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TMC Labs Innovation Award Winners
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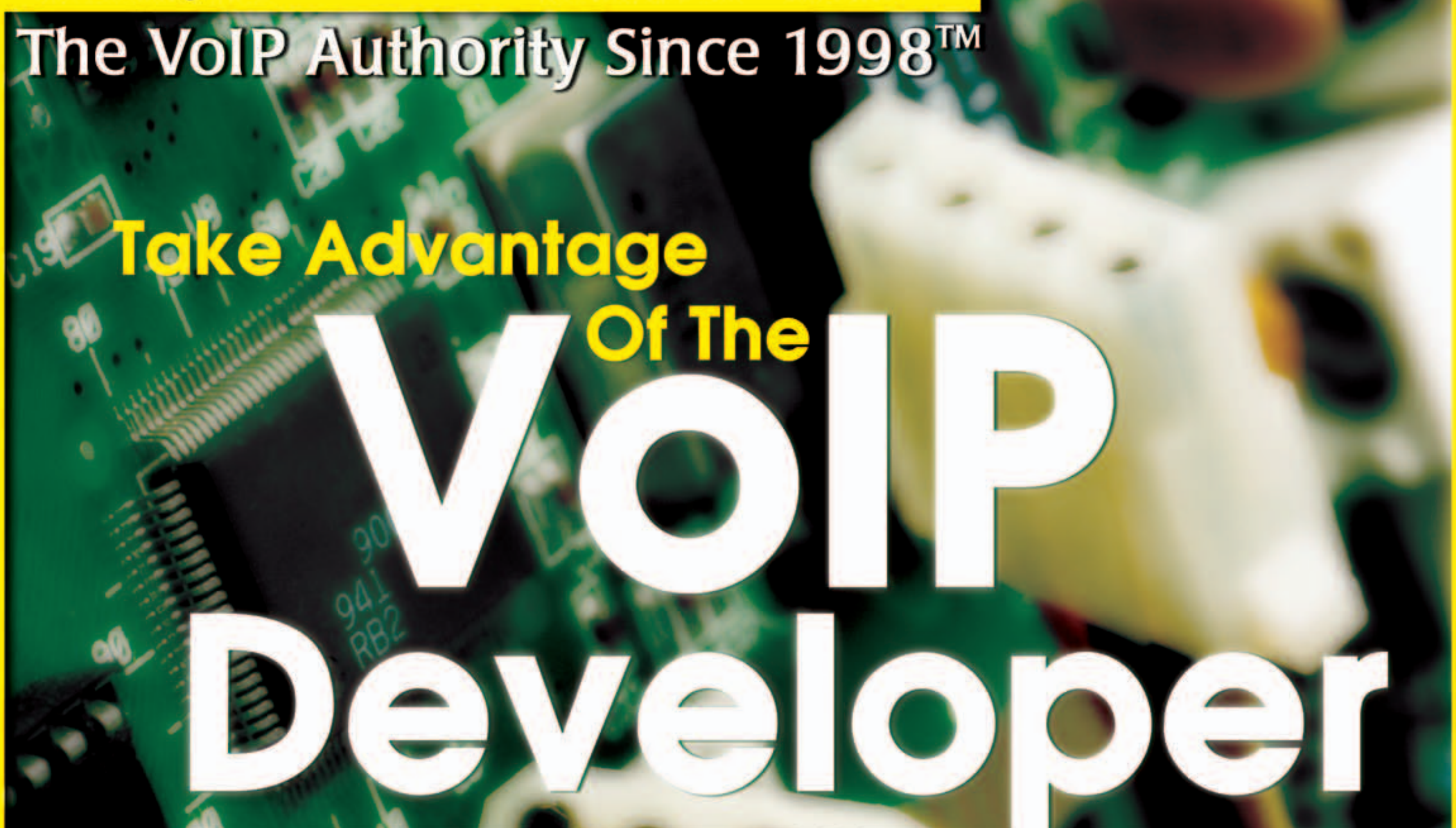

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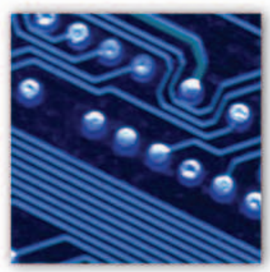

Also In This Issue:

- ClearOne's CEO on the Future of VoIP
- Telephony@Work CEO Eli Borodow on IP Contact Center
- Telx' Hunter Newby Cuts Through The Clutter

INTERNET TELEPHONY®

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

Group Associate Publisher,
Dan Goldemen
(dgoldemen@tmcnet.com)

EDITORIAL
Editorial Director, **Greg Galitzine**
(ggalitzine@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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EXECUTIVE OFFICERS
Nadji Tehrani, *Chairman and CEO*
Rich Tehrani, *President*
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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
Advertising Director - Eastern U.S.; Canada; Israel
Anthony Graffeo, *ext. 174,* (agraffeo@tmcnet.com)
Advertising Director - Western U.S.; International
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SUBSCRIPTIONS
Circulation Director, **Shirley Russo,** *ext. 157*
(srusso@tmcnet.com)
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EXHIBIT SALES
Sales Office Phone: 203-852-6800
VP of Conferences and Online Media
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The VoIP Authority

By Greg Galitzine

IMS Redux

Last month, in this space, we touched upon the significance of — and the tremendous opportunity presented by — IMS (IP Multimedia Subsystem). It was yet another chance for me to tout the mantra, "It's all about the services."

I just returned from Supercomm, and wouldn't you know it — one of the major themes of the show was IMS. It seemed that everyone was jumping aboard the IMS bandwagon (unless you were selling IPTV, that is).

Sonus Networks (news - alert), no stranger to cutting edge service provider offerings, announced their IMS solution at the show. The company made the point, however, that since their products were essentially born and bred to "do IP," the product line on offer from Sonus is IMS ready and has been since day one. Key components in the Sonus IMS solution include its existing products: the GSX family, PSX, SGX, ASX (define - news- alert) and Insight Management Systems.

