. This feature also helps overcome logistical challenges in times of emergency by supporting efficient communication and collaboration among remote workers.

Extension mobility allows users to log onto any networked telephone and have calls to their extension routed to any phone from which they are working including cell phones. Soft phones further broaden extension mobility since employees can plug in anywhere with broadband access, and work from their "normal" business extension and access all the information the office phone system provided, including directory or customer information.

The end-user features mentioned above are available to any organization, but for companies with TDM-based call centers,

implementing an IP contact/call center is even more critical. For example, by dynamically balancing call loads between multiple, geographically separate call centers, IP-based contact/call centers both enhance the customer experience plus the technology can automatically re-route calls to other agents at other contact/call centers if one or more of the locations is off line for any reason.

Although Implementing IPC can reduce cost and risk, it also introduces the potential for new risks that need to be managed. This should be carefully addressed during the planning stage. Organizations must account for additional security concerns stemming from the increased ability to forward and remotely access voicemails, e-mails, and other corporate data. Frankly, it is more of the "same old, same old." In other words, make sure your employees know the security and privacy policies and obey them. It is also important to review compliance regulations, taking into account any telecommunications laws that may newly apply once certain data is linked to an IPC network. TMCnet.com and the ECA (http://www.encomm.org) are great resources for this type of information.

Finally, one of the most avoidable causes of business interruption and decreased productivity is lack of preparedness. The redundancy of the IP network is irrelevant if redundancy isn't also built into the server and storage elements of the communications infrastructure. Beyond that, an implementation is only as successful as its systems management and user training.

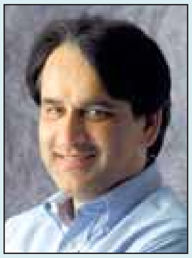
The end result of deploying IP Communications will be a transformed business continuity model that reduces cost and risk while improving productivity on a day-to-day basis. IT

Rich Zimmermann is Director of Network Solutions and Michael Croy is Business Continuity Practice Manager with Forsythe Technology, Inc. Zimmermann is also Chairman of the Board of the Enterprise Communications Association (ECA).(news - alert)

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

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

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



Sonus' Hassan Ahmed

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

I recently had the real pleasure of interviewing Hassan Ahmed, CEO of Sonus Networks. (news - alert) The company was around from the beginning and has been a true pioneer in VoIP. There are many companies today competing head to head with Sonus who downplayed VoIP and the products Sonus sold in the late nineties. Now Sonus is a benchmark for which many other companies aim.

Sonus was a Wall Street darling with a valuation well over $10 billion at its peak; it also crashed to lows that put it in line with most other VoIP (define - news - alert) companies in the 2001-2004 downturn. During that downturn, many industry publications and analysts suggested that people won't buy from Sonus, that they will only buy from large telecom equipment manufacturers — the incumbents, if you will. If you believed the word on the Street, you would have assumed that Sonus was doomed. Much to the contrary, Sonus was one of the few that weathered the storm and, just like the VoIP industry in general, bounced back like a SuperBall off concrete, surprising everyone at once.

Here is my long overdue interview with Hassan Ahmed.

What can you tell me about the down years, 2001-2003?

Ahmed mentioned that Sonus was one of the last companies into the meltdown and the first out. The company's big focus at the time was how to keep from succumbing to the downturn — his specific words were, "Powerful downdraft." He went on to say that the company earned its stripes on the way down; he is most proud of his team and the company, since many others didn't make it through. They saw the bottom with a 20-cent stock. They also thought the end of '02 was going to mark the end of the downturn.

At a certain point, they realized their growth would come at expense of something else, that the overall pot was not going to increase. They had to be able to demonstrate short-term value and focused their products on that. They had to make sure that, for service providers, it was worth taking money from legacy projects and applying it to new ones.

Sonus realized it was not a field of dreams, where you tried every new technology. Furthermore, they realized that VoIP was becoming global and all carriers were moving to VoIP. More specifically, they zeroed in on Europe and Japan. The company decided to focus on a distributed architecture and, according to Ahmed, they didn't make naïve bets.

The company had decisions to make regarding its target audience. Ultimately, the decision was a focus on supplying large public infrastructure. They didn't want small "mom and pops" and CLECs: They wanted the large service providers.

This focus, of course, led Sonus to focus on scale and reliability, as they realized these weren't things they could bolt on later. This customer-focused decision helped reinforce the decision to build distributed systems and related architectures

These decisions subsequently drove large performance increases. For example, Sonus' softswitch could handle six million busy-hour call attempts while others were in the few hundred thousand range.

Hassan continued to note that the industry is now coalescing around IMS. "It's all about SIP and distributed networks. These were bets Sonus made early on, and they were bets that paid off nicely. We have been hardening and developing this technology for eight years. For us it is just software upgrades."

What about industry comments about service providers only buying from incumbent providers?

Hassan started by telling me that popular thinking was that the class 5 market would be bigger than the class 4. Many companies decided to focus on class 5 replacements. They thought that that incumbent circuit switched providers have thousands of class 5

switches and these would grow packet interfaces. Migrating these old switches to VoIP was going to be the right answer, as this is where the money was.

"There was no question class 5 was bigger in ports, but the world doesn't change overnight." Ahmed said. "You have to ask what the right way to do it is. The core has to evolve before the edge. Sonus embarked on a path focusing on dominating the core first."

They also bet that class 4/5 divide will have different switches, since the ones on the edge terminate in copper pairs. In the IP world the connection to the network can be disaggregated from the delivery of service, Ahmed said. You wouldn't just morph the switches in the network. The difference was software, not hardware.

The fact that you have thousands of circuit switches is not an advantage — that is not where you deploy the switches in the IP world. Also, look at the wireline world: lines were declining. Morphing circuit switches didn't make sense

How does SIP play into your plans?

SIP is, in their view, a very important protocol from a variety of perspectives. It is a protocol they have honed and field-hardened for 15 billion minutes a month. It is used for Peering IP networks at VoIP level. Sonus used Sip-T early on to allow two networks to communicate with each other.

Sonus felt one of the big values of SIP was that it could open up the service model that once was closed — you used to have to go to a switch vendor for custom programming. They further provided the Open Services Partner Alliance or OSPA.

"This has enabled a lot of creative companies to develop really interesting applications," according to Ahmed.

Sonus feels the IMS architecture is built on technology they pioneered and the choices they made early on; the rest of the industry has now coalesced around. In the beginning, the industry was all about cost savings, not separate networks. Now, cost is table stakes. The wireless and wireline opportunities are now about service and service convergence, according to the company.

Hassan was very passionate when he said, "The goal is to empower next gen consumers with the way they want to communicate." He looks at his children to see the next generation of consumer. In them, he sees lots of IM sessions that lead to VoIP as well as lots of collaboration.

Consumers want to get services once they have subscribed, regardless of how they connect. The common service model needs to be agnostic in terms of how you join the network. That is what IMS is all about.

Creating services and driving innovation is a big part of Sonus' focus. The IMS platform is about taking open service architecture to the next level. In weeks you can build brand new applications.

Ahmed continued, "In the wireless arena, VoIP is now being adopted by operators. When wireless operators have broadband wireless access, the last mile becomes IP. The way you build wireless networks is IP instead of circuit-based MSCs."

"IMS plays a prominent role in the future as we see it," Ahmed exclaimed.

Where is the growth in the world?

VoIP started in the United States. Sonus then invested in Europe, Japan, and other parts of Asia. Their Japan investment was very good, for it became the second strongest Sonus market. Customers tell them they are the de facto VoIP standard in Japan. They work with wireline operators (Softbank, NTT, Softbank, JCOM) and now wireless providers.

Europe is a different story. There have been a few false starts and, early on, there were few decisions being made. After the slow start, however, Europe is now growing nicely and Sonus recognized better performance there in the last quarter. Ahmed thinks availability of licenses is what will keep a ceiling on growth in this area.

What about China and Huawei?

They haven't seen Huawei outside Asia — not in the VoIP market, anyway. Rather, they have seen the large incumbent circuit switch providers.

What about the competitive environment?

From a competitive standpoint, Sonus sees Lucent and Nortel in the U.S. and Alcatel, Ericsson, and Siemens in Europe. Finally, Cisco is not as focused in the VoIP carrier space as the other players.

Can you stay independent and compete with all these players?

Ahmed replied:

"We are five years in and this is a 15-year opportunity. Innovation counts. All operators are coalescing on a common approach that is increasingly friendly towards technology we pioneered.

Sonus is a bonafide provider to large networks. Our focus is on building our company and we don't worry too much about others and if they need to be part of them. We are a leader and can continue to build on that leadership."

What is biggest impediment to your growth?

The company's growth is a function of development of the global market. It's not about early adopters anymore — all providers have embarked on VoIP. Different opportunities arise at different times and, as the market develops, so will Sonus.

Ahmed points out that anyone can become a service provider, which has enabled a set of operators like Yahoo! and Google to become providers

He went on to say these are interesting times and these developments enable more competition, as you don't need to own the last mile.

It was an honor to have this long overdue interview and it is exciting to see one of the early adopters in our space doing so well. As the IMS market matures, Sonus will no doubt continue its position as an industry leader and continue to supply the next generation of converged communications solutions. IT

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Echo Cancellation and VoIP

By Greg Galitzine

There are a number of impediments that can affect the overall quality of a voice over IP (VoIP) call, including echo, latency, delay, and others. Each of these can degrade the perceived quality of a voice call to various degrees ranging from being slightly annoying to downright unacceptable.

This problem is not unique to VoIP, however through years of trial and error, and simple experience, the voice quality issues facing the PSTN (define - news - alert) have been addressed through network optimization and better network management. As IP networks converge with the PSTN, these voice quality degradations need to be dealt with.

So, what is echo anyway? Dictionary.com offers this standard definition: "Repetition of a sound by reflection of sound waves from a surface." In telecommunications parlance, and as defined by Tehrani's IP Telephony dictionary, echo is "...a type of transmission impairment in which a signal is reflected back to the originating source."

Simply put, echo is the sound of the speaking party's voice returning to their ear via the handset or headset speaker. Lest we think all echo is bad, it is common practice to allow a little echo (with delay in the range of 25 ms) so that the speaker can hear their own voice. Studies have shown that this is a reassuring sound. But aside from some minor allowances, echo can wreak havoc on a VoIP conversation by degrading the conversation to unacceptable levels.

According to the VoIP Wiki at voip-info.org, there are three potential sources of echo in a standard system. From the Caller's perspective, these are:

• Within the caller's telephone; a certain amount of the signal from the microphone is fed straight back to the earpiece.

• At the hybrid at the callee's end. An improperly balanced hybrid won't correctly filter out the entire transmitted signal, and will reflect some of it back down the other half of the trunk. (A Hybrid is a device that combines transmit and receive signals from two pairs of lines to one single pair).

• At the handset at the callee's end. If the callee isn't holding the handset against his head, or if the handset is poorly designed, it's possible for the microphone to pick up the sounds coming from the earpiece, and reflect the audio back down the line.

Likewise, according to the Wiki, there are three ways of dealing with echo:

• *Elimination at the source*. This would mean going out and making sure all the hybrids in a network are tuned and properly balanced.

• *Echo suppression*. A simple voice activated switch which turns off transmission from the speaker to the listener whenever the speaker is silent. Negative side effects of this approach include voice clipping and choppiness, which may result in voice degradation, the likes of which the solution was meant to avoid in the first place.

• *Echo cancellation*. A mathematical approach to subtract exactly the right portion of the transmitted signal from the return signal to eliminate the echo.

This last approach is often the most feasible and practical.

Echo Cancellation

Echo cancellation is the process of extracting an original transmitted signal from the received signal that contains one or more delayed signals (copies of the original signal). Echoes may be created in a baseband or broadband signal. When echoes occur on an audio baseband signal, it is usually through acoustic feedback, where some of the audio signal transferring from a speaker into a microphone. When echoes occur on a broadband signal, it is usually the result of the same signal (such as a radio signal) that travels on different paths to reach its destination. In either case, echoed signals cause distortion and may be removed by performing via advanced signal analysis and filtering.

In order to properly determine the signal echo, it is necessary to measure the echo. Among the characteristics that need to be gauged is something referred to as echo return loss or ERL. ERL is the amount of attenuation or loss of a signal that is returned to the originating speaker. Essentially, this is how much lower the echo audio sounds compared to what was originally sent. ERL is critical to effectively canceling echo, because often an echo canceller cannot deal with echo that has not been attenuated to at least some degree.

Companies like Sangoma are actively studying the various impediments that can degrade voice quality. Through the use of dedicated processing power and finely tuned telephony cards, and the efficient management of system resources, Sangoma is able to design systems that deliver high-quality reliable voice. IT

Greg Galitzine is the editorial director of Internet Telephony.

The Optimization of Soft PBX Systems

By David Mandelstam

On any PC-based telephony news group, one of the questions that comes up continually is, 'How big a PC do I need to carry X number of calls?' Responses from the users eventually resolve themselves into variations of 'What precisely are you trying to do?'

A popular solution is to play it safe: Install a state of the art multiple processor machine or perhaps several of them. These machines will typically run at less than 10 per cent capacity. The solution based on simple oversizing is not a bad one for a single installation given the fairly economical cost of PC horsepower. However, if this is something you will do more than once, perhaps as part of your business or part of a multi-installation rollout, the cost of unused processing power becomes significant. Getting it just right can save a lot of money, but you do not want to undersize your system.

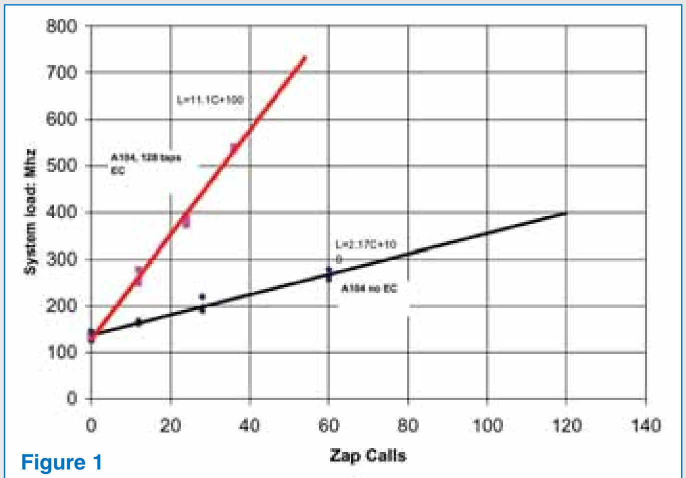

Figure 1

However, getting it wrong on the small side will result in sharply degraded voice quality and reliability. You will hear ticks and pops or distortion, and calls will terminate unexpectedly. At very high system loads, voice will be heavily distorted and the system itself becomes unstable. For this reason, most installations are run at system loads below 60 percent of capacity.

Sangoma has been taking a look at system loads for this part of the soft PBX for some time as we have a responsibility, i.e., the interface to the Public Switched Telephone Network (PSTN) as handled by internal PCI telephony cards.

The PSTN load consists mainly of:

- The DMA load of moving data in and out of the PCI bus.
- The interrupt routines needed to control the transfer of voice and signaling data.
- Loads related to software-based echo cancellation.
- Housekeeping routines required to monitor and control the digital T1/E1 or analog front-end devices.
- Per-call related loads such as data transfer and signaling.

A typical load curve is shown in Figure 1.

The intercept and slopes of the lines depend to a certain extent on the characteristics of the processor used. As a general rule we find that for instance in the case of a single quad port T1/E1 card, the quiescent CPU usage with no call activity consumes in the region of 100 MHz on average. This is about three percent of a 3GHz processor. The quiescent load includes the DMA time of the PCI bus, which is very small, and the interrupt and housekeeping loads. The quiescent load is small, but not negligible, and there is scope for additional system optimization.

You can see that by far the dominant load is software echo cancellation, which causes a rapid increase in system load as the number of calls increases. The loads shown are representative of a 128 tap (16ms) software echo canceller. There are several software echo cancellers included in zaptel, but the load appears to vary only slightly between them.

It has been estimated by Intel that a complete echo cancellation routine, optimized for PC processors, will consume about 30 Mhz per active call. The software echo cancellation routines that are part of the zaptel drivers used by the Asterisk system all consume only about one third of that. This is a very significant reduction, and has been accomplished by the use of highly optimized routines, by eliminating the non-linear processing part of the echo canceller, by reducing the rate at which the echo canceller trains and by restricting the length of echo handled to 128 taps (16ms).

Notwithstanding the compromises due to optimization, the better software echo cancellers in zaptel can provide perfectly acceptable echo cancelling, especially if tuned by adjusting audio gain levels and using zaptel's ingenious echotraining feature.

Without software echo cancellation loads increase with the number of calls much more slowly, at the rate of about 2.2 MHz per call.

For a 3Ghz system without an echo canceller, the TDM load is small enough that a full T3 of 672 channels could be handled, while still staying below the threshold of 60 per cent of capacity. Any machine operating at around 1GHz would be capable of supporting 120 calls (i.e., four full E1 spans) on this quad-port board, as long as there was no echo cancellation load. With software echo cancellation enabled, even a 3GHz system would be maxed out at a load of 120 calls.

Of course, the TDM load is only one of the components of the load on a soft PBX system. Dominating everything is the load of transcoding or voice compression, if present. While there is not a great deal of data on measurement of transcoding loads, anecdotally it appears that transcoding of G.711 (i.e., normal U-Law or A-Law voice) to the G.729 compressed voice standard seems to consume over 30 MHz per call.

For the same reasons that graphics handling has moved into dedicated graphics processors on video cards, notwithstanding the enormous computation power of modern PCs, it has made sense to move the CPU intensive task of echo cancellation from the PC to dedicated hardware such as used on Sangoma's A104d. Not only does this reduce system load, but there are no compromises, so that the system can deliver true carrier-grade echo cancellation with extended echo tails. The same argument suggests that the voice compression/transcoding loads should also be shifted onto dedicated hardware. This would leave the PC free to handle the less heavily routine tasks, and that will lead to smaller and more reliable telephony installations. IT

David Mandelstam is President/CEO, Sangoma Technologies. For more information, please visit the company online at http://www.sangoma.com. (news - alert)

WiFi Enabled Phones

VoIP technology is rapidly being adopted by enterprises, but employees — not to mention consumers — are on the go more and more, moving between desks, between conference rooms, and between buildings, not to mention cities. Which is responsible for much of the success cellular carriers have providers have enjoyed.

But most enterprises and SMBs are now looking to cut costs, and one way they hope to do that is by incorporating mobile devices into their IP-based communications systems — WiFi phones, that is. WiFi handsets liberate users from the constraints of traditional wireline handsets, allowing them to make and receive calls anywhere there is a WiFi hot spot — at home, at the office, around town, in an airport lounge, at local coffee boutiques, and in most hotels. It allows users to make or receive phone calls as long as they are within range of IEEE 802.11b/g wireless access points.

With that in mind, this month's product round-up features a selection of WiFi enabled phones. Some have the look and feel of a traditional cordless home phone, others resemble a cellular flip-phone. And still others fall into neither class, but are part of a new breed of ultra-small PCs, allowing for ultimate mobility.

Axcess Communications G1000

http://www.axcessasia.com

The Axcess Communications' (news - alert) G-1000, a SIP-enabled WiFi phone, is nothing if not convenient. It looks and feels much like any standard cellular flip-phone: It is small, lightweight, and fun to use.

The G-1000 comes with many of the same features customers are used to in their handheld devices, from a large variety of ring tones, to a call log, to the optional voice recorder.

This 802.11b-compliant phone also offers access to email, power saving features — which can increase battery time — and an earphone jack for added privacy. The unit weighs a mere 115 grams (~¼ pound).

Calypso Wireless C1250i

http://www.calypsowireless.com

Calypso's (news - alert) sleek new C1250i video phone is great for work or play — it is a WiFi/GSM-GPRS mobile phone capable of video conferencing over WiFi. It will no longer take several minutes to simply send or receive a single still picture or download a graphic intensive Web site. The C1250i connects you to the Internet at blazing broadband speeds of 11 MB per second — up to 200 times faster than connection speeds available from most U.S. wireless carriers. Or 200 times faster than most DSL or cable-modem speeds available.

C1250i's 30 frames per second video capabilities make phone-to-phone video conferencing a breeze. No more boring still images, just honest to goodness movie-quality video! And, with its built in digital camera, C1250i can also capture all of those special moments and share them instantly with friends and family as they happen.

Calypso has achieved this by building the world's first mobile phone that works on both traditional cellular/digital frequencies and the exciting new WiFi frequency meaning only Calypso phone users can connect to the Internet at these super fast speeds.

But what makes this phone unique is Calypso's patented ASNAP technology — it provides a unique system for carriers to identify and authenticate mobile users. ASNAP will create revenue sharing opportunities for companies who offer services like two-way, real-time video conferencing. Mobile Carriers will also save money and network bandwidth by switching users seamlessly to WiFi networks when near a shared hotspot.

Cingluar 8100 Series

http://www.cingular.com

The Cingular (news - alert) 8100 series phones will keep you connected while on the go. They feature a full QWERTY "slider" keyboard, the latest Windows Mobile 5.0 operating system, integrated WiFi, Bluetooth support, quad-band international coverage, and EDGE, the largest high-speed national wireless data network in the U.S. Cingular's nationwide EDGE network is available in 13,000 cities and towns and along nearly 40,000 miles of major highways.

The Cingular 8100 series Pocket PCs come in two versions, the camera-less 8100 and the 8125 with integrated 1.3MP digital and video camera. With the 8100 series Pocket PCs, mobile professionals can easily integrate with leading corporate and personal email applications integrated on the phone.

The Cingular 8100 series also will appeal to customers in the healthcare, field service and manufacturing sectors, particularly because of the devices' WiFi capabilities that allow access to on-campus, 802.11-based wireless local area networks.

Powered by Microsoft's newest mobile operating system, Windows Mobile 5.0, the 8100 series will be upgradeable to support the Microsoft Messaging and Security Feature Pack featuring direct push e-mail technology. Users will also benefit from Windows Mobile's persistent memory storage as it retains information even when the device's battery is depleted.

The phones also provide a rich multimedia experience to enjoy music and video, featuring an integrated multimedia suite with WMP 10 Mobile and stereo headset jack, The devices support multiple audio formats, as well as MPEG-4 video streaming. A mini-SD expansion card slot allows the user to add memory for storing music, games, photos and video.

Clipcomm CWP-100

http://www.clipcomm.co.kr

Clipcomm's (news - alert) wireless IP phone, CWP-100, is a low-priced WiFi portable terminal with handy and sleek design. Leveraging field-proven VoIP and wireless technology, CWP-100 is compatible with IEEE 802.11b standard and interoperable with major SIP/H.323 call servers, IP-PBXs and various terminals. The CWP-100 is very easy to use and efficiently managed thanks to its integrated provisioning system that has been applied and proven by Clipcomm's field-deployed VoIP CPEs.

As a VoIP terminal, CWP-100 offers many essential features, such as easy-to-use auto-provisioning, dynamic jitter buffer and firewall/NAT traversal. The CWP-100 can handle both SIP v2 (RFC 3261) and H.323 v4 protocol suites. When a customer requires other VoIP standards or application specific modifications, the CWP-100 can be easily customized by field-upgrading its firmware.

Longer talk and standby time are essential for any battery-powered voice terminals. Through a power management scheme optimized to its platform as well as the standard IEEE 802.11b power saving mechanism and highly optimized DSP software, Clipcomm's CWP-100 has industry leading performance for power consumption.

CWP-100's integrated dynamic jitter buffering mechanism as well as the enhanced QoS functions such as DSCP (Differentiated Services Code Point), IP ToS (Type of Service) and IEEE 802.1Q VLAN tagging provides users with continuous and well-managed voice quality in rapidly varying wireless office network.

Linksys WIP 300

http://www.linksys.com

The newly announced Linksys (news - alert) WIP300 WiFi phone enables high-quality VoIP service through a Wireless-G network and high-speed Internet connection using SIP v2 to connect.

This phone comes with all the features and functionality users will expect. It will connect at home, your office, or at a public hotspot, allowing you to make low-cost phone calls through your Internet Telephony Service Provider.

This phone comes with all the standard security measures — 128-bit WEP, WPA and WPA2 have all been integrated into the WIP300. You'll be able to access your POP3/SMTP email, as well as send and receive text messages via SMS. Other features include 16MB flash memory, a 1.8-inch color screen, and USB recharging. Some nifty audio enhancement features, like echo cancellation, have been thrown in for good measure.

The WIP300 WiFi phone operates in the 2.4GHz band, supports IEEE802.11 b/g, and the latest VoIP SIP protocol. The WIP300 also has a slightly fancier "brother," the WIP330.

Motorola A910

http://www.motorola.com

(quote -news - alert) Streamlined and distinctive, the Motorola A910 delivers a full suite of intuitive productivity tools to seamlessly manage your busy life whether at work or vacationing in Belize. This advanced mobile device can serve as your main means of communication providing everything you need and want in a single handset.

With sophisticated voice features and WiFi connectivity via UMA technology, all you need to do is speak your command and the handset takes care of the rest. The A910 does it all — not only can you listen to MP3 songs, create a personalized soundtrack to simultaneously play along with a selection of your favorite photos that you took with the 1.3 Megapixel camera with Lumi-LED Flash, you can also manage email and stay mobile.

Utilizing a balanced Linux-Java operating system and WiFi connectivity, the Motorola A910 surpasses its predecessors with user-friendly features — everything from text messaging to email management. Experience efficient text messaging with prediction features, intelligent addressing, and seamless language switching, or organize on-the-go with MOTOSYNC™ for managing emails, calendars, and contacts. To fit your fast-paced life, the superior J2ME performance will allow you to do it all with precision and speed.

NEC DTERM PSIII

http://www.nec.com

Employees need not be out of touch just because they have stepped away from their desks. With NEC's (news - alert) Dterm PSIII, today's on-the-move work force can stay in constant touch with customers and colleagues. They can walk freely around their workplace while on a call — with no "dead zones." NEC's Wired for Wireless solution provides continuous coverage throughout a multi-storied building or across a multi-building campus.

NEC's solution provides employees with a fully featured wireless extension to their desktop telephone. Access to advanced telecommunication features, such as voice mail, call forwarding and conferencing, provides the freedom and convenience of a mobile phone, with the advantages and features of a desktop handset.

The DTERM PSIII handset has been uniquely designed to support the professional's schedule and the

special needs of people on the run. With up to 4 hours of continuous talk time and 300 hours of standby time, the Dterm PSIII practically eliminates work interruption to change batteries.

This unit offers portability, clear digital signal quality, and has the same features of a desktop phone. Weighing less than 4 oz., and with 6 hours of continuous talk time and 300 hours on standby, the Dterm PSIII phone has been uniquely designed for people on the run. The Dterm PSIII provides the freedom and convenience of a mobile phone with the advantages and features of a desktop handset

The unit provides continuous coverage, allowing users to walk freely around their workplace while on a call. It also offers audio quality indistinguishable from a desktop phone.

Net2Phone XJ200

http://www.net2phone.com

(news - alert) The 802.11 b/g-compliant XJ200 WiFi handset means that users no longer need to access a wired network to benefit from VoIP calling. Callers can save in any number of WiFi network settings.

Larger businesses are increasingly using WiFi networks to extend their wired networks to off-site/telecommuting employees in home and satellite offices. With this solution, corporations can reduce the communication costs associated with establishing remote offices. And small businesses rely on the flexibility of WiFi networks to meet their ever-changing infrastructure requirements.

Users can utilize their home WiFi networks, or callers can take advantage of the rapidly growing number of free public hotspots to make and receive low-cost calls when they are away from their home or office.

With the XJ200, calls can be made through WiFi gateways and access points that comply with either the 802.11b or 802.11g standard. Both standards apply to WiFi LANs operating in the 2.4 GHz spectrum.

The XJ200 enables both inbound and outbound calling and allows users to choose telephone numbers from the U.S., U.K., Canada and the Netherlands. The XJ200 also features hotspot scanning capabilities, revealing SSID, signal strength, and WEP. The phone will automatically connect to available WiFi networks based on designated profiles and signal strength.

Nokia 9500 Unlocked GSM Communicator

http://www.nokia.com

(news - alert) The Nokia 9500 Communicator offers you a full set of business-critical applications, fast network connectivity and large memory storage — so you can keep in stride with your work on the go.

With fast data connectivity with Wireless LAN, browse the Internet in full color on a wide, easy-to-view screen. Work with office documents (not just email with attachments and memos), but presentations and databases too. The Nokia 9500 Communicator offers high-capacity built-in memory — 80 MB, plus a MultiMediaCard (MMC) slot for even more storage space.

Keep your Personal Information Management data in synch and up-to-date. With Nokia PC Suite and OMA Data Synchronization, you can easily exchange data between your Nokia 9500 Communicator and a compatible PC.

Print photos and documents from your Nokia 9500 Communicator straight to a compatible printer, without wires or computers in between. With Bluetooth wireless technology, you can connect your Nokia 9500 Communicator directly to compatible printers like the HP Deskjet 450 series printers.

This Nokia 9500 has tri-band operation: EGSM 900, GSM 1800/1900 and EGSM 850, GSM 1800/1900 networks in Europe, Africa, Asia-Pacific, North America, and South America where these networks are supported.

Samsung WIP-500M

http://www.samsung.com

Samsung's (news - alert) entry into the WiFi market is a natural extension of its success in the cellular market. The WIP-500M, Samsung's first venture into the space, has a sharp, innovative design and is lightweight delivering superior voice quality.

It features a menu-driven graphical display, which provides easy access to system features. The handset also comes with a desktop charger, headset, and holster for added convenience.

The WIP-500 (as well as Samsung's WIP 600M) offers users the same functionality they would get from any other office or wireless phone — from time-based features to call processing functionality to voice mail.

The only difference is that you can walk and talk at the same time. The phone operates via WLAN technology and is fully compatible with Samsung's IP-enabled office phone system. In fact, the Samsung system can support as many as 240 wireless handsets, which means that, as your wireless needs expand, so, too, can your system. Samsung's system features the capacity to give priority to voice IP packets, thus ensuring high quality communication.

SpectraLink NetLink Wireless Telephone

http://www.spectralink.com

(news - alert) NetLink Wireless Telephones operate on a converged voice and data network to reduce costs and simplify management while significantly improving employee mobility, responsiveness, and productivity. NetLink Wireless Telephones provide the richest functionality while integrating with the broadest range of enterprise applications and networks.

By operating on industry standard 802.11b networks from a wide variety of vendors, NetLink Wireless Telephones are truly converged devices that blend the cost effectiveness of the data network with the proven reliability and funcationality of traditional telephony systems.

SpectraLink's NetLink Wireless Telephones integrate with market-leading PBX and IP-PBX platforms from around the globe thus leveraging the functionality and investment already made in the facility's PBX.

The newest family of SpectraLink handsets, the NetLink e340, h340, and i640 Wireless Telephones, is designed for a broad range of enterprise applications, from general office to industrial environments. These compact handsets offer a rich set of features including a high-resolution graphic display, menu-driven functions, and messaging capability, all within a light, ergonomic design. Push-to-talk functionality is available in the industrial-grade NetLink i640 Wireless Telephone for broadcast communication between employees, eliminating the need for two-way radios or walkie talkies. Carrying options and accessories are available to suit users in a wide variety of applications.

UTStarcom F1000

http://www.utstar.com

(news - alert) The F1000 handset provides all the features and functionality of a VoIP terminal adapter together with a standard cordless telephone, but has the advantage of letting the user talk from any available WiFi access point instead of just the base that they have at home.

The F1000 features a sleek form factor similar to today's cellular phones, has increased talk and standby times,

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and is priced significantly lower compared to other models on the market today.

Unique power management features, combined with an energy efficient design, give the F1000 handset a standby time of up to 80 hours while typical WiFi phones have a standby time of about 8 hours. At three to four hours, talk time is 6 to 8 times longer than the 30-minute talk time of other WiFi phones. And when the battery runs low, the F1000 recharges in only 2 to 3 hours.

The F1000 WiFi portable handset supports a wide variety of SIP-based VoIP features and functions. Service providers can offer high-value call features, such as three-way calling, call waiting, and call transfer, and more, based on the capabilities of their call servers. The F1000 also enables voice processes, including comfort noise generation, voice activity detection, and echo cancellation, as well as IP protocol features such as Real-Time Transfer Protocol (RTP), Session Description Protocol, Dynamic Host Configuration Protocol (DHCP), and Point-to-Point over Ethernet (PPoE) authentication.

Viper Networks vPhone

http://www.vipernetworks.com

Make inexpensive VoIP phone calls at any location with 802.11B WiFi access over Viper's (news - alert) Global network. The Viper Networks WiFi vPhone is a next generation intelligent IP Communications device, which combines SIP-based VoIP communications together with WiFi.

The WiFi vPhone Phone is great for branch offices as well as SoHo and Hot Spot application. Make inexpensive VoIP phone calls at any location with 802.11B WiFi access over Viper's Global network.

Use your vPhone to call any phone in the world at a fraction of the cost of traditional telephones, all with amazing call sound and clarity. Manage, store and call all your contacts quickly an easily with the included vPhone Dialer software. Easily view your past call activity and purchase additional minutes from your convenient Viper Networks online account. The vPhone connects and installs in minutes with the included vPhone Software and USB cable.

On the go, owners can use the WiFi vPhone's scan feature as a wireless "Hot Spot" finder. With Viper Networks, customers pay only for the minutes used. There are no monthly fees, no contracts, and no hidden charges. And no PC is required to make or receive calls.

Zyxel P-2000W v.2

http://www.zyxel.com

(news - alert) The Prestige 2000W is compliant with the IEEE 802.11b standard and interoperates with any existing 802.11b or 802.11g wireless AP and gateway. Its brand new application is developed to support open standard SIP, which interoperates with major SIP-based call servers, IP-PBXs, and various VoIP client devices. It is not only an ideal alternative for ITSPs (IP Telephony Service Providers) to deploy their VoIP services; it can also be the wireless handset, which is applied in corporate IP-PBX-centric VoIP environment.

Easy-to-use and convenient, the Prestige 2000W delivers high quality voice functionality in a cost-effective way. The call control protocol of the Prestige 2000W is based on SIP v2 open standard, which is interoperable with major SIP-based call servers, IP-PBXs, and other standard SIP-based client devices.

With an improved power-saving design, the Prestige 2000W can be used for a relatively long period of standby and talk time. With an extended life for each recharge cycle, the Prestige 2000W is available where and when you need it.

By configuring a remote IP address in the built-in phone book, the Prestige 2000W provides a direct IP-to-IP call feature when there is no intermediate SIP proxy server available in the network. The Prestige 2000W can also establish an 802.11 ad-hoc network (computer-to-computer network without Access Point), which allows users to use the handsets as wireless intercoms. IT

VoIP: The SMB Opportunity

By Greg Galitzine

According to recent IDC/CompTIA research defining the reseller opportunity for selling VoIP to small and medium-sized businesses (SMB), it appears that the SMB market is ready for VoIP. (define - news - alert)

The report finds that resellers are presented with a "significant opportunity to sell converged communications solutions to SMBs," with the majority of respondents saying that they recognize the business value in converged solutions, such as unified messaging and integrated voice and data applications.

The research shows that 13 percent have already deployed a merged solution, while an additional 40 percent said they are currently evaluating or will make the evaluation within 18 months. Less than one-third of the respondents said they had no plans to carry out or evaluate the merging of voice and data communications over a common network.

While the numbers show an increased awareness within the SMB community of VoIP and its benefits, it's clear that not all SMBs are convinced that VoIP is a worthwhile consideration for their business operations. Perhaps it's simply a matter of not being aware of all the potential benefits that accompany an upgrade to IP-based communications technology.

Cost

The first thing many people think about VoIP is that this technology is primarily a cost-saver. For many enterprises, that is enough of a reason to consider VoIP, and in fact, the cost savings element has many aspects to it that merit consideration. Tremendous cost savings come in the form of lower telephone bills. By converting voice into packets and transporting these packets over an IP network, either a private WAN or public Internet links, corporations are able to avoid the Public Switched Telephone Network (PSTN) and the associated tolls. It becomes theoretically possible to drive the cost of voice transport down to zero.

In the case of an enterprise with multiple branch offices, this is especially true. By using the company's data network, enterprises can eliminate all costs associated with calling between branches. Furthermore, they can have all locations served off of a single IP PBX, thus enabling extension dialing between far-flung locations. Now, by simply dialing a co-worker's three- or four-digit extension, you can speak to a distant colleague as if he were in the very next cubicle, when in reality they might be sitting on the other side of the same building, across town, or in a remote office located on the other side of the world.

If the data network reaches a remote location, so too do the telephony applications that are enjoyed by employees at the main corporate location. Applications such as conferencing, voice mail, unified communications, click to dial: all of these new productivity enhancing services are enabled across the enterprise.

The IDC/CompTIA research found that, when asked about unified messaging, "25 percent of respondents claimed they had already deployed the solution and felt it provided 'good business value.'" Another 41 percent who had not yet deployed unified messaging said they recognized it would provide good business value.

Another major benefit of VoIP for SMBs is that managing the telecommunications system becomes increasingly simplified due to the elimination of the need to look after multiple networks. By combining separate voice and data networks into one network, VoIP enables cost savings from a network infrastructure perspective as well. In so-called 'greenfield' deployments, there is no need to run two separate network cables (one each for voice and data). Furthermore, network administrators need manage only the single converged network.