In an announcement timed for the event, Sonus launched its new IMX Application Platform designed to further extend its service creation capabilities into the IMS environment. A quick look at the announcement tells us that "… the Sonus IMX Application Platform is a Web-based multimedia environment that enables wireline and wireless service providers to rapidly develop, integrate, launch, and manage enhanced telecommunication applications and services."

According to Hassan Ahmed, chairman and CEO of Sonus Networks, "IMS is the industry's first framework that allows wireline and wireless operators to leverage the disruptive force of broadband, breaking down the former barriers to service delivery and opening the door for the delivery of new services. The power of IMS is that it allows for a truly presence-enabled network and represents a profound change in the way service providers offer communication services."

As a major player in next-generation networking, it's imperative that Sonus stake a claim in the IMS space. Seems to me they're right on track.

In related news, Ericsson's IMS solution has been selected by TDC (the Danish service provider). The solution will allow TDC to offer its fixed network customers various services including IP telephony and IP Centrex to business users.

The IMS-based IP Centrex solution comprises a complete set of personal and group services, with support for such services as video telephony, conference calling, collaboration, presence management, instant messaging, e-mail integration, and support for remote workers.

Dan Sörensson, executive vice president of TDC Solutions, had this to say: "In the years to come, many of our customers will add IP and mobile telephony to their communication needs. Ericsson's IMS solution supports our strategy to offer new convergent and innovative services for our customers, no matter what access network they are using."

-Greg Galitzine, ggalitzine@tmcnet.com

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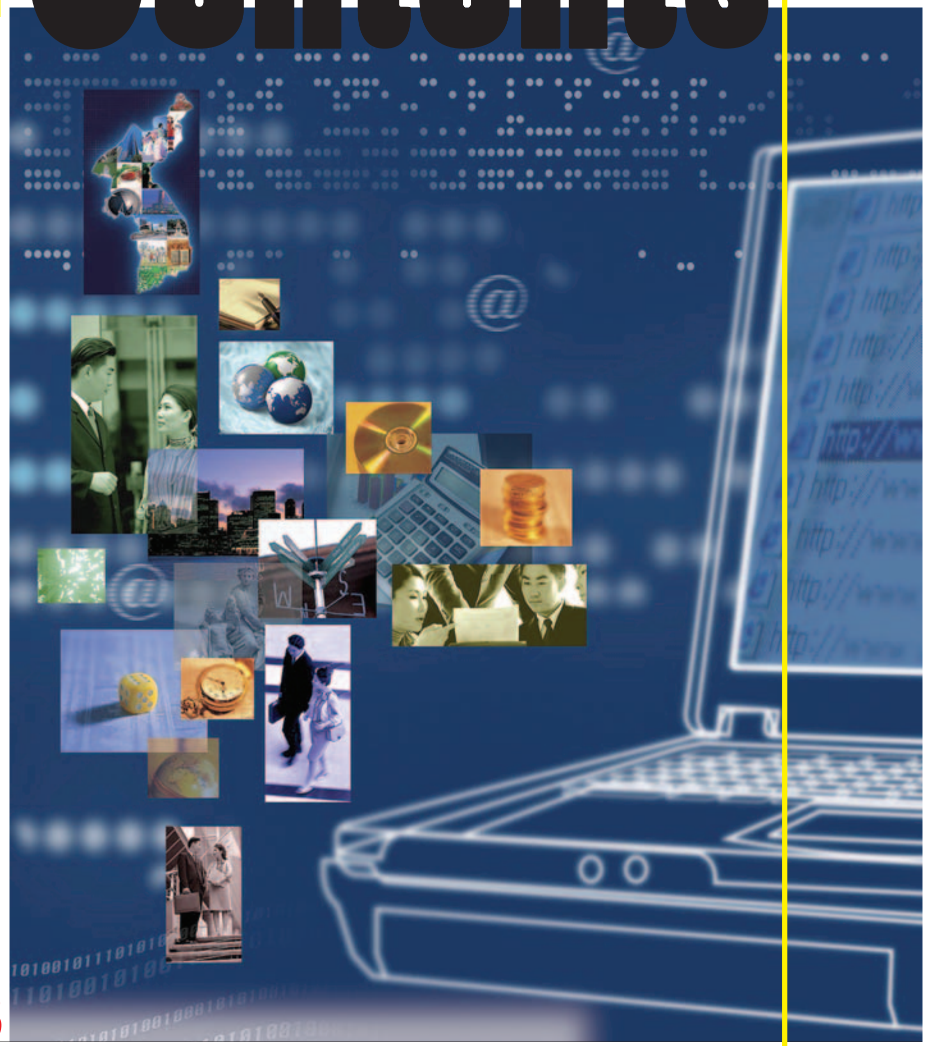

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Top 10 Visitors to TMCnet.com (Most Active, By City, Global)

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4. Middletown, New Jersey, United States
5. New York, New York, United States
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7. San Jose, California, United States
8. Mountain View, California, United States
9. Singapore, Singapore
10. Toronto, Ontario, Canada

QUOTE OF THE MONTH:

According to International Telecommunications Union data for 2004, the U.S. ranks sixteenth on the broadband penetration chart (with a rate of 11.4 subscribers per 100 population). Business Week criticizes the FCC for creating "a cozy duopoly of broadband providers: the Bells and the cable-TV companies," and recently reported that about 20 percent of Americans have no way to get broadband, and another five to 10 percent have only one choice, their local cable company.

The picture doesn't get any better when you look at speed and price.

– John Cimko (page 46)

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 over 3.9 million unique page visits per month, translating into nearly half a million unique visitors, TMCnet.com is where you need to be if you want to know what's happening in VoIP.