VoIP also reduces the cost and complexity associated with moves, adds, and changes. Many enterprise VoIP solutions enable administrators to manage the system via a Web-based browser interface and allow managers to enact changes to an employee's phone settings and voice mail settings (for example) remotely, and without the need to call the phone system's manufacturer to send a representative to make those moves, adds and changes. The costs can add up quickly. VoIP practically negates that expense.

There are many benefits to trading up to an IP-based telecommunications system for the SMB marketplace. Cost savings, an increase in efficiency, easier system management, and better integration of business process applications and telecommunications applications all point to a brighter future courtesy of VoIP. Many in the SMB community have already seen the light and are prepared to embrace VoIP on its merits. So this begs the question: What are you waiting for? IT

Greg Galitzine is the editorial director of Internet Telephony magazine.

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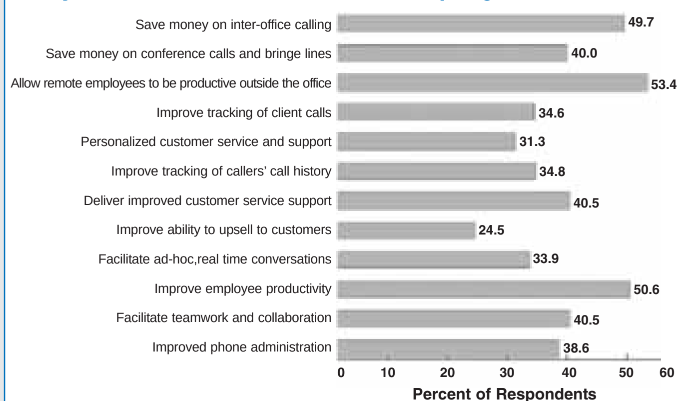
SMBs Looking Adopt To VoIP

Jay Krauser

In an increasingly competitive business climate, organizations of all sizes continue to look to innovative technologies to solve fundamental business problems while slashing costs. The growing adoption of VoIP technology and the applications it supports has provided organizations with the solutions needed to improve corporate efficiency while lowering operating costs. This scenario is certainly the case when looking at multi-national organizations that employ thousands of people. However, large corporations are not the only companies that are deploying VoIP technology.

In an age when small and medium-sized businesses (SMBs) are looking to gain market share while expanding their national and global reach, VoIP solution sets provide an appealing answer to many of the unique challenges facing these organizations. Recent studies suggest that the adoption level of VoIP technologies among SMBs continues to surge with an even greater percentage currently evaluating the benefits and solutions on the market.

Major Business Drivers for SMBs Adopting VoIP Services



Source: Yankee Group 2004 Small & Medium Business Broadband VoIP Survey

After the successful launch of NEC's UNIVERGE SV7000 MPS telephony server, a pure-IP solution for the SMB, the company has seen a sharp increase in businesses that are looking to gain an understanding of the options available to them. Yet, organizations are looking for more than just a cost-saving instrument that allows them to make phone calls. When deploying a new communications solution, businesses are purchasing a full array of tailored applications, hardware components and services that ensure the solution will expand as the organization grows. Productivity enhancing tools such as NEC's OpenWorX Communications Portal, Unified Messaging, and SoftPhones give SMBs the ability to conduct everyday tasks more efficiently while at the office or on the go. In addition to a wide array of applications, businesses of all sizes are increasingly relying upon specialized services to ensure the survivability of their communication system, the lifeline of any organization.

Applications and services continue to be hot-button solutions that existing and new customers are evaluating. However, new business objectives and needs have prompted prospective and existing customers to investigate new ways that an organization's communication system can expand as their business grows. In response to their customers' growing demands, SMBs have increasingly opened new offices around the globe to provide more localized support. With new offices come increasing communication complexities and demands.

In a recent study conducted by The Yankee Group, SMB respondents were asked to list the major business drivers that were prompting them to evaluate VoIP technologies. In the study, 53.4 percent of respondents answered that the ability of remote workers to conduct business outside of the office was a major factor in their decision to investigate VoIP technology (Source: Yankee Group 2004 Small & Medium Business Broadband VoIP Survey). In response to customer demand and an increase in SMB interest in VoIP technologies, NEC's SV7000 MPS telephony solution was developed to provide the needed flexibility to tackle the most complex organizational needs.

The SV7000 MPS IP telephony solution provides a unique set of advantages to those businesses seeking an advanced information system that is both flexible and dependable. With the ability to serve as a stand-alone IP telephony server, a high-density media converter/media gateway or a fully-functional remote media gateway controller for remote office use, the SV7000 MPS sets a standard yet to be attained by any other product on the market. With the SMB in mind, NEC developed the SV7000 MPS to work with an organization's existing hardware and application infrastructure in order to provide remote users with the same features available to those in the home office location. The SV7000 MPS's remote media gateway controller gives remote workers the ability to access the organization's productivity enhancing tools from any remote location. Additionally, remote users may use an organization's communication solutions, such as softphones, presence applications, and other tools, to ensure that standard business procedures are not only completed, but are done so more efficiently.

As SMBs evaluate VoIP technologies and the value they provide, NEC will continue to provide innovative solutions, like the SV7000 MPS, that not only prove to lower operating costs associated with phone charges, but also increase organizational efficiencies between home and remote office locations. NEC's commitment to innovation continues to provide growing organizations with the tailored solutions they need in order to remain competitive in today's business climate. IT

Jay Krauser is general manager of Product Management Division, NEC Unified Solutions. (news - alert)

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Guaranteed Voice Quality With MOS

Financial institutions have extensive data networks in place with best-of-class communication links, both nationally and internationally. Add to that the need for cutting-edge voice communications and CRM applications all while keeping overhead under control, and financial institutions make the ideal early adopter for converged networks using VoIP (define - news - alert) technology. The benefits of VoIP are clear — however careful risk management is critical, as leading UK bank Abbey and BT discovered.

The VoIP Decision

In February 2003, Abbey announced the decision for a company-wide move to an integrated voice and data IP network. The new network would benefit Abbey in two areas — the initiative would lead to improved workforce efficiency as part of the bank's cost review program, announced to the stock market in July 2002, and new applications would deliver better customer service.

Yasmin Jetha, Executive Director, IT at Abbey, said "the move to the new network would allow the company to deploy new applications, such as CRM software, more efficiently and, at the same time, help to reduce costs. Upgrading will improve both our internal and customer facing communications, increase flexibility to meet future business needs, and enable us to deliver our new strategy."

Already a BT customer, Abbey awarded BT a £125 million, five-year outsourcing contract to install and manage a consolidated company-wide integrated voice and data telecommunications solution. Abbey selected BT based on BT's expertise in IP infrastructure, convergence, wide and local area networks, and outsourcing. The service was based on BT Global Services' world leading Hosted VoIP and Multimedia platform. The scope of the project required the transformation of a staggering 800 branches across the UK.

The £125 Million Guarantee

Abbey took a bold step in selecting VoIP as part of a national network upgrade. At a time when most companies were avoiding costly upgrades, Abbey based its aspirations on a young and relatively unproven technology for the scale of the envisaged rollout.

As part of the agreed risk management strategy, BT offered various service level agreement terms; one was an industry first — voice quality on the MOS (Mean Opinion Score) scale. Within the contract, it had been agreed that the voice quality would achieve an MOS of 3.7 or higher, with BT aiming to consistently deliver an MOS of 3.92 (PSTN lines typically have an MOS of 4.1).

MOS on What Terms?

Like VoIP, network management, in terms of MOS, was uncharted territory and, consequently, the terms of the contract did not specify how MOS would be calculated.

Before Psytechnics became involved, the concept was to perform a subjective test every quarter by manually surveying people within Abbey.

This proposal carried the issues of time, cost and logistics. Who would do the test? What questions would be asked? Could quarterly tests really be compared? How accurate and meaningful would the survey be? Consequently, subjective testing was ruled out as impractical on many fronts — scalability, repeatability, and difficulty in interpreting results.

Instead, BT and Abbey decided to explore the potential of objective voice quality measurement technology. BT asked if Psytechnics would act as an independent third party and perform a voice quality audit for Abbey.

Psytechnics proposed the use of the ITU standard for measuring MOS, P.862 PESQ, a test method objectively measuring voice quality, which has been adopted by the world's leading test and measurement companies.

Education, Then a Test Plan

Abbey had limited experience with MOS, subjective testing, or objective voice quality measurement. Thus Psytechnics' initial task was to educate

Abbey. A critical part of the education was confirmation that the contractual point of an MOS of 3.7 was sensible. Psytechnics validated that 3.7 was a reasonable target both for the choice of technology and the customer experience.

Psytechnics also explained that two objective test methodologies were available — end-to end (PESQ) and continuous monitoring — each with its own benefits. Psytechnics advised end-to-end testing would be a good start; however, real-time monitoring would also be required in order for BT to be proactive in addressing customer-affecting problems.

First Test, First Problem

Psytechnics conducted MOS testing on the Proof of Concept (POC) system installed a few weeks prior. The purpose of the POC was to give Abbey the opportunity to experience the quality and feature set offered by the new system and to validate that it worked before going to field trials. For all intents and purposes, the POC acted as a real branch office. Calls could be made internally, out to the PSTN, as well as to any existing Abbey phone system.

The voice quality test for the POC system measured the quality between the different components of the new network and its interactions with the existing networks. Tests were also performed on the backup systems. As the POC system had been "up and running" for a few weeks, no problems were expected. Surprisingly, an issue was discovered.

MOS testing showed a problem for calls to existing Abbey PBX users, but, interestingly, it only happened for one in four phone calls. Analysis led the network operators to track the problem to a single media gateway, where the fault was identified and corrected within two hours and the voice quality returned to being above 3.7 for calls.

The surprise discovery of the problem on a VoIP system thought to be "field trial ready" proved that BT's ability to provide customer-ready communications was going to require MOS information. More importantly, finding the fault using MOS measurements confirmed that BT and Abbey had made the right decision to include Psytechnics' MOS testing in the SLA.

Moving to the Real World

The next phase was to roll out a VoIP trial system to a number of branches across the country to prove that what works in the lab can also work in the real world. The fact that all 800 sites would have identical configurations meant the field trial test plan could be reduced to only three sites. One of the benefits of VoIP is that if a problem is addressed at one site, it can be swiftly addressed at all sites.

Psytechnics performed testing between three branches and the head office in Milton Keynes. MOS tests were performed between the branch office and the PSTN connection and also between the branch office and a FeatureNet connection (Abbey's existing PBX system).

Fifty calls were made to each site with 20 PESQ measurements per call. A total of 1000 PESQ test points per site were gathered over a two hour period. Of the 150 calls made to all branches only two calls had an average MOS below 3.7, representing a conformance metric with the service level agreement of 99.8%.

However, six calls recorded individual PESQ test points below 3.7. In other words, there were times during those calls where the voice quality momentarily dipped. Further analysis revealed the degradation coincided with single speech events and was equivalent to about three seconds of significant degradation during 22 minutes of tested speech.

Strictly speaking, the SLA target of 3.7 for all calls wasn't achieved. Yet, in practical terms, the field trial systems were fit for purpose — even the PSTN can have three seconds of problems from time to time. The trial was deemed a success and the decision was made to continue with the full 800-branch rollout. The rollout plan called for 20 branches per night to be converted, but again, the lack of MOS measurement (customer experience) led to unsatisfactory results.

With each new branch BT installed there would be minor issues, which would first be identified by customer complaints — not the ideal way to manage a network. The BT team found that a significant amount of time was spent trying to verify whether it was a problem with the VoIP network or with the external network (such as a mobile network), or whether it was actually no problem at all.

Heritage Trust FCU Improves Call Center Performance By 50%

Credit unions are unique in the financial world: Because the members are the owners, there are no shareholders to whom profits must be paid. Everything is returned to members in the form of better rates on loans and deposits and lower or no fees on products and services. Summerville, South Carolina-based Heritage Trust Federal Credit Union is intent on providing a place where members can find not only service offerings, but answers and solutions. To achieve that, HTFCU is committed to continually providing member service excellence across all of its business units — including its contact center.

Opportunity

With more than 70,000 members, call center agents handle close to 1,000 calls daily in the Heritage Trust Call Center located at the corporate office in Summerville. These inbound and outbound interactions include tasks such as opening new accounts, taking loan and second mortgage applications, purchasing certificates of deposits, setting up appointments for investment purposes, assisting in researching/resolving account problems, and issuing ATM and debit cards. To effectively meet its members' service expectations, HTFCU decided to invest in a solution to monitor and record member interactions to ensure the highest levels of service were being delivered. And, as a financial institution, HTFCU needed to record 100 percent of its calls for compliance purposes. Prior to purchasing a customer interaction recording solution, the company's recording method consisted of voice recorders on the telephone, which proved to be both inefficient and unreliable. It was clear that HTFCU needed to invest in call recording technology that was fully automated. Ease of use, reliability, and robust archival capabilities were all additional characteristics the company desired in the recording technology it would select. An additional requirement — to meet network infrastructure needs — stipulated the solution had to operate within its converged Cisco Voice over Internet Protocol (VoIP) environment.

Solution

To fulfill its customer interaction recording needs, HTFCU implemented ContactStore for IP, Witness Systems' recording platform that allows for 100 percent customer interaction recording in an IP environment. HTFCU selected ContactStore for IP based on its high reliability and archival simplicity.

"In our selection process, we needed a compliance solution to meet our regulatory requirements first and foremost, but HTFCU also needed a fully reliable, scalable system that would work in tandem with our existing IP infrastructure. For these and other functionality capabilities, we invested in ContactStore for IP," explains Ariel Smith, Network Services Manager at HTFCU.

ContactStore for IP handles HTFCU's call volumes with great ease, recording 100 percent of the credit union's member interactions. Further, the solution makes managing the captured interactions much easier. According to Smith, supervisors are able to work 30 percent more efficiently because the solution is so user-friendly.

ContactStore for IP also met HTFCU's archival needs. Like most financial services companies, HTFCU needed to archive its calls for one year. The credit union needed fast access to these interactions through efficient archive and retrieval. ContactStore for IP automatically tags and categorizes calls, so when

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interactions are stored, they can be easy to retrieve — for instance, in the case of a dispute. Once a call is located, it can be played and restored back into the system within a matter of seconds.

Since implementing ContactStore for IP, the solution has helped HTFCU make great strides in its delivery of service. Its supervisors live-monitor calls on a daily basis to ensure the highest levels of service are delivered. Being able to listen to an interaction real-time provides the opportunity for HTFCU supervisors to offer immediate feedback on the best ways to handle members' requests. Using the solution's "Exec Record" function, supervisors can record and save the interaction from beginning to end, regardless of when the recording was initiated. Additionally, recorded interactions for each call center agent are reviewed and evaluated weekly, and performance scores are incorporated into quarterly and annual reviews.

"From a certain standpoint, a member calling for a particular service is the boss — we want to make sure his or her experience with our call center agents reflect the highest caliber of service," comments Smith.

Using ContactStore for IP, call center agents are trained using 'best practice' interactions that were captured using the solution. This means HTFCU is able to deliver real-life scenarios to call center agents to enhance the training experience. Agents are then coached on the elements of a high quality contact.

"Since we started using ContactStore for IP, we've seen our call center agent performance scores increase by 50 percent," says Smith. "By observing the qualities of high-value calls, our call center agents are able to experience and replicate how to most effectively answer questions and offer solutions, which, in turn, keeps our members happy."

Often, problems took a number of days to track down, but only a few minutes to rectify. The operations team confessed that the hardest part was correlating the drop in quality given in customer complaints against what actually was happening in the network. Thus, the need for continuous, real-time MOS monitoring with diagnostic information was realized.

BT decided to accelerate its plans to implement such a system for the Abbey network and approached Psytechnics for help introducing an interim real-time voice quality monitoring solution with diagnostic capability.

Real-time Monitoring Shows Real-Time Results

The decision to implement real-time 24x7 monitoring revealed more than just problems with packets and route configurations. On one occasion, the 24x7 monitoring system, PsyQ, identified a branch suffering from voice quality issues. Further investigation revealed that builders had dismantled the room in which the router had been. The VoIP system had transferred to its backup ISDN connection, which caused the drop in voice quality. PsyQ spotted the problem, while the legacy network management systems missed it.

In one case, Psytechnics witnessed BT giving a tour to an Abbey employee. The tour ended with a demonstration of PsyQ from a BT customer support agent who had been using PsyQ for approximately one week. The agent showed how PsyQ had reported some bad calls at one of the Abbey's branch offices within the last two hours, and then he confidently drilled down into the branch data showing the quality and diagnostic information for the affected calls.

PsyQ revealed the issue was either an unreliable link or LAN congestion and so, his first task was to check the access router, which revealed no problem. He then requested a check on the branch router, which revealed a flapping link, and corrective measures were taken.

The agent was asked, "Did you need PsyQ? Surely a router alarm would have told you about the link flap?" The agent's response was, "PsyQ lets me prioritize customer-affecting faults over the thousands of alarms raised every day. PsyQ saves me time and, in fact, no one knew about the router problem before me." PsyQ's MOS and diagnostic capabilities enabled a speedy solution — fault recognition to isolation and identification within 10 minutes.

This example, and most other problems spotted by PsyQ would have been considerably more difficult to characterize, quantify, and locate if the operations team relied purely on customer complaints and legacy network management systems.

What was Learned?

With the VoIP systems now live in all 800 branches, the emphasis has shifted from end-to-end testing to real-time monitoring, so that BT is instantly aware of customer-affecting problems. BT has made further investments in Psytechnics technology and is aspiring to have a deeper integration, such that it works closely with existing infrastructure and network management systems.

Refining the SLA will be a necessary step. Though an MOS of 3.7 had been declared the minimum, the SLA did not allow for exceptions. The VoIP system delivers average quality of MOS 3.92. However, a very small amount of degradation during a call could drop scores below the threshold.

The contract specified a hard limit of 3.7 although, in practice, isolated drops in single test scores do not affect overall customer opinion. Psytechnics will be publishing a guideline for MOS-based SLAs to assist both service providers and purchasers.

In terms of business, everyone is pleased with the benefits of MOS. John Blake, Head of Hosted IP Telephony for BT Global Services said, "Having MOS capability has differentiated our customer proposition from other suppliers of VoIP. Because of the reputation we've earned at Abbey, we have significant other prospects." IT

Talking International VoIP

By Greg Galitzine

Parlez-vous VoIP? Sprechen-Sie VoIP? Usted habla el VoIP? Parlate il VoIP?

There is no denying that VoIP (define - news - alert) is truly becoming an international phenomenon. I recently had the opportunity to speak with Karen Schapira, Global Marketing Communications Manager at Go2Call regarding that company's role in the global VoIP marketplace, and she shared with me some insight from Go2Call as well as a list of 10 considerations for service providers looking to provide a VoIP offering worldwide. For more information, feel free to visit the company online at http://www.go2call.com.

1. *Provide a comprehensive offering:* Go2Call (news - alert) delivers a full suite of hosted services and applications, including gateways, softphone, ATAs, "best of breed" to service providers, PTTs/Telecom Carriers, Satellite providers, ISPs, Broadband providers, and Licensed VoIP Operators. The platform is designed to empower providers to launch hosted PBX services to enterprises, broadband phone services to residential subscribers, and calling card services to resellers and call shops.

2. *Focus on emerging and/or deregulating markets:* Go2Call has seen markets in the Middle East, Asia Pacific and Latin American embracing and gradually adopting VoIP solutions. Many countries in these regions are in the process of deregulating.

3. *Obtain high quality domestic termination.*

4. *Embrace comprehensive reporting and monitoring tools:* Providers must help manage the customers' business. It is extremely important to allow the customer to manage their business, which is why Go2Call offers customers an advanced multi-tiered billing system providing flexibility for pre- and post-paid options while giving service providers multi-level control of billing activities.

5. *Advanced Billing Solutions:* Monthly billing for DIDs, top cards, calling cards (see #1).

6. *Strategically place POPs, ALGs, feature servers:* Go2Call's PoP locations provide customers around the world with a network strategically optimized for both call quality and cost. Each location is carefully selected to ensure that they can offer their end users a reliable calling experience. We determine the best carrier for each call by optimizing the relationship between quality and cost.

7. *Work to find the right distribution channels and strengthen those relationships.*

8. *Offer Private-label capabilities:* Go2Call provides private and white-label branding options, which essentially enables customers to provide powerful and lucrative VoIP services while building equity in their own brands and reducing their implementation time and cost.

9. *Localize your solution:* Offer your solution in multiple languages, currencies, features, call routing, etc… We have worked with the largest Japanese ISP (Nifty) to provide them with an entire Web site and dialer localized in the native tongue. We offer currency and feature lists in other languages as well. Recently, we've customized various solutions in French, Spanish, Turkish, and Romanian.

10. *Keep local staff:* Look to hire company staff that live in the regions where you are doing business, for they will best understand the cultural barriers.

Internet Telephony Conference & EXPO EAST: Best of Show

Those of you who were able to join us in Fort Lauderdale for IT EXPO East in January know first-hand what a tremendous show it was. The conference sessions were well-attended and informative, the keynotes by Tom Ridge and Ron Insana were SRO (Standing Room Only), and the conference hall itself was magnificent. But none of it would have been possible without the exhibitors on the show floor.

Naturally, from among the more than one hundred vendors, some exhibits stood out more than others. Some were notable for their sheer size; others announced new products; some were notable for their large crowds; and still others featured eye-catching scenery. The following is a list of vendors whose products or services stood out most.

COMPANY NAME	WEB SITE	COMPANY NAME	WEB SITE
1. Aculab	http://www.aculab.com	16. MultiTech Systems	http://www.multitech.com
2. Adomo	http://www.adomo.com	17. Netcentrex	http://www.netcentrex.net
3. AT&T	http://www.thenewatt.com	18. NetLogic	http://www.netlogic.net
4. Belden/CDT Networking	http://www.power-sense.com	19. Nortel	http://www.nortel.com
5. Brooktrout	http://www.brooktrout.com	20. Psytechnics	http://www.psytechnics.com
6. Dialexia Communications	http://www.dialexia.com	21. Quintum Technologies	http://www.quintum.com
7. Digium	http://www.digium.com	22. Samsung	http://www.samsung.com/bcs
8. Excel Switching	http://www.excelswitching.com	23. Sangoma	http://www.sangoma.com
9. Extra Strength Phone	http://www.extrastrengthphone.com	24. Sentito	http://www.sentito.com
10. GlobalNet	http://www.gbne.net	25. Talkswitch	http://www.talkswitch.com
11. Grandstream Networks	http://www.grandstream.com	26. TomatoVine	http://www.tomatovine.com/corp
12. Iperia	http://www.iperia.com	27. Vonexus	http://www.vonexus.com
13. Iwatsu	http://www.iwatsu.com	28. Xelor Software	http://www.xelorsoftware.com
14. Lumenvox	http://www.lumenvox.com	29. Zoom Technologies	http://www.zoom.com
15. Minacom	http://www.minacom.com	30. AGN Networks	http://www.agnvoip.com

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INTERNET TELEPHONY® EDITORS' CHOICE AWARD 2006

SurfinBird IX67

Intertex
12 Federal
Dennis, Massachusetts 02638
Phone: 508-385-6335
Fx: 208-474-0956
Web site: http://www.intertex.se

Price: The SurfinBird IX67 (news - alert) has a MSRP list price ranging from $185 to $329 depending upon the hardware options selected (802.11g/b wireless access point and FXO/FXS are optional features). The SIP Switch, SIP-oriented IP-PBX has an MSRP of $500.

Many small to medium-sized businesses are looking to leverage standard SIP phones along with SIP trunking for less expensive call termination by simply installing a SIP (define - news - alert) server on their network. Unfortunately, most VoIP solutions don't address this market or, if they do, it's often a full-fledged IP PBX system that may be overkill in both price and features. Enter Intertex, which offers all-in-one CPE devices that can include built-in ADSL modems and firewalls as well as built-in SIP servers. TMC Labs tested the Intertex SurfinBird IX67, which features an integrated firewall, router, and SIP server, four Ethernet ports, as well as the optional 802.11b/g WLAN antenna and built-in telephony ports (FXO/FXS). Supporting up to 30 simultaneous SIP calls, along with a powerful dial plan configurable via a Web interface (See Figure 1), the IX67 is perfectly positioned for the SMB market.

One key advantage of this solution is that it meets requirements for E911 by using the backup PSTN connection. From the browser interface, you can simply configure 911 to dial out through the PSTN connection instead of over VoIP. It also allows you to configure a redundant secondary SIP proxy server and routing features to allow local extension dialing in the event of a broadband failure. This optional redundancy functionality is built into the IX67 and is turned on via a purchased license key. It also makes the IX67 a great backup solution for any SIP-based virtual or hosted IP PBXs that requires broadband connectivity. Using the IX67 you can be sure to have 100% 911 access and local extension-to-extension dialing in the event of broadband failure. For example, Cisco (quote - news - alert) and snom phones can have a backup proxy in their configuration. The system can be set to switch over using the IX67, guaranteeing inner office calls as well as 911 and other calls through the built-in FXO gateway and/or an additional four-port PSTN gateway. All basic PBX functionality including some advanced functions, such as call waiting, three-way calling, and call transfer, will continue to function in the event of a broadband outage. Basically, the IX67 becomes your local PBX for all your SIP phones.

RATINGS (0–5)
Installation: 4.5
Documentation: 4.5
Features: 4.75
GUI: 4.25
Overall: A

SIP Switch users can register to the built-in registrar proxy using proper email-like addresses like, such as tomsmith@acme.com. This gives free access to and from all users in the open SIP world that can dial SIP URIs. The SIP Switch will securely route inbound SIP URI calls as well as inbound calls made to your PSTN number owned by a SIP service provider. In fact, the IX67 features some pretty powerful inbound routing capabilities. On an inbound call, it can ring one SIP address or multiple SIP phones one after the other in sequence; or it can ring several SIP addresses at once. It can even forward the call to voicemail after a predetermined period of time. The SIP Switch also features ENUM lookup before passing a call to the PSTN. If there is a known SIP address for the telephone number you are calling, the call will be forwarded to that address, so that it will be a 100% IP call instead of using the more expensive PSTN. For prevention of SPIT (Spam over Internet Telephony), the IX67 features the ability to restrict incoming callers that are dialing the system using a SIP URI.

The IX67 features a Stateful Packet Inspection (SPI) and packet filtering firewall to ensure that your LAN stays private, secured, and protected from malicious attacks. In addition, the unit

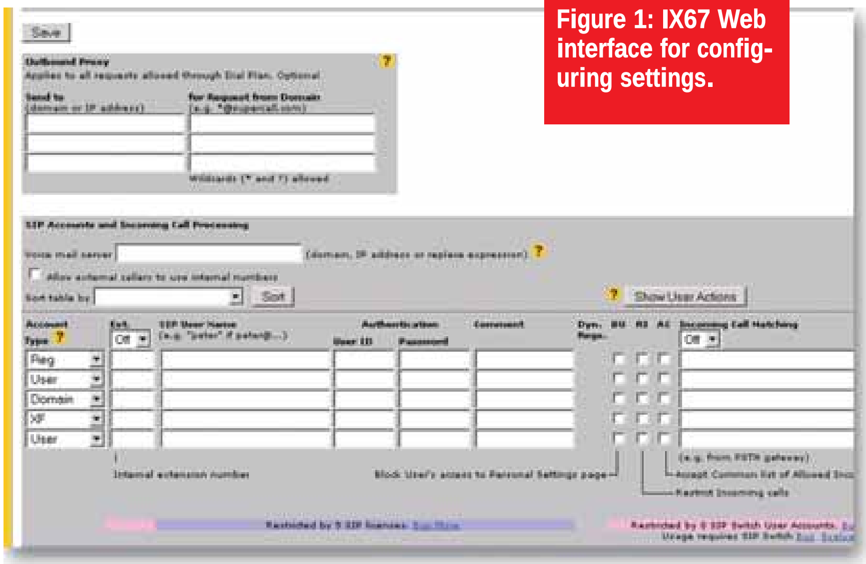

Figure 1: IX67 Web interface for configuring settings.

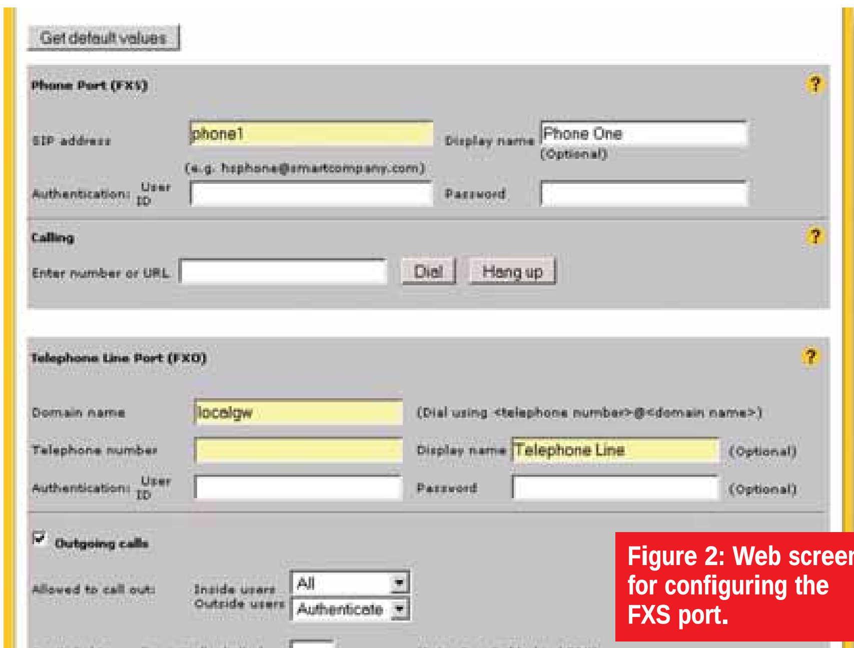

Figure 2: Web screen for configuring the FXS port.

does QoS, giving SIP RTP and signaling traffic a higher priority. It supports tagging packet prioritization with Diffserv and TOS and bandwidth limiting. It also supports fax pass-through, and you can upgrade the IX67 to support ENUM lookup. It also offers optional VPN termination in addition to the standard VPN pass-through capabilities. This supports IPSec, PPTP, and L2TP with certificate handling. DES and 3DES are hardware accelerated. Support for AES encryption is also included.

The router is delivered with several preset confgurations, and the IP addresses are retrieved and distributed automatically via a built-in DHCP server (or you can set them manually). A Wizard is also built into the Web interface for creating a dial plan, though we would have liked the Wizard to support more complex wizard-driven configurations.

Relatedly, the dial plan screen in the Web interface is very flexible, but also very complex to learn. You can, for instance, configure "8" to dial out via the FXO port, "911" to dial out via the FXO port, "9" to dial out through Vonage (using a softphone account to get your SIP logon credentials), and so forth. Suppose you make a lot of calls to the Iran and you find an inexpensive Iranian termination service provider. You can configure "7" to dial out through this Iranian SIP service provider. Better yet, you can have the system detect "01198" (011 = international dialing, 98 = Iran country code) and have the IX67 automatically route the call to the Iranian termination service provider. Thus, you can build your own LCR (least cost routing) table using the IX67.

Another nice feature is that the FXO Port is accessible from both the LAN and WAN with authorization and it is accessible for emergency calling and other local calls. The FXS port (an analog phone extension mapped to a SIP address) is also accessible via both the LAN and WAN (See Figure 2). It's worth mentioning that the front panel displays the current status of the product and its connected networks.

Conclusion

We liked the Least Cost Routing (LCR) capabilities of the IX67. The IX67 runs on VxWorks, a popular real-time operating system and it certainly has the horsepower to perform SPI on the firewall to ensure proper security, as well as powerful VPN capabilities and powerful SIP routing capabilities, all within a relatively inexpensive box. We certainly liked the convenient dialing between SIP phones, soft SIP clients, and regular telephony (PSTN), using URIs, E.164 numbers and internal extensions. Add to the fact that it supports PBX-like functionality (call waiting, three-way calling, call transfer) and a PSTN backup in the event of broadband failure and this neat little box is deceptively powerful for its size. IT

Building Momentum for VoIP:

It All Comes Down to Execution

To the casual observer, the enterprise communications industry must seem as competitive as professional football.

NFL coaches strive to instill in their team a fundamental philosophy that success is predicated on a number of factors, like commitment, passion, and work ethic. Business communications vendors also seek to project the impression that their ability to serve customers through advanced technology is what sets them apart from the competition. As important as it is for NFL players to embrace their team's underlying philosophy, it is equally critical that communications providers, distribution partners — and even customers — align themselves to ensure that new technologies and solutions meet the expectations of the enterprise.

Pro football teams and communications solutions providers also hold the belief that their most distinctive competitive advantages, whether they are game plans, go-to-market strategies, or technological developments, are completely hidden from the competitive view. NFL coaches may chose to hold closed practices and use decoys to keep information away from the opposing team. In our business, many vendors intentionally deliver misinformation in an attempt to shield their real purpose from competitors.

As much as we'd hate to admit it, however, there is very little vendors don't know about each other. We all compete against each other on a daily basis, so we understand the way specific products are positioned, configured, and distributed. It's nothing that we haven't seen and, frankly, this knowledge makes little difference in the selling process.

As much as strategy, road maps, and playbooks may prepare us, the game is still won and lost on the field. As Voice over IP (VoIP) gains more and more traction among end users, it will be increasingly important for providers to stick to basics and not get completely overwhelmed and sidetracked by new technologies.

Just look at the history of VoIP (define - news - alert) over the last ten years. When it first hit the mainstream in the latter half of the 1990s, the pundits, visionaries, and even the providers, touted the technology's primary benefit as a means to eliminate toll charges by transporting voice traffic on the data network. But, as is often the case in emerging technology, real life experience was vastly different.

Though it took awhile, manufacturers eventually learned that the real benefit of VoIP communications resides in powerful applications that enable businesses to be more productive and efficient and, at the same time, reduce costs. Vendors, resellers, service providers, and systems integrators began to call signals from the

By Craig W. Rauchle

same playbook. As a result, an influx of solutions such as presence management, Web conferencing, collaboration, desktop sharing, and other IP-powered tools began to permeate the market. But as technology changes, and customer requirements continue to evolve, we're starting to see that this game plan will only take us so far.

Within today's communications industry, there are several issues that should have a tremendous impact on the ability of providers to satisfy customer needs.

One of the more visible opportunities is the growing recognition that industry standards will evolve into a primary technology protocol. Already, we're seeing a significant deployment of 802.11 wireless infrastructure in the enterprise space, and we expect to see similar activity with SIP over the next few years.

The benefits of standards-based protocols to businesses are numerous. SIP, for example, enables multi-vendor solutions in an open environment, which by anyone's measurement is a very positive step for an enterprise. Customers will have the opportunity to seamlessly integrate business-enhancing applications into their networks and, if they choose, install a variety of third-party endpoints into their infrastructure without worrying at all about the problems of interoperability and performance. SIP is about delivering choice and, we, as an industry have to come to terms that this is a paradigm we need to address. It's going to happen, and happen soon.

As technology itself becomes more innovative, more robust, and more beneficial for the customer, the battle for market share isn't going to be centered on which vendor's products are more cutting edge or visionary; rather, it's going to revolve around which solutions do more for the customer. It will be about getting back to basics, such as offering a realistic return on investment, delivering applications that tangibly improve business processes, and providing, at a moment's notice, comprehensive and ongoing technical support and customer service.

In the ever-changing world of VoIP, we are seeing that different vendors have a variety of approaches to address their customers' evolving needs. For instance, service providers and hosted vendors with the ability to bundle voice, data, video, applications, and a variety of other communications and system administration services, offer customers a single point of contact to purchase all of their communications needs. What's more, the ability to have an outside party host, manage, and maintain the system emancipates businesses that do not have the resources to manage their own networks.

On the other side of the coin are the traditional CPE manufacturers, who have a long and proud history of developing advanced enterprise communications technology. These vendors have a strong track record in creating numerous solutions that enable businesses to run more productively and efficiently. System scalability, flexibility, and ease of management are good examples of how CPE manufacturers transfer real business perspectives into tangible solutions to address unique customer requirements more easily than service providers, who are normally equated with more commoditized offerings. CPE vendors also claim that they can deliver superior service and support, based upon the fact that they are solely focused on business communications. This contrasts with the RBOCs and other providers, who must also attend to the needs of hundreds of thousands, if not millions, of consumer customers.

As VoIP continues to gain acceptance, and standards-based solutions begin to enter the market, customers ultimately will have much more choice in the devices and applications they can successfully integrate into their communications infrastructure. The role for manufacturers, service providers, and resellers becomes a lot more complicat-

Keeping It Simple… By Keeping It VoIP

Connecting to information services through the public network doesn't make business sense

By Nitin Krishna

After making the investment to migrate voice traffic to the cost effectiveness, quality and flexibility of voice over IP (VoIP), the idea of a carrier connecting to an information services provider through the public switched telephone network seems, well, so analog. More specifically, the logic of staying completely off the PSTN when contracting for telephony services tracks perfectly with the decision to use VoIP in the first place.