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

Rumor Control: Yahoo! Buys Dialpad. Ready To Offer VoIP?
Word on the street is that Yahoo! is buying DialPad…
http://tmcnet.com/128.1

Intrado Offers Comprehensive E911 Solution
Intrado's comprehensive E911 solution meets the requirements of the recent FCC mandate for VoIP service providers.
http://tmcnet.com/129.1

Services Expand World Of The Internet
Next time you're walking down a city sidewalk, look out for the Internet. It's all around you.
http://tmcnet.com/130.1

MCI Rolls Out Business VoIP In Europe
MCI Inc. announced this week that it begun its European roll-out of MCI Advantage, a VoIP-based service for the enterprise.
http://tmcnet.com/131.1

IPdrum Introduces Skype Cable For Mobile
IPdrum introduced the IPdrum Mobile Skype Cable, a bridge between Internet telephony and mobile phones.
http://tmcnet.com/132.1

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By Rich Tehrani

When Opportunity Knocks, Answer The Door

You've no doubt read about Just in Time Communications (JITC), the concept of squeezing the inefficiency out of communications. I've met with most every major telecom player about this idea and whether they refer to what they are doing as JITC or not, they all seem to be after the same goal: Making us more productive and efficient through the use of effective communications.

Inter-Tel

One of the companies on the fast track to boosting our productivity is Inter-Tel (news - alert), a communications company that has been around for a good long while. A public company, the Arizona-based organization isn't as big as its largest competitors so it has chosen instead to be quicker and more technology-centric than its competitors. In the nineties, it was one of the first companies to market with CTI solutions for the mid-market. The company also embraced VoIP (define - news - alert) before just about any other PBX player.

Now they are at it again, pushing out leading-edge applications ahead of the curve. To facilitate their latest announcement, the company purchased a leading Web collaboration company named Linktivity last year. Inter-Tel has kept the Linktivity brand alive and in fact released products from the subsidiary just this past December.

Most recently Inter-Tel announced collaboration products named Remote Support and Web Conferencing. Remote Support is Web-based and allows agents to remotely view and control a customer's desktop so they can effectively troubleshoot and resolve problems or issues with software, etc. This solution can also be used to install software remotely, configure software, and co-browse to allow customers to see links to important resources on the Web.

The product also allows a queuing capability that visually displays specific information on incoming callers, a recording and playback feature developed to enhance technical training and development, a system recovery tool to allow complex problems to be resolved quickly, and online management tools to track and access service histories and other relevant data. What is nice about the application in my opinion is the fact that no software needs to be loaded on a client PC to make this work. I frequently speak as part of webinars and in order to do so in many cases, I must download some sort of software to my desktop. In some cases I need presenter software and also regular "attendee" software. I have come across many enterprises who don't allow such software to be installed, meaning collaboration software requiring installs can't be used.

The Web Conferencing product would obviously be similar to the above product but used generally for different applications and as such, the program's ability to allow for document sharing, file transfers, and other collaborative activities makes sense. Users can take advantage of one-click Web and multipoint video conferencing. The program also integrates with Microsoft Outlook e-mail and calendaring and supports 128-bit encryption as well.

Other points worth noting... These products are smart enough to step down in functionality and interface complexity to deal with over-zealous firewalls. There is an HTML interface that is employed if Java doesn't work for some reason (yes you have to have Java downloaded for this to work). There is also a Pocket PC client for those of you with iPaqs and other PDAs. Speech technology can be used to change preferences if needed. For current Inter-Tel customers, the company has done its best to be backwards compatible and if your PBX is four years old or newer you can use these applications. I think this is pretty impressive.

I am further impressed by the licensing terms allowing for concurrent users as opposed to per-seat. Packages start at 10 concurrent users, 15, 20, and then up to 100.

Two trends are driving JITC, the first is the realization by all hardware vendors that they are in a dead end business. China will soon eat their lunch. They can choose to do what IBM did and partner with a Chinese company to manufacture products, but still, margins are being squeezed out of hardware so even if you go the China route which many are already doing, you aren't going to make lots of money selling PBXs. Right now there is a good amount of money being made selling phones but that won't last forever as soft phone adoption grows, SIP becomes more ubiquitous, and commoditization takes hold.

That leaves software and after you make a soft client you need to figure out what other applications you can sell. Productivity tools are a logical step which is why we are seeing so many PBX companies preaching the productivity mantra. Problem is, they are still all calling it something different.

What does all this mean for you, the purchaser of these systems? Everything. Productivity boosts beyond your wildest dreams. Think I am dreaming? Imagine your users able to instantaneously launch audio and Web and even video conferencing, as easily as you IM. But in this case, the telephony is tied in. Imagine being able to show others a question you have with a document. You can get an instant response on your queries. You can work together to improve the document, get instant sign-off, and record all versions of the document while recording the conversations and corrections regarding the document in question.

Productivity will shoot up when these applications become commonplace, the physical location of workers will not matter. Instead of caring where people are you will need to worry about your worker's proximity to broadband. We are talking about the connected, seamless, all encompassing corporation that moves at the speed of light and takes no prisoners. If you think technology has made business fast-paced because of e-mail and IM, you aint seen nothing yet. The game will get more fun, you will become more productive and we all win. Will you gain your nights and weekends back to spend with your family instead of

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checking volumes of e-mail? Sorry — probably not. But on the bright side, if you make your company this much more productive you can likely ask for a nice fat raise. Happy negotiating.

DSP Board Happenings

I recently blogged about how Howard Bubb left Intel/Dialogic after an impressive career helping to build an industry we once referred to as CTI or computer telephony. During Howard's tenure, Dialogic grew like crazy, they developed an ecosystem of profitable partners and made lots of money for a lot of people to share. Bubb's company was late to the VoIP gateway market but soon thereafter became the market share leader. Natural Microsystems gets credit for really popularizing the VoIP gateway in the late nineties.

Another departure from the industry was Mike Ross who everyone knows as one of the nicest guys in telecom and with a legacy that others can only envy. He was an essential part of Dialogic during the eighties, founded Rhetorex (now part of Brooktrout) and headed up Aculab in Florida for over five years.