As wireless, traditional wireline, next generation, and cable telephony carriers transition to IP networks, their customers — the folks making the calls — are still expecting to connect to the telephone services to which they have become accustomed — including directory assistance and operator services. In addition, these callers are looking for more enhanced features on their communications instruments, be it through voice (like movie listings and show times, restaurant reservations, and travel assistance), data (information delivered directly to the phone unit or handset through text messaging — including directory assistance listings, sports scores, maps, and turn-by-turn directions), or video (broadcast, narrowcast, or personal). Delivered through call center operators, speech recognition systems or user-generated connections directly to the source, IP networking is tailor-made to generate and deliver these services to the content hungry customer.

The Buried Cost

The capital investment to transition carrier networks to secure VoIP is hefty, but the returns come with the first call. Bypassing the PSTN and connecting callers through VoIP — through toll bypass — brings savings right away through the VoIP-enabled information services provider. However, if that provider is still connecting to call centers or automation systems through a series of PSTN-based switches, the connecting costs start adding up very quickly. Those costs have to be passed on somewhere. As a result, many of the high-margin services the carrier would like to offer to callers may come with a steep price tag.

There is another "charge" — it's the penalty paid when the PSTN-based information services provider can't technologically offer the emerging services users want. The flexibility and versatility of the VoIP platform allows for rapid incorporation of new voice, video, and data products — at the speed and cost effectiveness that this ever-changing marketplace demands.

Network Quality

It could be said that — compared to the reliability and trustworthiness of the more than century-old public switched telephone network — using the upstart Internet to carry these high-margin and vital services might be a bigger risk than a lot of carriers are willing to take. After all, how comfortable can a carrier actually be when it is sending its callers — that is, revenue and plenty of it — into a network that can be visually represented by a cloud? On the other hand, the information service provider can't expect a network cobbled together with low-grade servers and a couple of routers in the garage to support the higher-volume broader-banded needs of VoIP carriers and their users.

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The provider has an obligation to the carrier to offer a redundant network architecture designed to operate at carrier-grade quality. The provider's investment in a network operations center (NOC) affords around-the-clock management and support of its VoIP infrastructure, using real-time network-monitoring tools to provide status and performance conditions.

Just like carriers, ensuring quality at all levels is vital to the credibility of information service providers. Service level agreements (SLAs) are necessary to hold these providers accountable, and afford a written and implied trust for ongoing uptime and network quality.

Content users want

The power and promise of VoIP can be summed up in one word: content. Despite its best efforts over the years, the technologies offered through the PSTN pale to the flexibility offered through IP. Now, audio, video, and data content can ride on the same network infrastructure. And the forward-thinking VoIP carrier that contracts with an innovative information services provider can together deliver technology enhancements that meet user demand. Consider some of these features, effortlessly enabled through a VoIP platform:

- SMS: Sending information (one- and two-way) directly to the user's handset. But it's not only for the wireless set — consider multimodal units to download requested maps, pictures, video and other real-time data through a third-party provider.
- Preference: The ability to recognize and store a caller's preferences for language, delivery options, and the type of user device, and offering a customized call flow based on that information.
- Speech-driven Personal Address Book: Allows users to store listings and connect from any phone anytime.
- Games and Goodies: Virtual reality has never been more real — playing competitive matches, handset sports and other interactive opportunities.
- Video Clips: News clips, music videos, even home movies — all stored and recallable to the handset (consider settling a sports argument by dialing up the video clip and watching the answer to the trivia question).
- Ringtones, ringback tones, and other personalization features: One connection brings high volumes and high margins from users looking to change their sounds often.

And in a world of "quadruple-play," cross-promoting services through a single telephony connection is natural and carries more revenue opportunities. Back-end messaging intelligently communicates to callers based on demographics, language preferences, frequency of calling, caller's location, even time of day, week, or month. For example, a cable telephony user dialing 4-1-1 for directory assistance can hear about (and order) a pay-per-view boxing match that night, or a wireless offering, or even another enhanced service through speech or touch pad. It's then confirmed via text message or call back, and the data is securely sent to the billing department.

This convergence is happening now. The industry is experiencing an explosion of content and devices that appeal to nearly every interest, segment and demographic imaginable. And those that haven't been marketed, well, just wait a minute, and they will be. That's why considering the easiest, most cost effective and customer focused solution to delivering services is essential — especially since that's what carriers are or will be using: VoIP technologies. IT

Nitin Krishna is director of product development and management for INFONXX. For more information, please visit the company online at http://www.infonxx.com. (news - alert)

ed under this scenario.

We will all be challenged to devise entirely new go-to-market strategies that will combine the functionality, features, reliability, and support typically associated with premise-based solutions. However, the growing demand by businesses to have their communications needs managed and maintained by outside resources will also have a profound effect on this new dynamic. As a result, I expect we'll see a sharp uptick in platforms that are sited at data centers, central offices, and resellers. All of these solutions will eventually be standards-based, and businesses will place as much — or even more — emphasis on selecting their SIP-based products as they do in choosing their communications platform.

So who will best serve businesses by delivering VoIP in a manner that is most beneficial to the customer? As NFL coaching great John Madden once said, it all comes down to fundamentals. Chances are it will be providers who are flexible enough to take the best elements of both CPE and service providers and create bundles that will pragmatically address customer needs.

Much like the game of professional football, the continued growth of VoIP in the marketplace all comes down to execution. As technology continues to evolve, and SIP and other standards become more prevalent, it will be the providers that can deliver the most compelling aspects of premise-based and hosted solutions, that can deliver flexibility, and that can tangibly impact business processes, that will reap the benefits of these opportunities. IT

Craig W. Rauchle is President and Chief Operating Officer of Inter-Tel, Inc. For more information, please visit the company online at http://www.inter-tel.com. (news - alert)

Making the Most of Mobility

How to ensure performance and productivity are not overlooked in the march towards mobility.

Enterprises of all sizes are deploying wireless networks, investing in devices and extending application access to their mobile workers at an increasing rate. This growing popularity of mobile computing can be attributed to several key factors. Devices are getting smaller and more powerful. The amount of information they can store and the number of applications they can run is growing. Declining prices for this hardware, along with increased standardization of operating systems and protocols is also making mobile computing solutions economically viable to a wide range of organizations.

It's clear these factors are making mobility more attractive. According to IDC's October 2005 report, *Worldwide Mobile Worker Population Forecast and Analysis 2005–2009*, the mobile worker population will account for more than one-quarter of the global workforce by 2009. That's an increase from more than 650 million workers worldwide in 2004 to more than 850 million.

As mobile computing grows in popularity, the benefits of maintaining a mobile enterprise are becoming more accepted. Encouraged by the promise of increased responsiveness to customers and enhanced productivity and work/life balance for employees, enterprises are finally recognizing the strategic importance of mobile workers.

Implementing a mobile computing policy is an essential element to ensuring these deployments go smoothly and promote worker productivity. While the benefits of a mobile workforce are clear, many enterprises have delayed their mobility initiatives due to concerns about security. As a result, most mobile computing policies that do exist revolve around security guidelines — many of which actually impede an organization's ability to reap the productivity and efficiency benefits that prompted a wireless roll-out in the first place.

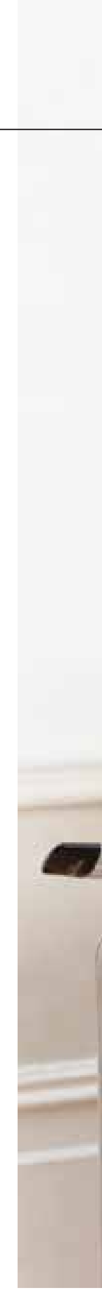
Key Considerations

Today, although security is a critical part of managing a mobile workforce, organizations are realizing that it's important to consider several other key factors, such as performance and ease-of-use. While exact requirements may vary depending on the size of each individual organization, there are five things enterprises must consider when devising a mobile computing policy:

1. Security
2. Performance
3. Productivity
4. Autonomy
5. Cost, control, and management.

Security

The physical boundaries between public and private networks are blurring, increasing the already critical need for security. It's obviously important to ensure that mobile workers can't be snooped on while accessing the corporate network remotely, or that key data and access privileges can't fall into the wrong hands if a device is lost or stolen. Because mobile workers often roam across multiple networks, including some that the enterprise does not own or control, such as WiFi hotspots, security policies need to be tailored according to the network in use. A mobile computing policy should also take into consideration the need to comply with government or industry regulations.



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By Andy Willett

This may include data encryption or user authentication. The policy should also make provision for regular backups.

Performance

Mobile workers are often on the frontlines of the business, working directly with customers on mission-critical tasks. They do not have the time to cope with application crashes and slow connections. When setting access policies, consideration should be given to bandwidth and device-type so that users can enjoy optimum performance and customers can benefit from maximum effectiveness. For example, consider limiting access to bandwidth-intensive applications when workers are using a slower connection, but increasing their access privileges when they switch to a higher-performance network.

Productivity

By the same token, higher security precautions and usage limitations traditionally associated with mobile devices must not impact the convenience of mobility. If users have to re-authenticate or reconfigure — or, worse, recover lost data — every time they lose a connection, an application crashes, or they suspend their mobile devices to conserve battery life, their dissatisfaction with the network will increase, while their productivity decreases. If these problems occur, they will either find a way to bypass security precautions altogether or simply refuse to use the device. Consider making the process as seamless as possible for the user by maximizing application persistence.

An organization must also consider its policy on user support: mobile workers may need to be given priority over stationary or remote workers. Equally, ergonomic and health considerations should be factored in here, as they affect user effectiveness both short- and long-term.

Autonomy

As with any policy, a degree of control is given to the user. With that control comes responsibility, so a mobile computing policy should include guidelines on acceptable and appropriate use bearing in mind security, productivity and etiquette.

Cost, Control, & Management

A mobile computing policy should also consider aspects such as asset tracking and standard software distribution, to prevent use of unauthorized applications and devices, ensure mobile assets are utilized effectively, and minimize the management burden.

Mobile devices are difficult to manage. It is frustrating to troubleshoot, since there is no physical connection and problems can be intermittent. In the mobile computing policy it is, therefore, important to consider policy management features that let IT managers prevent or restrict mobile workers from using designated applications or networks based on the type, speed or name of a network to which a user or device is connected.

With these five factors in mind, here are the steps involved when drafting a mobile computing policy:

1. Map the territory

Consider whom the policy affects, where they are located, and what devices and networks they will be using. This may include networks (such as hotspots) and devices (kiosks) that are beyond your control, so any default settings should assume that all networks are untrusted.

2. Draw up the wish list

Consider to what information those users need access, how frequently and when they need it. Do they require simple e-mail and Web access or do they need to upload information to a central database in real-time?

3. Compare against existing policies

The governing principles of who has access to what should be laid out in an existing security policy; the mobile computing guidelines should reflect this.

4. Access and acceptable use

Next, it's time to lay out the mobile computing guidelines themselves focusing on access privileges, permitted activities, and acceptable use for each type of device, user and location/network. Consider not just what makes the action secure, but also how to optimize network and application performance. For example, it might be necessary to limit access to high-bandwidth applications from low-bandwidth connections, or to minimize authentication procedures for non-confidential data when users are roaming across several networks. Outline here the technologies that will be used to enforce policy compliance.

Most mobile computing policies actually impede an organization's ability to reap the productivity and efficiency benefits that prompted a wireless roll-out in the first place.

5. Incident response plan

Any mobile computing policy needs to include an incident response plan in case of a security breach. IT also needs to include a corporate policy towards unacceptable use, whether deliberate or through negligence.

If a wireless solution is doing its job well, no one should know it's there. For the mobile worker, this means real-time, continuous access to the right data and applications whenever and wherever they need it. For IT professionals, this means a mobile computing solution that offers users what they need and want. It also means a solution that has the least drain on IT resources in regards to deployment, administration, and management.

Consequently, a successful mobile computing policy is essential to ensuring that the needs and expectations of mobile workers and IT professionals are fulfilled. The policy should not only ensure mobile workers are safe and compliant, but also that they are productive and able to do their jobs without frustration, interruption, or inconvenience. Ultimately, the net result should be a faster return on any mobile investment through enhanced productivity and increased customer satisfaction. IT

Andy Willett is senior VP of sales at NetMotion Wireless, a provider of remote secure access solutions that allow users to connect to mission-critical applications over wireless networks. For more information, please visit the company online at http://www.netmotionwireless.com. *(news - alert)*

In Enterprise Wireless, One Technology Does Not Fit All

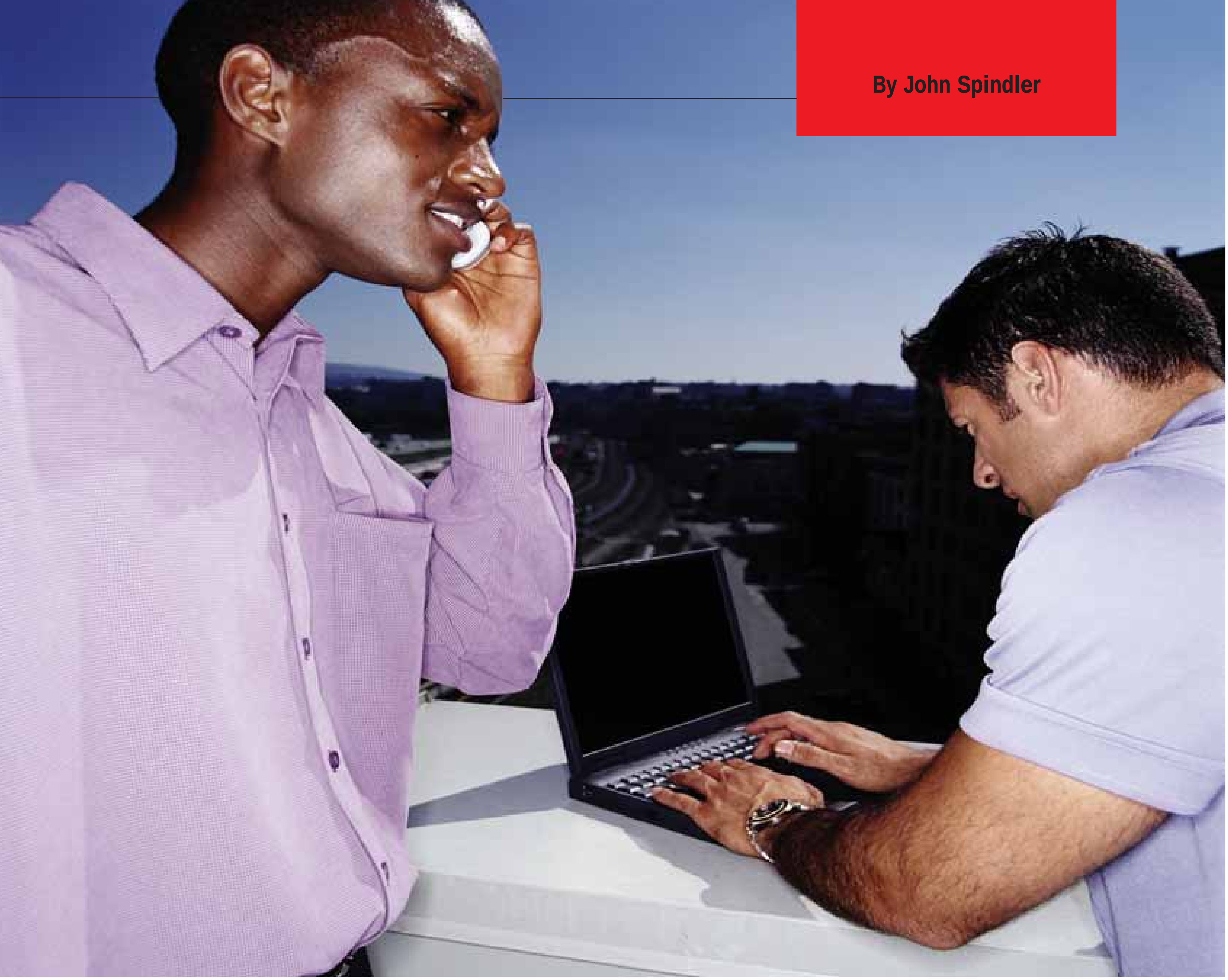
The industry regularly produces new wireless technologies such as 2.5 and 3G mobile, 802.11, and UWB, so it's ironic that enterprise IT administrators often hope that just one of them will solve all of their problems. Take WiFi, for example. As enterprises deploy WiFi to support data network access, many IT administrators are looking at voice over WiFi (VoWLAN) as a way to put all voice and data traffic on the same IP-based network. Theoretically, this strategy should reduce management costs and enable "free" voice calls over the corporate network. But today, VoWLAN leaves a lot to be desired when it comes to scalability, QoS, and cost reduction. The reality is that certain wireless technologies are simply better for specific applications, and trying to make one technology suit every application can be a costly and time-consuming exercise in frustration.

To evaluate the pros and cons of VoWLAN (define - news - alert) and 2.5/3G mobile wireless technologies when it comes to meeting corporate mobility needs, it's important first to understand that WiFi and 2.5/3G cellular are two fundamentally different technologies with fundamentally different design goals.

WiFi and VoWLAN Technology

WiFi (802.11) was designed to support IP data connections within a limited area. To cover large areas, it's necessary to deploy multiple WiFi access points (APs). Multi-AP networks theoretically deliver up to 54Mbps of bandwidth and have centralized management that simplifies AP configuration and security, but WiFi remains a fairly primitive technology as radio networks go, and VoWLAN is in its infancy. Consider:

- Although an 802.11g AP has a proclaimed data rate of 54Mbps, this is only achieved for a single user close to the antenna. As the user gets farther away, the AP rapidly drops to lower rates. At the fringe of a coverage area, bandwidth is typically limited to two to six Mbps.
- Multiple users in the same cell contend for the same bandwidth. Some packets collide and have to be retransmitted, so real throughput rarely exceeds 10 percent of the proclaimed rate.
- While IP data packets can be easily delayed a short period without affecting performance, IP voice is a very time-sensitive application, and packet delivery delays will reduce quality of service.
- To support voice, the AP must be "deloaded" to minimize packet collisions. Most APs cannot handle more than seven simultaneous IP voice calls with reasonable quality, and they achieve this level only when there is no other IP data traffic present.
- Using vendor-recommended engineering rules, a mixed voice/data network of APs in a typical office requires approximately four times the number of APs as a data-only network.
- APs cannot distinguish between IP voice packets and regular IP data packets today, and cannot independently control the quality of service (QoS) for each type of traffic. Although the industry anxiously awaits the ratification of the 802.11e standard (which will introduce QoS capabilities), current QoS mechanisms are primitive and proprietary.
- WiFi uses the public USM band, which is shared with cordless phones, microwave ovens, other unregulated devices, and even the WiFi network in



By John Spindler

the neighboring office. Any of these can interfere with your own WiFi network and lower its performance (and the first service affected is IP voice, because it's the most sensitive to disruption).

• Although WiFi-only phones are available for as low as $100, users are reluctant to carry an extra handset along with their cellphone. The optimum solution is a dual-mode (VoWLAN/cellular) handset, but today there are only a couple of these on the market and prices are around $600. These prices aren't expected to reach the same range as high-quality cellular phones for two years.

• Even if the previous challenges were all resolved, there's still the problem of interconnecting WiFi and cellular networks. Most industry analysts don't expect seamless WiFi cellular interconnection until at least 2008.

2.5 and 3G Mobile Technology

In contrast, today's 2.5 and 3G mobile networks were designed to support data rates from 300–400Kbps to two Mbps, along with voice traffic for millions of users over very large geographical regions. Because of these requirements, these cellular networks handle voice and data traffic much differently.

• All base stations are connected to a base station controller, which knows about each active user and coordinates client access among them to balance the overall client load. If a base station is unable to handle a client request, it hands that client off to the next nearby base station.

• Base stations control QoS in both directions to every client.

• One 3G base station radio can handle anywhere from 30 to 85 simultaneous users without QoS degradation, and base stations are engineered to support multiple radios, allowing hundreds of simultaneous voice calls and data sessions.

• Base station controllers automatically coordinate channels among their connected base stations to avoid interference. Cellular frequencies are dedicated to each wireless carrier, so there is no interference between networks. In fact, such interference is prohibited by FCC regulation.

• Voice and data traffic are handled separately, rather than contending for the same bandwidth.

Despite these advantages, in-building cellular coverage is often poor. Steel, concrete, and other building materials block or absorb the radio energy, causing lower QoS or bandwidth and even dropped calls as users move throughout a facility. However, there's an easy remedy for this problem: mobile wireless extension technology.

Mobile wireless extension technology has been around for years, and many companies already use it to improve cellular service within their facilities. In-building wireless extension gear includes a rooftop antenna (if necessary), a micro base station in the wiring closet or communications room, expansion hubs on each floor (if needed), and distributed remote antennas to extend radio coverage. By extending in-building cellular coverage, companies can immediately provide high-quality mobile voice services to any number of users. In addition, costs for this solution can often be shared with the cellular carrier.

Enterprise Wireless Goals and Requirements

With these technical differences in mind, let's look at the goals and requirements for enterprise wireless deployments. Basically, enterprise IT managers want wireless systems to:

Provide seamless voice and data coverage throughout their facilities.

As the wireless user population grows and companies adopt applications such as voice or RFID, companies must enable access anywhere.

Ensure the highest possible quality of service.

Voice users, in particular, are sensitive to poor connections, and QoS is more difficult for voice because users are often moving.

Minimize IT administration and equipment costs.

Any wireless network requires additional equipment purchases, but it is far less expensive to deploy WiFi for data applications than for voice applications, because wireless data requires fewer APs and less QoS management.

Minimize network services and per-minute costs.

WiFi networks avoid carrier network usage charges, but any costs involved with deploying, configuring, managing, and maintaining a WiFi network must be factored into the total cost. Along with APs and controllers, it may be necessary to purchase new wireless LAN cards or client devices that support higher data rates (802.11g or a).

VoWLAN Versus In-building Cellular

Considering the business requirements, it's relatively easy to evaluate the effectiveness of an all-in-one (i.e., WiFi data and VoWLAN) solution against a combined WiFi data network and 2.5/3G mobile voice solution.

Seamless voice and data coverage.

Since data users will likely be seated someplace, WiFi data can be used to provide access for this application. But for voice, in-building 2.5 or 3G cellular provides more pervasive coverage at a lower cost.

Ensure the highest possible quality of service.

WiFi delivers higher bandwidth for data applications for stationary users, while cellular offers far higher QoS for voice users. In addition, enterprises are increasingly adopting mobile applications for today's smartphones, and the applications are designed to function well under the constraints of 2.5 or 3G cellular data bandwidth. For instance, Salesforce.com (one of the most popular CRM applications on the market) allows the mobile salesperson to see critical data on a RIM Blackberry. (news - alert)

Minimize IT administration and equipment costs.

WiFi data networking equipment is relatively inexpensive and easy to manage, while problems with QoS, interference, scalability, and handset costs make

It's relatively easy to evaluate the effectiveness of an all-in-one solution against a combined WiFi data network and 2.5/3G mobile voice solution.

VoWLAN very expensive to deploy and manage. By extending cellular coverage indoors, companies can leverage existing infrastructure and client devices, which are largely managed by the carrier at no cost to the enterprise.

Minimize network service and per-minute charges.

For VoWLAN, the company must bear the cost of replacing cellular handsets with dual-mode handsets, and the WiFi network must be upgraded with up to four times as many APs to support even modest levels of voice coverage. But in-building wireless equipment costs can usually be shared with the carrier or building owner, and most cellular carriers now have rates as low as $39.95 per month for corporate users with unlimited mobile-to-mobile calls within the same network.

In summary, it's clear that VoWLAN today is a lot more complex and costly than simply extending cellular wireless with an in-building system. At the same time, VoWLAN can't deliver anywhere near the quality of service that voice callers demand. In-building mobile wireless extension allows companies to offer quality voice services throughout their facilities without major equipment expenditures, and to support the growing array of mobile applications designed for today's smartphones. When it comes to supporting mobile data and voice services, one technology does not meet all application requirements. IT

By John Spindler, vice president of marketing, LGC Wireless. For more information, please visit http://www.lgcwireless.com. (news - alert)

Fixed-Mobile Convergence:

Moving from Theory to Reality

The year 2006 will likely see fixed-mobile convergence (FMC) move from industry theory to a closer reality. FMC, the cross pollination of wireless and wireline services and providers, will find wireline service providers no longer strictly married to landline networks. Instead, they will compete and partner with the wireless providers that continue to gain market share with 'sticky' services such as video and personalized ringtones. What we'll see is a true blurring of what is considered wireless and what is wireline, moving ever closer to the day when that distinction becomes meaningless.

Driving the move toward FMC is a merging of a number of trends, including subscriber churn and loyalty, next-gen telephony services, revenue generation possibilities and infrastructure technological advances. When combined, these strands are creating the case for current investment by carriers that will lay the groundwork for true FMC in the next few years.

Much of what is driving FMC is nothing more than traditional market forces of supply and demand. At one time, services such as voice mail, call forwarding, etc., were common for wireline, but were considered advanced features by mobile users. That time has long since passed, and mobile service providers are now heavily concentrating on increasing the value of their networks via revenue-generating, next-generation services. Mobile subscribers have quickly become very accustomed to having these services at their fingertips, while also enjoying the flexibility of mobile telephony. The result: more and more users are leaving the world of landline communication.

This could be disastrous news for traditional telephony providers that have yet to take part in the technological evolution. The future telecom windfall will come from advanced services like full-motion video. For that type of service to succeed, the provider needs to guarantee delivery and optimum user experience through broadband and IP technology.

Optimum user experience is evolving in the form of dual-mode phones, which will allow users to have one phone for both the home and mobile use. This will put users in even greater control over who will emerge as a leading provider, as they will demand flexibility of location, premium content without any latency or performance issues, and they will require personalization. The world of telecom is moving from a network-centric view of capabilities to a subscriber-centric view of demands. In the past, subscribers' choices were limited to what the network was designed to do. Now, the subscriber is demanding more and more content and services, forcing a new network designed to support myriad capabilities and content. The type of services being delivered are not the traditional network-centric single services (call forwarding) but subscriber-centric offerings (e.g., presence and buddy lists with conferencing and messaging.) This is also blurring the lines between wireline and wireless, and we will continue to see partnerships in the market where providers on both sides will join to leverage their strengths and keep customer churn to a minimum.

Beyond emerging features such as video and ringtones, what other kind of personalized services are we expecting to see? Services such as personal ring back or location-based (concierge) services are already rapidly gaining popularity among subscribers of the world's forward-thinking service providers. A personalized ring-back service may take the form of an advertisement, whereby a department store would notify a sub-

By Harold Klett

Much of what is driving FMC is nothing more than traditional market forces of supply and demand.

scriber of a sale, while a location-based service may include a restaurant — in the city where the subscriber happens to be — sending a menu and/or video clip of the dining room to a mobile handset.

Other popular revenue-generating, enhanced services include, but are not limited to, the following:

Prepaid Mobile Gaming — Gaming services enthrall and captivate subscribers because of the nature of a highly interactive, personal and media-rich experience. The ability to inject tones, sounds, voice, and both Web and media content into riveting peer-versus-peer prepaid gaming services will engage subscribers by offering a rich and dynamic service experience. Here subscribers can identify other available subscribers via presence and engage them in an online game, simultaneously talking to each other while playing.

Mobile Centrex — This capability consolidates enterprise voice capabilities into a hosted Centrex offer. In addition, dual-mode handsets allow the employees to use one phone on-premise and remotely while carrying the same feature set thereby increasing productivity and reducing cost.

Missed Call Alert — This service detects missed calls in a wireless network and sends a text message to alert the customer. The customer can then call the originator back. This existing service can be extended to add visual indication of the caller or include video messaging.

Welcome Roamers — The Welcome Message represents an attractive family of services, as service providers can capture roamers and log them into their networks. The Welcome Message can be sent periodically to the roamer during his/her stay in the visiting country, and can contain useful information such as weather forecasts, currency exchange rates, and local favorite restaurants. Service providers can reach out to both the inbound and outbound roamers with customized information, thereby improving the service experience for the mobile roamer and increasing their loyalty toward the service provider.

Location-Based Services — This service allows mobile operators to offer enterprises a chance to "push" their company's presence to mobile subscribers who are in the same "area." Whenever a mobile subscriber moves between two different "areas" of the mobile network, the Mobile Station (MS) sends a Location Update to the network. When the application detects the Location Update, it searches the database for advertisements to "push" to this customer.

As new types of services built with rich-content and multimedia are deployed, the underlying infrastructure must allow access to and from any device. Services must be deployed in an environment where they can be flexibly adapted to changing subscriber demands, starting small to minimize risk and cost, and then scaling as subscribers grow. Networks need to be built using open standards that allow universal access and content delivery. Where will these standards and technologies come from? Internet and Web-based technologies that have been created and developed during the Internet revolution, built for interoperability and speed, will be the key.

As both the mobile and fixed providers look to avoid the headache of constant customer churn, they will see these new services as the path to increasing customer loyalty. Service intelligence needs to be maintained in this mixed network environment whether it's IP or TDM, fixed or mobile. The IP Multimedia Subsystem (IMS), defined by 3GPP, provides a standard architecture to address these requirements and is factored in as part of almost every network evolution. According to a recent industry survey, roughly 25 percent of carriers expect to see mass deployment of IMS technology in their networks before the end of 2006, while a further 38 percent expect this to happen in 2007 or 2008.

IMS has been defined as a common architecture/subsystem for providing innovative services to mobile and fixed networks. IMS is the architecture that merges the Internet with the cellular world, and makes Internet technologies, such as the Web, e-mail, instant messaging, presence, VoIP, and videoconferencing available nearly everywhere. These promising technologies include highly scalable, incredibly flexible infrastructure to deliver consumer-driving applications and services across both fixed and mobile networks.

The other infrastructure key to this puzzle is an IP-enabled media gateway, which allows networks to support the current, powerful set of applications on portable mobile devices, as well as a whole new world of speedy, consumer-oriented applications, including video-rich mobile gaming, mobile blogging, and instant sports score updates that would stream to and from SIP mobile terminals and existing infrastructure used by millions of consumers.

The nexus of customer demand and device capability, the market's recognition of the eventual convergence of traditional wireline and mobile networks through partnerships and new services, and the infrastructure technology now available through IMS and IP-enabled media gateways, has set the stage for fixed-mobile convergence to become a strategic initiative now and a reality in the not-too distant future. IT

Harold Klett is vice president of product management at Excel Switching Corporation. For more information, please visit the company online at http://www.excelswitching.com. (news - alert)

Fixed Mobile Convergence and the Evolution to IMS

To compete with Internet-based VoIP providers, the challenge for service providers today is to deliver a robust suite of multimedia services that converge the Internet with the mobile network — cost effectively and profitably. Most service providers are looking towards Internet Protocol (IP) technology at the session layer to make end-to-end delivery of multimedia services cost-effective. For a true IP network to be realized, the most utilized application — wireless voice — must eventually be transitioned to become IP end-to-end. However, a number of issues prevent service providers from immediately migrating tens of millions of voice users to an all-IP service. Current bandwidth and latency constraints on cellular data access networks and a large investment in IP technology in the core network is required to replace the circuit-switched SS7 and Intelligent Network (IN) technology used by voice.

Instead of this overnight, wholesale conversion to IP, an evolutionary process is needed in both the core and access networks to achieve the all-IP goal. Today, a business case for service providers transitioning to an all-IP network for multimedia service transport remains uncertain. However, voice, as the initial multimedia service, offers a proven revenue and profit solution whose value is substantially enhanced by fixed-mobile convergence — by seamless, single-number reach anywhere, anytime. The key is building the bridge with fixed broadband and unlicensed wireless VoIP to achieve rapid user adoption that accelerates the business case for an all-IP network deployment based on the current base of voice users.

Service providers can justify the investment in an all-IP network that conforms to the emerging IP Multimedia Subsystem (IMS) (define - news- alert) standard — using voice as the killer application.

Is There Demand for IMS?

Mainstream adoption of multimedia services is underway. Today, consumers and enterprises are beginning to purchase voice, data, and video services that are delivered across a wide variety of end-user devices and over different networks, both mobile (CDMA/GSM) and fixed (DSL, Cable Broadband, WiFi). As a sign of the growing importance of multimedia over non-traditional platforms, the National Academy of Television Arts & Sciences plans to award a new Emmy category for original programming created specifically for computers, mobile phones, PDAs, and similar devices.

But as the number of separate services, devices, and networks increases, consumers also want a simple and consistent 'seamless' experience with shared information across those networks. To meet this demand, many service providers are considering major investments to support fixed-mobile convergence (FMC) services. Pyramid Research has estimated that traffic moving over a converged fixed-mobile network will grow to over $80 billion in revenues by 2007.

The challenge for today's service providers is to deliver a robust suite of converged multimedia services, cost effectively and profitably. Most service providers are looking to IP technology

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By Sanjay Jhawar

at the session layer to make end-to-end delivery of multimedia services cost-effective. For a true IP network to be realized, the most utilized application — wireless voice — must eventually be transitioned to become end-to-end IP. However, there are issues! Current bandwidth and latency constraints on cellular data access networks (EVDO Rev A, 3G) would severely degrade voice service. In addition, in the core network, a large investment in IP technology is needed to replace the circuit-switched SS7 and Intelligent Networking (IN) technology used by voice.

Today, the business case for wireless service providers transitioning to an all-IP network for multimedia service transport remains uncertain. Despite the number of new multimedia service trials, it is too early to be able to clearly make the case for an all-IP migration based on what is currently known about the adoption rates and profitability of other multimedia services, such as Push-to-talk over Cellular (PoC), multiplayer gaming, and mobile TV.

However, voice is a service with a proven revenue and profit potential, whose value is substantially enhanced by FMC. The explosive growth of the Skype-Out voice service that converges IP technology with the PSTN demonstrates the strong market demand for converged voice services — allowing Skype to monetize usage in addition to advertisement. At the time of writing, Skype claims five percent of its user base, or about three million subscribers, pay to use Skype-Out for Internet to circuit-switched voice calls. The key, therefore, is building the bridge applications that converge current circuit–switched voice services with fixed broadband/unlicensed wireless VoIP (WiFi) to the cellular network to achieve rapid, Skype-like, user adoption. As Skype has proven, this can accelerate the business case for an all-IP network deployment based on the current base of voice users.

IMS Requirements and Definition

When making the transition to an all-IP network, service providers must keep in mind subscriber demand for seamless functionality and consistency across multiple service provider networks and types. To achieve this, the mobile industry has developed a standard for all-IP operations called

Internet Protocol Multimedia Subsystem (IMS). IMS is defined by the third Generation Partnership Project (3GPP) standards body. The IMS standard promises to allow service providers to manage a variety of services that can be delivered via IP over any network type — including the mobile network's packet switched domain (GPRS, 1xRTT, EVDO, 3G). With IMS, service providers will use IP to transport both bearer traffic and Session Initiation Protocol (SIP) for signaling.

IMS is a framework for managing robust applications (services) and networks (access) providing multimedia services. IMS defines, among other things, an "Application Server" to be the network element that delivers services that subscribers use. IMS manages applications by defining control components that all Application Servers must use and share. These shared control components manage "housekeeping" matters, including subscriber profiles, IMS mobility, network access, authentication, service authorization, charging and billing, interoperator functions, and interoperation with the legacy phone network.

While multimedia services (applications) will operate more efficiently and seamlessly using IMS, many multimedia applications can operate today on a stand alone basis, but control and service systems are dedicated to each multimedia service (increasing operations cost, and limiting bundling flexibility). IMS generates efficiency by consolidating common functions that can be shared by many applications by enforcing common policies (e.g., Quality of Service) across a wide range of applications. The goal of IMS is to make new service introductions faster and less expensive by leveraging IP technology, while providing a consistent and controllable user experience.

Why Deploy IMS and How?

Service providers seem to have embraced a vision of delivering robust multimedia applications that will help differentiate their service from their competitors — helping to reduce churn to the 'Skypes' of the world. To achieve this vision, service providers desire to deploy end-to-end IP-based multimedia services via IMS. However, since many of the multimedia applications are new and unproven, it is difficult to prove the business case for a network-wide IMS deployment before these new revenue-generating services can be rolled out and assessed. As such, service providers are faced with the difficult choice: either invest in a new network architecture before demonstrating its rate of return, or do nothing and potentially lose ground to competitors.

However, there is an alternative. Instead of a network-wide IMS deployment and investment, service providers can initiate phased deployment. Service providers can lower their risk by deploying and launching FMC voice and messaging services to prove the business case for network-wide deployment of IMS.

The revenues and profits associated with a known service, like voice, will demonstrate the potential for further investment in the IMS architecture. Furthermore, the investment in FMC capabilities can be leveraged to launch many additional data and multimedia services. Subsequent FMC services benefit from the existing infrastructure and achieve profitability faster.

Key points of consideration for service providers exploring the pre-IMS to IMS evolutionary path include:

1) Network Convergence: Bridging legacy 2G and 2.5G networks with IP broadband networks and allowing user attachment to these networks to ensure delivery of a variety of voice and data applications (including SMS, MMS, Message Waiting, and more).

2) Single Number: Supporting single number identity across any SIP device (PC, mobile device, dual-mode device, etc.) that is best suited to an end user's location, interface preference and circumstance.

Voice is a service with a proven revenue and profit potential, whose value is substantially enhanced by FMC.

3) Handover: Offering dual-mode handover between cellular and WiFi networks.

4) Enterprise Connectivity: Connectivity options to seamlessly expand an enterprise extension to the mobile network via SIP and Gateway MSC functionality to popular IP PBX and IP Centrex systems.