Apparently Bubb is at a start up and I haven't had a chance to track down what he is up to. Mike Ross is going to do something grand in my opinion. We recently spoke and I am impressed with some of the ideas he has about starting a new business. I can't share anything specific with you as of yet.

Taking Mike's place is Mike Donoghue, a veteran of Brooktrout that I had a chance to sit down with recently. Donoghue seems more than up to the task of leading this company into the future and as his first move he will be relocating the company's headquarters to Massachusetts from Florida. Not only is this where he lives but it gives the company access to loads of talent from Brooktrout, NMS, and AudioCodes. They are hiring so send a resume if you are interested (not to me please, to Mike).

I also recently had a chance to speak with Allan Pound, the founder of Aculab and he is more energized now than at any time I have seen him in the past. Actually to be specific, I saw him this engaged when he launched Prosody, a very dense DSP resource board in the late nineties. Now he is back with what seems like an amazing evolution of this same product: Prosody X. This new board is also very powerful, allowing up to 600 channels of media processing. The core of the card is IP — actually an Ethernet switch. This is in addition to a TDM (define - news - alert) matrix used by past Prosody cards. There are a number of TDM options and strong SS7 support. We can further expect PCI Express and cPCI variants of this product in the future and ATCA is on the table with no definitive decision to move forward as of yet. The price/performance of this solution is said to be 30–40 percent lower than competing alternatives.

If the "X" product is overkill for you, you may want to consider Prosody "S" as this product is HMP-based and fits better in an SMB environment with the ability to scale.

Speaking At SuperComm With The IPCC

At SuperComm a few weeks back, I was invited to speak on a panel moderated by the IPCC's Michael Khalilian. The IPCC is the leading VoIP association of service providers and equipment manufacturers and Mike did a stellar job moderating a standing-room only crowd eager to learn about VoIP and what the future holds in this space. The panelists (including Time Warner Telecom's Erol Turner and SIPquest's Vish Raju) were first rate and I learned lots while I wasn't speaking. I even took notes.

Khalilian started off by saying that the FCC shouldn't just make policy but they need to also provide tools allowing service providers to do what is requested of them. Of course he was referring to 911 and the ability for VoIP providers to provide e911 at reasonable rates.

He went on to ask what happens when you try to make a 911 call on a cell phone that goes out of service? Is Nokia liable for such issues? He has a valid point. Today, if a 911 service provider tells you that you need to sign up for 911 and the customer fails to do so, it is the service provider's fault. The cellular network after over a decade of development is off the hook if they drop a 911 call or if they don't have coverage in areas they claim to have coverage. Seems like a double-standard to me. Part of this issue gets back to the definition of a service provider as true service providers are protected to some degree from such lawsuits. VoIP providers are not generally classified as service providers and subsequently need to deal with certain liability issues.

Mike went on to discuss that service providers today must give all the class 4 and 5 features that customers are used to. You can't skimp on features and finally, the future of VoIP is WiFi telephony and bundled services. Who can argue with that?

Erol Turner of Time Warner Telecom, a telecom industry veteran, discussed his fascination with SIP and the growth rates it is experiencing. Turner said that in five years, even his refrigerator will be controlled by SIP. Is he serious? I am not sure but he said he would put money on it. This means he will either have an unlisted non-SIP number in five years if he is wrong, or he'll be rich.

Speaking of riches, the most common question I seem to get from people recently is how they can make money in VoIP. Turner told the audience that if he had to leave his job and do something else, he would become a systems integrator. I guess this means that Time Warner telecom job pays pretty well for now.

Vish Raju from SIPquest (news - alert) explained that we need to improve the user experience and we need to enable virtual work-teams. His presentation echoed mine on Just in Time Communications in many ways. IMS excited Vish a great deal and he sees this as the future of telecom. "Communications will eventually become less 'siloed' and SIP and IMS will help make this happen." He said. The promise of IMS is how the panel finished off and he described IMS as "Bell head" a "Net Head" and a "VoIP Head" coming together.

What's New?

I met a number of companies at the show and one of the first on my list was BEA Systems. My talk with the company was exciting as it reminded me a great deal of what I heard from Lucent in 1999. Back then the premise was that the really exciting applications were going to be developed by 17–25 year-olds. So I smiled when I heard company representatives repeating this now half-decade old prediction. I believe it to be true as well and just like shopping online the prediction was a bit ahead of its time.

BEA Systems

BEA (news - alert) is well connected with service providers, already providing valuable infrastructure to this community. They are in a great position to continue supplying service providers with SIP solutions as the world transitions to VoIP, SIP (define - news - alert), and IMS. By enabling applications to be developed using Java and SIP Servlets and allowing devel-

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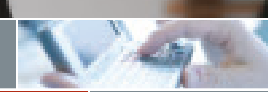

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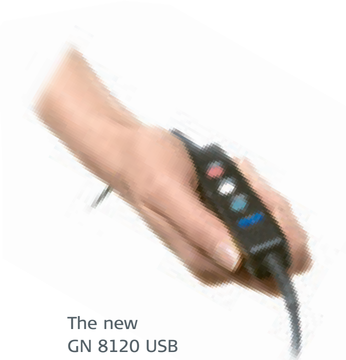

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The company's various products from SIP Server to Communications Platform work together to allow wireline and wireless providers to among other things, take advantage of a new source of developers on their platforms.

Broadsoft

Broadsoft (news - alert) announced some new customer wins at the show but what is more interesting to me is their offerings in the hosted PBX space — applied to mobile carriers. Their Mobile PBX product allows service providers to allow employees to park and pick up calls with full Web control. You also get unified messaging and many of the features of VoIP without the need for VoIP. Of course VoIP can be offered as well in the fixed line world and as wireless and wireline converge, service providers can now offer the best of both worlds and tie it all together for their enterprise customers.