Conclusion

In summary, the 3GPP/3GPP2-based and IMS-evolvable approach allows new revenue generating voice and messaging fixed mobile convergence services to be easily and quickly created and tested in pre-IMS target markets before a service provider invests in a full IMS network deployment.

To reduce implementation cost and risk, and to accelerate time-to-market, service providers should seek infrastructure solutions that are future proof and fully IMS-compliant. Any solution needs to be designed to bridge existing 2G and 2.5G networks with the IMS — maximizing new revenue generating opportunities, to allow mobile to WiFi handover to be quickly and cost effectively implemented, with a minimal number of new network-based IMS 'boxes' — maximizing network operational efficiencies. It also should integrate easily with existing and future SIP-based enterprise capabilities — making it the platform of choice for deploying new enterprise-based services. IT

Sanjay Jhawar is senior vice president, marketing and business development at BridgePort Networks. For more information, please visit the company online at http://www.bridgeport-networks.com. (newls - alert)

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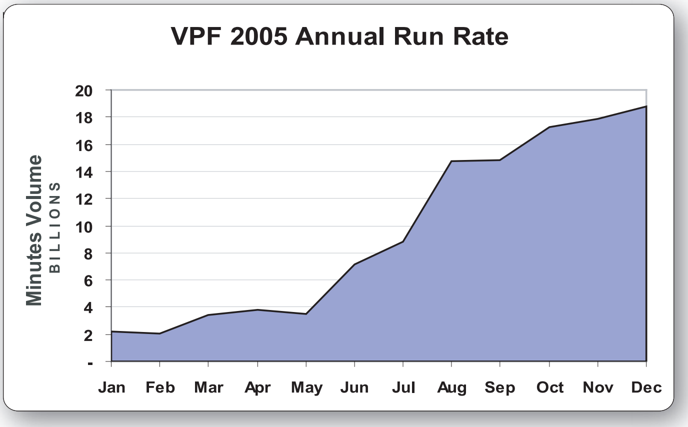

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VoIP Fraud:

Scenarios and Solutions

Voice over Internet Protocol (VoIP) (define - news - alert) services are clearly gaining popularity in both the consumer and enterprise markets, and with good reason. New peer-to-peer and independent service providers, such as Vonage and Skype, offer the opportunity for users to reach out and touch someone for a fraction of the price of a phone call using the traditional public services telephone network (PSTN). In fact, adoption of VoIP as an alternative to PSTN is growing so quickly that the Yankee Group forecasted the number of users at the end of 2005 to be 3.2 million people, which is triple the number at year-end 2004. Recent forecasts from Atlantic-ACM, a telecommunications industry research firm, show the enterprise VoIP market is expected to grow at a CAGR (2003-2009) of 47 percent, with consumer VoIP revenues poised for triple digit growth.

The potential profits for those service providers offering VoIP services, and the opportunity for consumers to take advantage of inexpensive international and domestic long-distance phone calls sounds like a win-win. But, with the increased use have also come scams and security concerns. Unfortunately, today's Internet reality is that when a new service grows in popularity, there are those who endeavor to exploit the service for personal gain, or to disrupt legitimate services for others, and VoIP is no exception.

Today's VoIP Makes Fraud Easy.

Imagine sitting down at the local coffee shop with your favorite mocha and deciding to catch up with a good friend in Europe. To save on a costly cell phone bill, you connect your laptop up to the wireless hotspot and log into your VoIP softphone to make the call for pennies a minute. Unbeknownst to you, a casual onlooker checks out your username and password as you login. That person doesn't just have a few free phone calls in mind, instead they send your information to their friends and acquaintances to use with their freely downloaded VoIP phone software from the same provider. The number is eventually shut down, but not before a number of calls and a huge phone bill is rung up. Without better security for VoIP, this technique is so easy to reproduce that it will become an all too common scenario. But does the consumer or the service provider pay the bill?

Not only is this abuse case easy to recreate, but the array of alternative methods is overwhelming. For example, instead of peering over your shoulder, anyone sitting just outside the coffee shop with a laptop and a wireless card can easily monitor all of the network traffic and record your call, as well as gain detailed connection and account information. Many VoIP services don't enable encryption or authentication, opening up the opportunity to acquire account credentials entirely from monitoring the WiFi hotspot.

Then there is the reported case of fraud that's recently been in the news where an Austrian retail business entity resold VoIP minutes from a service provider here in the United States. The European business was billed for usage to the tune of $20,000, but what the U.S. VoIP provider did not know was that those calls were to a European equivalent of a fee-based 1-900 number service. By the time the U.S. VoIP

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By Peder Jungck

provider received invoices for $450,000 from the 1-900 phone services, the European 'business entity' was nowhere to be found.

How Do We Make VoIP Safe?

Many experts agree that this is going to get worse before it gets better. IP Telephony has inherited all of the potential for exploit that both the telephone systems and IP data services have traditionally possessed. But while wireline and wireless telephony services and IP data services have had many years to be hardened against abuse, VoIP is relatively new with characteristics that require protection beyond that for standard telephony and IP data services.

There are three main areas of fraud in VoIP being observed today, and some very specific ways that service providers can protect themselves.

Identity theft and account cloning

Identity theft has been on the rise in recent years, but nowhere is it more prominent than on the Internet. To combat identity theft and verify users, stronger authentication systems can be put in place. With strong authentication, just because someone knows a user's phone number and possibly their password, they would still not be able to access their account. To strengthen an authentication system, service providers could modify softphone client software by embedding user information into the application before download, making it unique to that user. Alternatively, service providers could require users to declare unique hardware details for computers they intend to run softphones on, such as MAC addresses. If the user wasn't calling from a registered address, VoIP service gateways mechanisms could block usage by user. While it may take away the some network roaming freedoms, it could be an optional safeguard and may even contractually identify whether un-authorized use is the service provider or the consumer's burden. The wireless industry has spent millions to protect cell phones from being cloned; VoIP phone cloning has few protections today.

Billing fraud

Enabled by the borderless nature of the Internet, a VoIP caller can use unauthorized billing or credit card details from anywhere in the world. To combat billing fraud at the service provider level, even though the location specifics of the caller aren't known, call authorization can be enforced and verified at gateways. This can include identifying which called numbers have fees associated with them versus which do not, and whether or not the provider is willing to pay that fee. Additionally, service providers can block calls to and from services where there is no prior agreement in place and certain providers can be 'blacklisted' until they work out a billing agreement. Gateway systems exist that can sit in front of the VoIP switching infrastructure, track each call set up and tear down and validate destination/origination authorization in real-time. And VoIP service policy gateways are equally applicable between the client-to-service provider link and between wholesale providers at a peering connection.

Call Hijacking

Vulnerabilities in the SIP protocol used in most VoIP services today enable hackers to inject control signals and hijack calls. Most VoIP phone systems have sophisticated call control mechanisms such as three-way calling and call transfer. By leveraging these features, someone outside of the calling parties can monitor a call in progress and hijack the session by transferring both ends of the call. Since commands are often sent un-encrypted without authorization for each command, this is an easy technique to carry out. Simple call hijacking, and even call eavesdropping, can be avoided by encrypting VoIP communications. By encrypting data, the hijacker isn't able to gather information about the call itself to determine how to inject the control message. Encrypting calls can introduce a cost and performance burden on the service provider and the end user equipment, but some promising alternatives exist. A less intrusive method is for service providers to introduce active VoIP policy controls in front of their call session control function (CSCF). This gateway solution can block or challenge call control messages for authenticity before passing them along to the CSCF. Since most consumer VoIP services don't require call transfers, this technique can effectively resolve a large problem with no change to the call servers or consumer software.

IP Telephony has inherited all of the potential for exploit that both the telephone systems and IP data services have traditionally possessed.

As we've discussed, remedies to some of the more common fraud practices in VoIP do exist, with some being more burdensome for the service provider than others. In most cases, the remedy with the least impact on service provider cost and network performance is a high-speed VoIP service policy gateway. Suitable service policy gateways analyze VoIP control signaling and media traffic at very high speeds, actively enforce required VoIP service security policies, and can protect against a wide range of fraudulent activity. Additionally, service policy gateways must be able to adapt their methods to constantly changing techniques and security threats. Positioned at peering points and/or in front of VoIP call session control infrastructures, these VoIP service policy gateways can protect both subscribers and service providers.

VoIP has a lot of promise, and offers a great new world, but it is inheriting known issues from previous technologies, so service providers would be wise to relearn old lessons. Now is the time to do that. VoIP is in a rapid adoption phase and there is little doubt its use will continue to increase.

Inconsistent implementations by different vendors and protocol vulnerabilities will be resolved with time, and services will become hardened as a result. Right now however, the vulnerabilities are like a neon sign for hackers. Lucky for all of us solutions do exist to stave off some of the fraudulent practices at work on today's VoIP networks. IT

Peder Jungck is founder and CTO of CloudShield Technologies, (news - alert) a provider of multi-gigabit, multi-function, programmable deep packet inspection (DPI) platforms targeted at large network operators. For more information, please visit http://www.cloudshield.com.

Advanced VoIP Testing Techniques

Maintaining VoIP Service Quality Using Integrated, Active Testing

VoIP has become a household word. Cable MSOs and Telcos are offering VoIP home phone service to a very receptive public seeking alternatives to their higher-priced, inflexible plain old telephone service (POTS). But paying subscribers sign up with expectations — providers need to offer wireline quality and reliability, efficient installation, and competitive pricing to differentiate their services from free, Internet-based voice messaging applications and low-cost cellular plans.

Some providers are making the grade — a recent international call quality study shows that Cable VoIP using low-compression CODECs actually sounds better and connects calls faster than traditional PSTN (define - news - alert) service. As consumers spread the word about VoIP as a cost-effective, high-quality alternative to traditional phone service, service providers will experience ever-stronger demand.

Maintaining service quality during large-scale deployment, sometimes to millions of subscribers, is a key challenge operators must tackle to capitalize on VoIP's early success. As high-priority, real-time voice packets begin to fill networks, quality must be monitored and optimized or service quality will quickly deteriorate.

Successful service providers plan large-scale VoIP deployment using a comprehensive, service-quality focused strategy. A recent Business Week article (Nov. 28, 2005) outlined how Comcast's rapidly growing VoIP service maintains PSTN-quality using unique provisioning and monitoring testing. Comcast's test strategy allows them to offer a 30-day satisfaction guarantee to new subscribers — an approach that has led to rapid customer growth. Operators who leave service quality to chance have been less fortunate, experiencing a high rate of customer churn as subscribers return to trusted incumbent providers. In a highly competitive industry, quality is a proven differentiator and a key asset to survival.

Actively Testing VoIP

To ensure effective VoIP deployment and continuing service quality, providers rely on passive and active test solutions for provisioning, monitoring and service level fault management. Passive and active telephony testing techniques were first developed by carriers over 20 years ago to measure the quality of digital PSTN voice networks. In traditional telephony, passive test systems are used to measure signaling performance, while active bit-error-rate (BER) testing is used to monitor PSTN call quality. The same techniques have their analogs in the VoIP world. Passive testing involves either collecting traffic statistics from softswitches, media gateways, and other network elements, or requires placing test probes at strategic network locations to "sniff" live IP traffic and filter out VoIP packets. A variety of algorithms examine packet header and payload information to calculate call signaling performance and some basic quality metrics like MOS, packet loss, jitter, and latency. Some passive solutions also report the breakdown of coexisting email, web, peer-to-peer, file transfer, and multimedia traffic. Passive testing is commonly used for basic service quality monitoring, and to assess network usage for capacity planning, performance and routing optimization.

Passive testing solutions have known limitations, some of which are specific to VoIP applications. Since passive testing relies on measuring live VoIP traffic, it cannot be used for provisioning or pre-deployment testing, necessary to ensure customers expectations and service level agreements will be met after installation. Security concerns around VoIP have resulted in increased packet-level encryption, rendering them invisible to passive test systems. Adding to the problem, probes can only analyze traffic on an operator's IP network, providing limited visibility into the majority of calls that take hybrid routes over partners' PSTN, IP, and mobile networks. Since call quality is affected by analog/IP conversion (CODECs), echo cancellers, customer premises equipment, wiring, firewalls, encryption algorithms, and network

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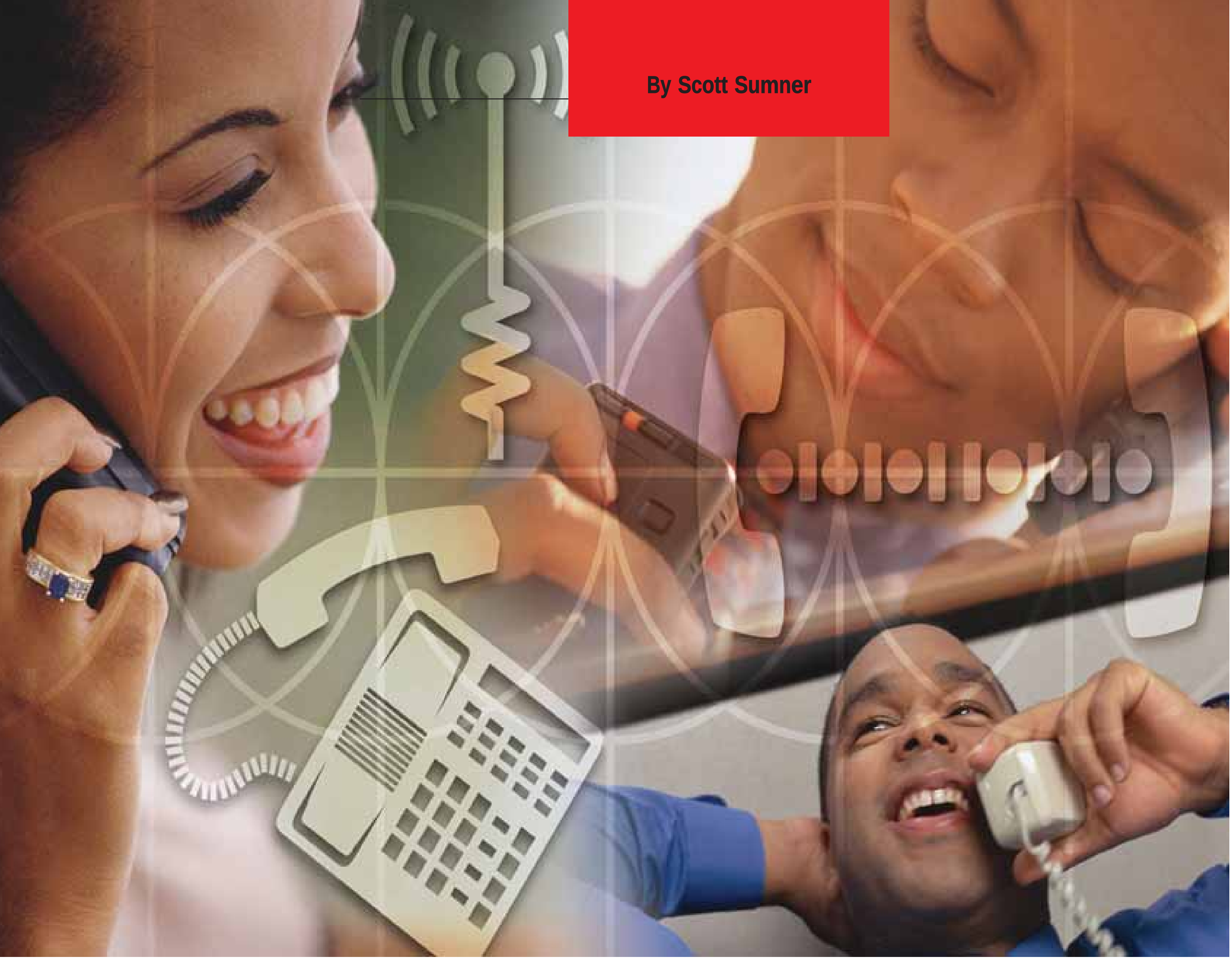


policies outside the reach of passive probes, measurements from these systems cannot accurately represent user-perceived end-to-end call quality.

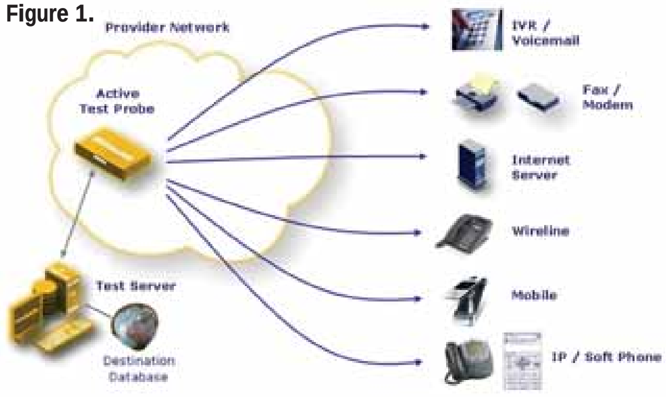
As a result of these limitations, VoIP providers often follow the example of PSTN operators and use a combination of active and passive testing techniques for monitoring and fault management. Whereas passive testing offers good visibility into call *signaling* performance, active testing offers operators the best possible technique to analyze end-to-end *media* quality, the true end-to-end customer experience of service quality. In active testing, probes place short test calls that are recorded then analyzed by industry-standard algorithms to produce a detailed, ear-to-mouth service quality assessment. Active testing can be used with any standard telephony interface including analog (2 wire), ISDN, mobile, or IP. Test calls can be dialed using SIP, H.323, MGCP, PRI, SS7, CAS, or mobile signaling standards. Because active testing is not limited to IP networks, it can be used to test calls placed over hybrid routes — for example from a cellular, through a PSTN network, to a VoIP phone — measuring end-to-end speech quality, call setup and session information. Active testing works well for provisioning and pre-deployment audits where test calls are placed alongside existing or simulated network traffic. Using automation, test plans can be scheduled to execute at regular intervals for trending, monitoring, and service level fault management. On-demand test calls can be used for remote troubleshooting.

Active Test Layouts

Test solutions are primarily designed to help operations staff and technicians find and fix problems before they are experienced by customers — like radar or night-vision binoculars, they help to identify, visualize, and isolate problems that would otherwise be impossible to detect. But just as you wouldn't use binoculars to read a book, different departments need different tools to address their specific test requirements. A technician installing and troubleshooting VoIP in the field needs a portable, straightforward instrument that can quickly assess service quality from a user's perspective; operations personnel need tools to remotely identify and replicate service issues to speed service restoration and minimize technician dispatches. Operations require service-level fault management. Billing needs to validate and maintain service level agreements (SLAs) with straightforward reporting. Active testing addresses these various

Test solutions are primarily designed to find and fix problems before they are experienced by customers.

Figure 1.