These services can also be supplied by an MVNO if you like and the company claims their solution is the first IMS compliant application server to be shipping. What I am seeing in the market between Broadsoft and Sylantro besides intense competition is growth. This segment is finally doing well after years of struggling in a tough market.

Ditech/Jasomi

In what may be the most strategic acquisition in VoIP after the acquisition of NBX by 3Com and Selsius by Cisco back in "the day" (prior to 2001) is the acquisition of Jasomi (news - alert) by Ditech Communications (news - alert). When I recently met with Ditech executives prior to this announcement I asked a whole bunch of questions about how their transcoding solutions would work with a session border controller (SBC). It seemed that the intersection of these two markets made a great deal of sense and Ditech's experience in codecs and wireless coupled with a session border controller would make a nice offering.

A few months later, the company announced they acquired Jasomi and are now offering what they call the best border services offering consisting of transcoding and encryption among other services. The combined solution goes beyond security and NAT traversal to QoS security, giving what the company calls an umbrella of control. As threats such a voice phishing become prevalent, security becomes an important issue and Ditech Communications is doing a great job getting out ahead of the curve. The future for Ditech? They see themselves evolving into the media layer of IMS.

One last comment on this transaction is that one of the influential factors in the transaction was Jasomi's decision to design their next generation SBC on ATCA architecture, the new telco grade form factor being touted by Intel among others. Obviously using the same form factor and bus means that cards can be mixed and matched in a chassis to get the desired level of functionality needed depending on the application.

Jasomi is the second SBC vendor to be picked up recently. A few others I have spoken with are in talks to sell. Cisco in fact was rumored to have just passed on an SBC deal a short while back with the word on the street saying that they will acquire someone soon. Interestingly not too long ago, those intimately involved in the SIP space declared SBCs superfluous, now, based on all the M&A activity, they seem to have become the glue that holds VoIP together.

Versatel Networks

You can get bleary-eyed learning about the various technologies in the VoIP space for service providers so whenever I speak with Versatel Networks (news - alert) I look forward to the meeting as they look at VoIP the way I do and that is with "revenue generating services" colored glasses. Their mission is to personalize VoIP and if you are looking to squeeze revenue out of VoIP, you better personalize it by developing device agnostic applications. Similar to BEA Systems, the mantra here is let developers who are not necessarily telecom experts have access to building tomorrow's killer application.

Some of the applications they allow are a family or workgroup oriented push to talk service. Today there are push to talk services marketed heavily and family plans promoted heavily. Where the two intersect is a sweet spot in the company's opinion. And mine.

I saw some intriguing demos that kids will drool over, like dragging a buddy list into a voice portal to allow instantaneous conversations. I saw an application that allowed a virtual number to be allocated to a conference bridge coupled with an autodialer allowing you to instantaneously connect to a number of callers. In other words, I could have a number, or an IVR menu option assigned to my editorial team for example. When triggered, all parties will be called immediately. You can also set up ring-back tones that are customized for a specific caller allowing you to play a customized message to you spouse for example. This message would be played instead of the typical ringing tone we are familiar with. I could go on but you get the idea. VoIP revenue generation is about services and these services need to be created easily. Companies like Versatel Networks are helping to bring these services to market.

Metro Ethernet Forum

As we walk down the path of new services let's keep in mind that many of them will take up massive quantities of bandwidth. As such we need to explore ways to get this bandwidth to customers effectively. One way is via Ethernet and if you believe a legion of analysts and the vendors who generally pay the analyst's salaries, metro Ethernet is where things are headed. In this case I believe the analysts are onto something and metro Ethernet looks like it will become the dominant broadband standard for the enterprise at some point.

I stopped in on the Metro Ethernet Forum's booth for what was one of the more professional presentations at the show, allowing me to witness IPTV, VoIP, guaranteed rate Internet access and more. The forum has recently contracted with a test lab to allow for certifications so that products will be guaranteed to work together. A full sixty companies that make hardware are part of the forum making it quite a force in the market. Associations like the MEF are crucial to enabling the next generation of bandwidth hungry applications and I salute the association and their partners for helping to get the industry moving towards interoperability across vendor's products.

VocalTec

VocalTec (news - alert) has made strides in improving its Essentra softswitch platform. They tell me they pretty much have full Centrex functionality and that they compete well with other players in the market. In a continuing sign of converging markets, there is an integrated application server as well, meaning no third party is needed. Furthermore, provisioning is done

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at once instead of separately. A continuing trend in VoIP and information technology in general is the move to open-source and following this movement, VocalTec (news - alert) has embraced Linux. Similar to solutions from Brooktrout's (news - alert) SnowShore division the whole package is sold in an IBM Bladecenter T — the ruggedized telecom based server that IBM seems to be having good success selling into the OEM market. In a single cage I am told you can support up to 250,000 redundant subscribers.

Persona Software

Longboard has renamed their company Persona Software (news - alert) and is focusing more squarely on the fixed mobile convergence market or FMC. They play in a space that sees companies such as Bridgeport Networks and Kineto Wireless and they focus on providing a richer set of applications than their competition. The Persona Platform can be an integral part of IMS allowing you to take advantage of many of the 3GPP features today without a forklift upgrade. A case in point is a great demo I saw of a dual mode phone working in the WiFi world transitioning in a few milliseconds to a GSM network and back. I asked how the hotspot would recognize the device and they told me that the service provider would have an agreement with the hotspot vendor and auto logon will take place via an HTTP server. The WEP key will be stored in a database allowing devices to get on the network and be billed accordingly. Dual mode phones will be all the rage in 2007-2008 in my opinion. I am looking forward to getting one for myself.

Bridgewater Systems

Are you one of those people who hogs bandwidth? You know who you are — you are one of the 5 percent of the customers that hogs up to 60 percent of network resources. You heard that right. I suppose many of these people are listening to the latest album from 50 Cent (sure, I am hip) they just downloaded and don't have time to read my column. A company called Bridgewater Systems (news - alert) makes products that help service providers by allowing for tiered service levels, policy based bandwidth management, metered services, and a whole host of applications from variable QoS services to triple-play to bandwidth on demand. Service providers need ways to generate more average revenue per user (ARPU) and using these techniques, you can achieve such goals.

The company recently decided work on interoperability with companies that manage deep packet inspection allowing for much more granular intelligence about subscriber usage patterns. This technology can be used to direct subscribers to a portal to purchase more bandwidth if they bump up against the maximum allowed threshold for example. Virus detection is another potential use for this feature.

UTStarcom

By the time you read this article a flood of WiFi telephony handsets will be on the way to the consumer market from companies like Vonage (news - alert) and Vox Communications (news - alert). UTStarcom (news - alert) is one of the companies behind the push towards economical WiFi telephony handsets. The company is as much of a force in the service provider space as they are going to become in consumer electronics with this new phone. They recently partnered with Cisco to provide a TV over IP (not to be confused with IPTV) solution to the Brazilian market. The company has also announced a third-generation IP DSLAM named the AN-2000 B1000. The DSL access module will allow bandwidth hungry applications such as IPTV to be rolled out easily. There is also an interesting product called the UBS Fiber Node, an environmentally hardened DSLAM that can be frozen, thrown in water, and launched in the nose of an ICBM (I made the last part up) and still work. We will be seeing more solutions from this company in my opinion as they are gaining tremendous momentum in the market.

Intel/ATCA

ATCA (news - alert) seems to be to have become the defacto Intel-based architecture for building service provider telecom applications. I am seeing more and more companies migrate from proprietary hardware to ATCA systems in an effort to reduce the need to deal with proprietary hardware design and furthermore the ability to take advantage of regular and predictable performance increases.

Intel (quote - news - alert)has noticed this trend and works with manufacturers to help them get their products on ACTA as soon as possible. The more equipment vendors that make the switch, the more processors and associated technology Intel sells. Great business model, isn't it? In fact if the recent move by Apple to Intel doesn't tell you about the future of competing in the hardware business with Intel, I am not sure what will... The point is that if you have a proprietary SBC and your competitor is betting on ATCA, you will likely be behind in cost/performance.

Intel is very focused on the emerging areas of telecom and IMS is an important region of growth for the company. They have had 95 ATCA design wins as of late and they are not taking this momentum for granted. They have published interoperability design guides to help different products work with one another like a single and happy TCA family. When I asked about some of the competing enterprise server platforms that are in the telecom space, they told me politely that an enterprise server is just not designed for telecom. They have a point and it isn't like they don't make money if you put your operations on an enterprise server (unless of course you're using AMD chips).

Simply stated, Intel sees a tremendous opportunity for vendors to develop on ATCA as they can increase the time to marker and in turn time to revenue.

Spirent

Spirent (news - alert) may have had more releases at SuperComm than any other company I visited with. One of the areas of focus was triple-play where Spirent will assure QoS to keep customers happy. They have developed 20 critical test scenarios and test each network element, end-to-end across the network and then continue to monitor QoS after deployment. In addition, the Abacus 53 IP telephony and Video Rollout System validates voice and video media quality in pre-production triple-play networks. Consider the 53 a condensed Abacus 5000 that is ideally suited for distributed deployment. Obviously testing new services and video QoS are crucial to get right before customer roll-out and that is why Spirent has chosen to launch a broad array of testing products to help service providers solve network problems before deployment and in addition be notified as soon as there is a problem with voice or video performance.

VoIP Developer

The inertia in the service provider market seems greater to me than in the enterprise space at the moment. Many developers attending TMC's VoIP Developer Conference (http://www.voipdeveloper.com) next month have told me they think the best ROI is in the provider space. There is just s much spending that is going to be done by service providers as they embrace VoIP. From IMS to new ATCA systems, I see this segment of the market growing fastest for the foreseeable future. I also believe we are at an inflection point in the enterprise space. A boatload of new resellers are learning to sell VoIP and they will soon give the SMB market a tremendous shot in the arm. IT

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Quintum Intros BX Series Tenor VoIP MultiPath Switch

By Johanne Torres

VoIP (define - news- alert) technology provider Quintum Technologies (news - alert) announced the introduction of the Tenor BX Series MultiPath Switch. The company described the Tenor BX as a complete VoIP switching system for enterprises with ISDN Basic Rate Interface lines. Quintum explains that with the Tenor BX in place, enterprises will now benefit from a variety of valuable applications such as PBX extension, remote office connectivity, long distance consolidation and call centers. The BX is built on Quintum's Tenor VoIP MultiPath switching platform.

The Tenor BX can support two, four, or eight BRI S/T lines, and a good VoIP CPE access option for service providers that are deploying services to customers that have PBXs that support ISDN BRI trunks.

Eatontown, NJ-based Quintum said the series increases the breadth of the Tenor product line allowing support of ISDN BRI installations. The company is sure that the addition of BRI provides the opportunity for both enterprises and service providers to deploy the Tenor VoIP access solution to a variety of offices that support analog, T1, E1, PRI and BRI interfaces.

"We have had a lot of demand for the ability to extend Tenor based network to locations that support BRI," said Chuck Rutledge, vice president of Quintum Technologies. "With the addition of the Tenor BX product, Quintum further expands it ability to support a wide variety of PSTN connections, and it has all the advantages of the Tenor product line, most importantly, the ease of installation, and transparent deployment."

"It is difficult to find any VoIP solution that supports ISDN BRI and the Tenor BX was the perfect solution for our customers needing VoIP deployment", said Josef Bressner, president of Communiports AG. "We are able to support VoIP off of an existing ISDN BRI based PBX without any modification to the existing network, it deployed transparently."