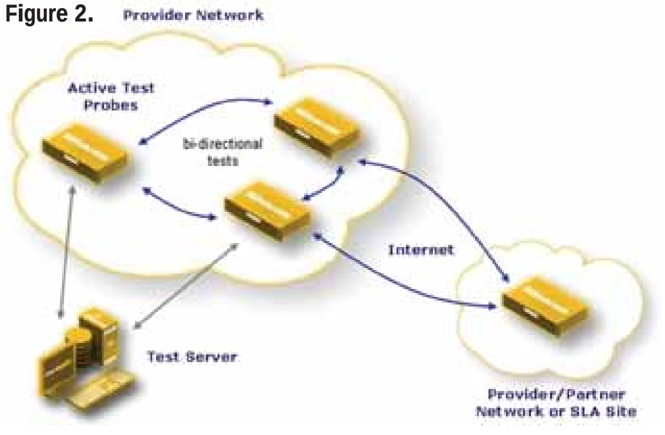


requirements using different test layouts, optimized for provisioning, troubleshooting, and monitoring. Test layouts include innovative test methods that can be combined to provide comprehensive coverage from subscriber to central office/headend, and all points in between.

Single-Ended Test Layout

The single-end (ping-style) test layout (Figure 1) measures voice, signaling, fax, Internet, and modem service quality without using a far-end probe. A test probe records call progress and content for playback and analysis. A location database provides public telephone, fax, modem, or IP destinations for testing on or off the service provider's network. Tests calls are designed to assess service quality quickly — minimizing long-distance billing charges by hanging up just after the connection is established. The test server typically reports results by area code, city, network provider or country. Single-ended testing is typically used by long-distance service providers to verify call quality over partner carrier networks, and to automate least cost routing processes. It can also be used for troubleshooting to specific phone numbers, and to routinely spot-check quality on geographically large networks.

Figure 2.



Probe-to-Probe Test Layout

In probe-to-probe testing (Figure 2), test calls are placed from one probe to another, providing complete control over test stream content and quality measurements in both directions. Tests can be initiated and received on any standard signaling interface (analog, PRI, SS7, mobile, SIP, etc.) over NextGen and legacy networks. Tests record call progress as well as the actual voice/data traffic for playback and signal analysis. Probe-to-probe testing is typically used to monitor VoIP, PSTN, video, Internet, and fax/modem service quality over large-scale IP, TDM, and mobile networks. Large providers will often place several dozen probes in POPs and central offices. Test automation is used to schedule network-wide test calling patterns between all probes in the network. When a service quality issue is identified, the operator can choose to focus test calls over the problematic route to quickly isolate and troubleshoot the problem.

Responder Test Layout

Responders replace far-end test probes when portability and cost are primary concerns (Figure 3). Responders are ideal tools for field technicians performing day-of-install testing and troubleshooting, and for regular customer-site SLA validation. Active test responders can receive and transmit test traffic, and are commonly connected to either 2-wire analog or IP ports. Analog connections allow technicians to replace a telephone handset with a responder to measure user-perceived quality. Residential VoIP installation usually involves using an analog responder to quickly test call quality from all the phone jacks in a home, measuring MOS, distortion, echo, and call connection statistics, packet loss, round trip delay and DTMF tone transmission transparency to ensure the customer will be satisfied with the new service. IP responders are usually used to test RTP traffic streams for VoIP and video quali-

Figure 3.

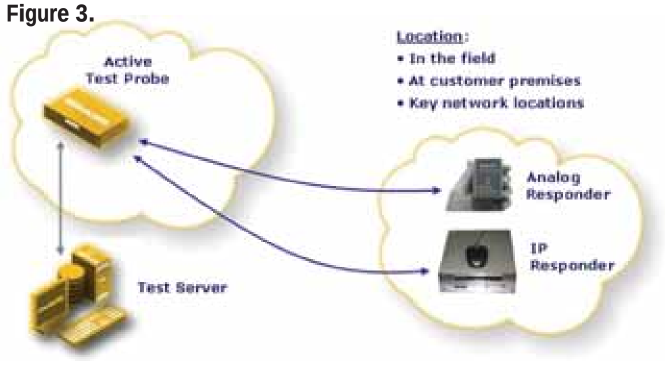


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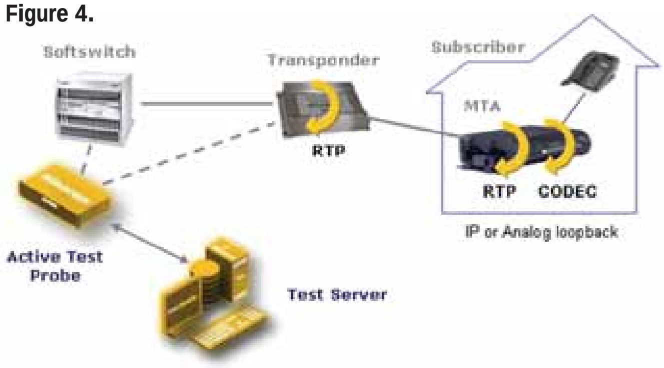
Figure 4.



ty assessment.

Loopback Test Layout

Integration with softswitches and network components allows active test systems to conduct analog and RTP/IP loopback tests from test probes to CATV transponders and subscribers' multimedia terminal adapters (MTAs). Loopback tests (Figure 4) reduce the need to deploy technicians to resolve network and service performance issues, and require no investment in portable test equipment. Loopback testing is often used by Cable MSOs to streamline operations and reduce support costs, and to accelerate problem resolution. Loopback testing can quickly isolate a service issue by testing to a subscriber's home, neighboring homes, and nearby transponders. Data from RF-level transponder tests, IP, and analog loopback tests to the MTA can be correlated to remotely determine on what network layer the problem originates and what steps are required to correct the problem.

Interactive Test Agent Layout

Voice-guided testing is the most recent active test technology; the test probe hosts an interactive voice response agent that uses natural speech to guide technicians or subscribers through a series of simple steps leading to detailed speech quality, DTMF, echo and noise tests without any far-end responder (Figure 5). Designed to reduce installation, troubleshooting, and trouble ticket resolution time, interactive testing is ideally integrated into customer support systems to provide immediate test data when customers call with service issues. Test results can be used by customer service to guide the user through corrective actions, by operations staff to remotely correct the problem or to dispatch a field technician if field repair is required.

OEM Tester Layout

Some active test systems test from a probe to third party test sets, TDM responders and network elements as test endpoints. This functionality enhances the flexibility of the test system by leveraging existing capital investments. In some cases, active test systems can add VoIP, Video, and other service level functionality to physical and network-layer DSL, Cable, and FTTx testsets that would otherwise have to be upgraded or replaced. Tests can be initiated by the test server, the legacy operational support system, or by the handheld testset if they support integration.

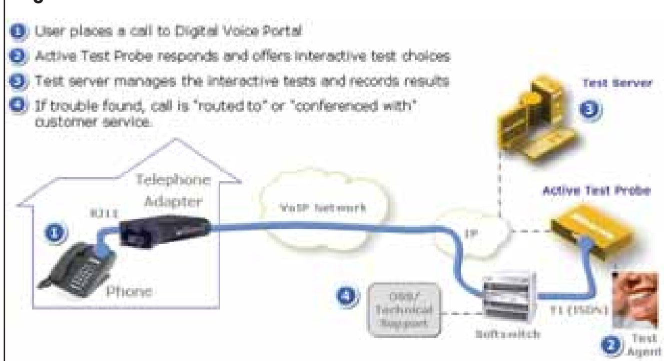
The Big Picture

A test automation platform is used to manage and integrate the results from the different test layouts and applications, centralizing all aspects of VoIP service delivery and maintenance. An integrated test platform enforces centralized standards and test plans, provides event correlation and reporting, and is easily integrated into existing operational support systems – allowing all departments to work and communicate within a single, quality-focused framework. With a single, integrated test platform, departments can efficiently collaborate to deliver the best possible VoIP service. For example, a technician troubleshooting VoIP at a subscribers' home would also have access to historic day-of-install test results, and NOC personnel can run remote tests to neighboring homes and share the data with the technician, effectively isolating the problem in time and space to accelerate problem resolution.

Despite the challenges faced by service providers deploying VoIP, it is possible to rapidly scale subscribers while maintain PSTN-grade service quality. An integrated, comprehensive test strategy that encompasses VoIP delivery from day-of-install to proactive service monitoring can be addressed using automated, active test systems. An effective service level test automation platform can integrate results from different test layouts, allowing all departments to work towards their respective goals within a common framework. With a test platform, a service quality focus can be integrated into existing operational support systems, fault management systems, and existing business processes — giving the visibility required to compete in today's competitive telephony landscape. IT

Scott Sumner is director of marketing at Minacom, (news - alert) a provider of Service Level Test Automation solutions for Telcos and Cable MSOs. For more information, please visit the company online at http://www.minacom.com.

Figure 5.



VoIP MARKETPLACE

VoIP MARKETPLACE

CEO Spotlight

Greg Welch
Chairman and CEO
GlobalTouch Telecom

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

GG: What is GlobalTouch Telecom's mission?

GW: In a nutshell, we are an ASP and a carrier's carrier and we enable our customers to leverage the benefits of VoIP by offering them the most cost-effective, compact all-in-one, powerful VoIP service on the market — anyone can come to us and we will put them in the VoIP (define - news - alert) business in 60 days or less.

GG: How did you get into the VoIP business?

GW: Well, we started out as long-distance service provider, which is a good solid business, but we were also aware that VoIP was evolving quickly and becoming an important factor. We started shopping around for a VoIP solution so we could become a reseller. As we researched the idea, we discovered that there was no comprehensive VoIP solution that suited our needs. The only avenue was to cobble together an offering mixing software and equipment from several vendors. This meant a lot of money and a lot of time to put together a solution with lots of finger pointing when something goes wrong or when we would want to introduce new features. Forget it.

Instead, we teamed up with a group of developers in Chicago, who were formerly with Webley Systems, and together we developed our own VoIP technology, including our own VoIP software, softswitch, SBC, even an IP-PBX feature and several other features. What we created was an all-in-one software-based solution that can serve 15k callers on one pizza box that costs us no licensing fee and can be scaled and upgraded at will. So, for us VoIP fostered a dramatic paradigm shift from a retail service provider to developing and reselling an integrated VoIP solution and becoming an ASP and a carrier's carrier.

> VoIP fostered a dramatic paradigm shift from a retail service provider to developing and reselling an integrated VoIP solution and becoming an ASP and a carrier's carrier.

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

GW: Well, here we are with a unique and powerful VoIP solution in terms of cost, features, and quality. The next logical step is to continue to develop applications that will change the way people communicate (e.g., moving to video and other unique applications, more features on our PBX to save costs and increase productivity, custom developed applications that enable our reseller customers to compete more effectively in their own markets, that very quickly pinpoint and fulfill a local market niche). This is what really goes to the heart of the next- generation promise as I see it.

GG: Now that it appears that growth and opportunity are the trends in the VoIP industry, what possible hurdles do you see that might upset this momentum?

GW: Technologically, I don't see any real obstacles. The only obstacles I see will likely be regulatory ones, fomented by lobbyists through the FCC as VoIP really starts to gain significant market share and consumer awareness — and ruffle feathers. Will we have a level

The future of VoIP is in services and killer apps that we have not yet even imagined.

playing field enforced by the FCC? I suspect the recent dust over 911 was probably the result of political maneuvering rather than "correct" regulation designed to protect consumers. If they really care about 911, why don't cell phones more accurately pinpoint the user's location. On the other hand, when the 911 issue was forced, it was just another chance for VoIP to demonstrate its flexibility, to adapt a feature as demanded by the market quickly and cost-effectively.

GG: What are some of the technology areas where GlobalTouch is increasingly focusing, and why are these areas important to the future of your company?

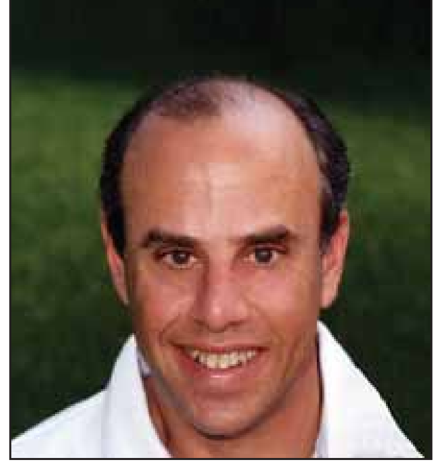
GW: We are focused on technology that helps us and our customers deliver any new feature, service or killer app, and pushing the envelope in terms of how quickly, flexibly, and scalably we can roll them out. Let me give you an example. We recently got a call from Sony Pictures' marketing department. The wanted to promote a new horror film, "*When a Stranger Calls*," by setting up a toll-free number for moviegoers to call and hear a "scary" call featuring an actor from the movie. They came to us after trying the traditional telco route. The quote they got was too much time for set-up at too high a cost. We, on the other hand, because we have our own VoIP platform, were able to set up and run the call with a few mouse clicks — we handled hundreds of thousands of calls during the movie's opening weekend without a hitch. So, in terms of VoIP technology, I think this is where we and the entire industry are headed — VoIP technology as a springboard for ad hoc, cost-effective services and applications — the horror film promotion was truly a "killer app."

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

GW: I feel that VoIP is the most exciting thing that we have seen in our industry since '92, and I think the buzz in the industry confirms this. We are only just breaking ground with VoIP Telephony; the future of VoIP is in services and killer apps that we have not yet even imagined. The excitement stems from that fact that VoIP is ushering in a new communications landscape of services and applications that will ultimately prove to be more profitable than voice, and will put the profit back in the voice by being bundled with it. VoIP is bringing voice back to its roots in a way where users will say a name into a device and it locates that person, no matter where they are or on what device, like people used to do via an operator. On the business side, VoIP is dramatically improving productivity, and reducing the cost of doing business. Field sales or service personnel armed with softphones on notebooks or even PDAs will click to dial ad hoc video conferences to review documents with the home office. This way, a contract negotiation or service call can be completed with considerable speed and efficiency, to name just one application in a field that is wide open. The point is that VoIP is poised to reap the benefits of an increasingly global economy and mobile society. And with the proliferation of hotspots, particularly the movement to municipally administered citywide hotspots, VoIP will allow users mobility, economy and flexibility that has never been seen before. IT

CEO Spotlight

Jonathan Shapiro
Chairman of the Board and
Chief Executive Officer
Alliance Systems



In the CEO Spotlight section in *Internet Telephony®*, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Jonathan Shapiro, Chairman of the Board and Chief Executive Officer of Alliance Systems. (news - alert)

GG: What is Alliance Systems' mission?

JS: Alliance Systems' mission is to design, manufacture, ship, and support communications and computing equipment that provide the infrastructure for VoIP, wireless, security, and other applications. Through our engineering, manufacturing, and value-added services, Alliance helps customers optimize their businesses by enhancing profitability and reducing time to market.

Our job is to help our customers get their products to market, support their channels of distribution on a worldwide basis, and support the complex implementations of VoIP (define - news - alert) and communications applications.

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

JS: Our vision is to focus on customers and technology. Our customers include OEMs, service providers, resellers, and end users who are designing and building mission critical applications. The transition from fixed TDM networks to wireless IP networks is a huge opportunity for Alliance. We have over 13 years of experience designing and support leading edge communication systems.

We have strategic relationships with all major suppliers in the VoIP industry and have installed more VoIP and communication servers than any company globally. Our products have been accepted in all areas of communications, and find their way into a wide range of mission critical environments:

Enterprise:
- Secure e-mail Gateways
- Network Security Appliances
- Interactive Voice Response
- Call Logging
- Automatic Call Distribution
- Telemarketing
- Speech Recognition
- Emergency Response Notification
- IP PBX
- Fax Services
- NAS and SAN Storage
- Microsoft and Linux Infrastructure

Service Providers:
- VoIP Session Controllers
- Network Management
- Test Fixtures and Equipment
- Policy Management
- Media Gateways
- Multimedia Conferencing
- Prepaid Wireless
- Unified Messaging Billing
- OSS Systems
- Auto Attendant
- Softswitch
- Voicemail
- Instant Messaging

GG: Now that it appears that growth and opportunity are the trends in the VoIP industry, what possible hurdles do you see that might upset this momentum?

JS: The biggest hurdle we see is the pace of new technology and applications and the increasing interoperability,

> We are seeing a huge potential in storage. We already have had requests for petabyte (1,000 terabyte) storage arrays.

The biggest hurdle we see is the pace of new technology and applications and the increasing interoperability.

deployment, and support of these solutions. The pace of innovation is so fast and the applications require so many technology suppliers that supporting increasing complex and very large scale systems takes tremendous technical and operational skills.

We have been working with the leading technology partners since the earliest days of the industry and have development the systems, processes, and methodologies to build and support these complex applications.

An example of the complexity we face is on any given day we are shipping dozens of highly complex session controllers, ATCA-based media servers, Open Source IP PBXs, and massive storage networks. It's taken us years to be able to design and support such a complex set of systems in an environment where five nines of reliability is considered the minimum.

GG: What are some of the technology areas where Alliance Systems is increasingly focusing, and why are these areas important to the future of your company?

JS: We are focused on several key markets: wireless, VoIP, enterprise communications, digital media distribution, and security to name a few.

Within these broad areas we concentrate on high availability server design, clustering, DC power and IP networking, VoIP, Video, and wireless infrastructure solutions.

We are seeing a huge potential in storage. The new service providers are coming to us looking for giant storage networks to aid in the delivery of IPTV, music, video, and entertainment content of all types. Within enterprises, regulations such as HIPPA and Sarbanes Oxley, as well as the demand for security, is creating demand for storage and logging of all types. We already have had requests for petabyte (1,000 terabyte) storage arrays.

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

JS: IP telephony is part of the worldwide transition to digital communications. As all devices become wireless and capable of full multimedia storage and content delivery, enterprises, governments, and service providers are all trying to integrate their separate networks. Part of the reasoning behind this is financial; IP traffic is a fixed rate cost, you pay for bandwidth versus the perminute per call of the voice networks. Open, standards-based systems are much less expensive and more flexible.

The future architecture will be composed of four basic building blocks.

1. Devices, every device imaginable will be IP enabled: cars, phones, TV, iPods, Blackberries.
2. Antennas, Antenna technology, whether its GSM, GPRS, WiFi, WiMax, will be capable of receiving ultrahigh bandwidth.
3. Routers, all the switches, PBX, cellular, TBD will be converted to routers.
4. Servers, all value added services will be run on servers, whether its storage, media, control. Open source, Web services, hosted services, and all VoIP networks are dependent on servers and software for intelligence.

Alliance is focused on providing the world's leading design and integration services for the worldwide deployment of servers and infrastructure for this transition. This is the biggest upgrade opportunity of all time and we hope to earn our fair share of this multi-trillion dollar transition. IT

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