The Tenor BX is currently available starting at $1,600.

http://www.quintum.com

3Com Adds To IP Phone Software

By David Sims

3Com Corporation (quote - news- alert) has announced a comprehensive enhancement to its Internet Protocol telephony software designed "to help small and medium-sized enterprises reduce costs, increase employee productivity and strengthen customer interactions," according to company officials.

Key features in this new, fifth generation release of the 3Com NBX IP Telephony system, the industry's first IP PBX (private branch exchange), include:

- New automatic call distribution option to allow businesses to easily expand their customer care service for up to 199 agents as part of their own in-house operations.
- Support for silent monitor/whisper/barge-in. When used with the ACD feature, supervisors can listen in on an incoming call, silently coach agents while on an active call and interrupt a call in progress if needed by barging into a call.
- Account code dialing and verification to allow professional service companies to more easily track and bill professional staff time.
- Support for the 3Com 3103 Manager large screen, Gigabit-speed phone with personal directory and call logs and support for the 3Com 3100 Entry, a single-line Power over Ethernet compliant IP phone designed to be a cost-effective alternative to traditional analog telephones.
- Support for two cordless IP phones for retail and small office applications.

With this release, 3Com offers a total of eight IP phones to complement the 3Com NBX system, including a competitively-priced Gigabit phone, the new 3Com 3103 Manager IP phone, which increases user productivity by managing up to 12 simultaneous calls.

The 3Com 3103 Manager Phone offers a long-term investment as the acceptance of "Gigabit to the desktop" becomes more widely accepted by enterprises. Pass-thru ports on 3Com IP phones also allow users to conserve LAN switching ports. For example, users can connect a personal computer to the 3Com IP phone, which is connected to the Ethernet port in the wall.

http://www.3com.com

IP PBXs Up, Overall PBX Market Down

Worldwide PBX/KTS revenue dropped 12 percent to $1.5 billion between 4Q04 and 1Q05, but is two percent higher than a year ago. So says an Infonetics (news - alert) Research report. The number of PBX/KTS lines shipped worldwide decreased to 6.3 million in 1Q05, 15 percent lower than in 4Q04.

According to forecasts, the market will bounce back to $1.7 billion by 1Q06 as the market continues to move towards IP voice technology and benefit from an improving economy.

Pure IP is the only PBX category to beat the budget blues in 1Q05, increasing 10 percent to $223 million, 36 percent higher than a year ago. Revenue is forecast to jump another 24 percent to $277 million in 1Q06. Pure IP and hybrid system revenue together total $1 billion.

TDM system revenue sank 20 percent and hybrid system revenue dropped 13 percent in 1Q05. Most vendors recognize the shift in the market towards IP and have moved their products accordingly. The TDM segment, which still accounted for half the market at the end of 2003, is now less than a third.

Highlights of the report include the following:

• The top IP PBX system vendors for lines shipped are Alcatel, Nortel, Avaya, Cisco, and Mitel.

• 3Com had the highest sequential line shipment growth, up 26 percent in 1Q05.

• Hybrid PBXs account for 57percent of PBX/KTS revenue, TDM 28 percent, and pure IP 15 percent; by 2008, hybrids will increase their dominance to account for 67 percent of the market, pure IP will increase to 23percent, and TDM will decrease to nine percent.

http://www.infonetics.com

SMC Expands Enterprise Wireless Solutions

SMC Networks (news - alert) recently announced the newest additions to its arsenal of solutions for extending enterprise connectivity with the flexibility of secure, robust wireless. Designed to expand the range, placement options, and throughput of the Enterprise wireless network for cross-campus and multi-story applications, SMC's newest antenna and amplifier solutions help networks go the distance indoors and out, point-to-point and point-to-multipoint.

Compatibility being paramount for easy set-up and expansion and reliable connection SMC's wireless solutions for the Enterprise comply with the 802.11g standard to provide speedy, long-range connections for users, whether via 802.11g or 802.11b adapters, or for end-to-end applications. Among the newest products for easy network expansion, the new EliteConnect 2.4GHz 500mW Power over Ethernet Amplifier (SMCAMP-500G) increases the signal strength and operating distance of any in SMC's suite of 802.11b/g Enterprise Access Points, Bridges and High-Gain Antennas. And, Power over Ethernet support via SMCAMP-INJ means that placement restrictions due to lack of electrical power are lifted, as it takes its power from the CAT5 network cable.

http://www.smc.com

AVST Launches CallXpress 7.7

Applied Voice & Speech Technologies, Inc. (AVST) (news - alert), has announced a new version of its flagship unified communications (UC) solution, CallXpress. Boasting new features and applications, CallXpress 7.7 is the ideal solution for companies looking for alternative options to outdated voice mail systems. The new version is enhanced with easy-to-use telephone user interface (TUI) options, increased unified messaging flexibility, and simplified administrative capabilities.

"AVST continues to drive the development of applications, features and enhancements with the mobile workforce in mind. Delivering on our commitment to enable communications anywhere/anytime, our solutions provide remote access to unified communications and collaborative applications through laptops, cell phones, PDAs and BlackBerrys," said AVST's Vice President of Product Management, Tom Minifie.

"IP technology is broadening the definition of the mobile workforce," said Jay Lassman, research director, Gartner. "Innovative mobility enhancements to system architecture and user interface are necessary to simplify anytime, anywhere access to business communications infrastructure."

http://www.avstgroup.com


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- Professional 24X7 Support
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LINKSYS
GRANDSTREAM
DIGIUM
CLIPCOMM
CISCO
CARRIER ACCESS CORP.
AUDIO CODES
voipsupply.com

Newport Unveils Carrier-Class SBC

By Johanne Torres

VoIP (define - news- alert) Session Border Controller developer Newport Networks announced release 2.0 of its 1460 session border controller software with enhanced redundancy and regulatory compliance in the U.S. market. Newport described the product as the industry's only carrier-grade session border control system. The company also announced it expanded the organization with new offices in Frisco, Texas. "Service providers now require the same predictability, reliability and profitability as the PSTN for voice and multimedia over IP," says Terry Matthews, Chairman of Newport Networks. "It's time for vendors to step up and help providers succeed with new packet-based services. The US market is absolutely ready for a session border controller of this caliber. Newport is here to raise the bar."

The company explained that where competing solutions address the challenges of completing calls behind corporate firewalls and Network Address Translation (NAT) devices, the 1460 adds architectural and cost-performance advantages. According to Newport, these bring the predictable quality and reliability expected of the PSTN (define - news- alert) to VoIP for the first time. The company says that as IP-based telephony and multimedia gain traction, the 1460 satisfies the stringent requirements of Tier 1 and Tier 2 service providers, re-defining "carrier-class" session border control.

The 1460 architecture offers the following features:

- Redundant Management Cards, ensuring manageability under card and network failure scenarios;
- Redundant power feeds supplying segregated power zones, ensuring that any power failure does not impact the ability to offer service;
- Automatic fail-over among cards;
- Link Aggregation (802.3-2002), allowing traffic to continue to flow even in the event of line failure.
- Support for up to over 120,000 concurrent calls
- Up to 1 million registered subscribers per node;
- Architectural flexibility allowing the product to be configured optimally to accommodate processor intensive applications such as SIP re-registration filtering;
- Architected for bandwidth-intensive multimedia applications such as video.
- Each call is policed to ensure that the bandwidth used matches the authorized bandwidth, thus preventing service theft and ensuring call-quality for other users,
- Session admission control: Allows service providers to manage QoS for each customer and therefore ensure that Service Level Agreements are met.

Newport explained that the chassis-based 1460 was designed for an extended life cycle in Tier 1 and Tier 2 carrier networks. The system can easily address new applications through software upgrades and the addition of new interface and processing cards. The 1460 allows advanced features, such as CALEA, to be implemented.

Release 2.0 of Newport Networks' 1460 session border controller will ship in the North American and international markets through third quarter 2005.

http://www.newport-networks.com

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VoIP Inc. Buys Caerus For $30 Million

By Johanne Torres

Its official: VoIP, Inc., (news - alert) and Orlando, FL-based Caerus, Inc., (news - alert) have merged. The companies announced that VoIP, Inc. has acquired Caerus, immediately initiating the process of merging operations. As part of the agreement, VoIP, Inc. acquired 100 percent of the business of Caerus, Inc., and its wholly owned subsidiaries Volo Communications, Inc., Caerus Networks, Inc., and Caerus Billing, Inc., in exchange for 16.9 million common shares of VoIP Inc. stock, a transaction valued at $30 million.

The deal will allow VoIP, Inc., to continue operations of Caerus, Inc., and its subsidiaries under their existing names, consolidating all operations under VoIP, Inc. After the merger, Steven Ivester will continue as CEO of VoIP, Inc. as well as all of the subsidiaries, and Shawn Lewis, Caerus' founder and CEO will become CTO of VoIP, Inc.

"The synergies of VoIP, Inc. and Caerus are unmistakable and cannot be found anywhere else in the industry. We are creating a combined company with one of the broadest product offerings, deepest distribution channels, largest sales and service support teams, which just doubled in size in the industry not to mention one of the largest proprietary VoIP networks in the U.S. It's a combination that will benefit all of VoIP, Inc. and Caerus, Inc. customers and suppliers," VoIP, Inc.'s CEO Steven Ivester said.

"Combining Volo's current customer traction, technology, and network with VoIP, Inc.'s existing portfolio of companies is a huge step for the industry. Now there is a true, complete end-to-end solution for this ever-expanding VoIP industry. Nowhere else can you find a turnkey solution from end-point devices to fulfillment, and support services of such magnitude," said Shawn Lewis.

"We're especially pleased that Shawn Lewis will continue to lead technology innovation and development efforts for VoIP, Inc., as CTO of the company," added Ivester. "He is a well-respected leader and innovator, having written the patents for the first softswitch and SS7 Media Gateway. One of his many accomplishments was the sale in 1998 of XCOM Technologies, Inc., a CLEC that he co-founded, to Level 3 Communications, Inc. for common stock, options and warrants valued at $154 million."

Volo Communications, Inc. (news - alert) is the licensed facilities-based CLEC (Competitive Local Exchange Carrier) and IXC (Inter Exchange Carrier). Caerus Networks, Inc. is the technology research and development subsidiary, and Caerus Billing, Inc. is the billing and mediation subsidiary. Caerus, Inc. and its three subsidiaries claim to have generated revenues during calendar year 2004 that totaled $14 million, and based on current contracts and purchased orders, revenues are estimated to exceed $38 million for calendar year 2005.

http://www.voipinc.com
http://www.volocommunications.com

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VoX Intros Total VoIP System

By Johanne Torres

Packet communications (news - alert) services provider VoX Communications announced plans to introduce a 'total VoIP system.' Using the new system, the company, through a nationwide network, will be terminating traffic with Global Crossing VoIP Outbound Services and using the wireless IP phones and VoIP customer access devices from UTStarcom, Inc.

The company's total VoIP system includes VoX's wholly owned packet telephony technology and its nationwide network with Tier One interconnection partners like Global Crossing, who operates a global IP-based network and SIP-based VoIP platform.

VoX explained that it is also deploying UTStarcom's wide range of VoIP customer premises equipment (CPE), including UTStarcom's iAN-02EX VoIP analog terminal adapter (ATA) for residential customers and UTStarcom's iAN-08E series VoIP gateway for SOHO/enterprise customers. The company recently added new CPE, UTStarcom's F1000 portable WiFi handset.

VoX Communications offers wholesale broadband voice, origination, and termination services for cable, wireless, and wireline operators and VoIP service to the small business and residential marketplace. The company's services feature Call Hold, Call Waiting, Caller ID, Call Transfer, Hunt Groups, Do Not Disturb, Call Forward, International Call Blocking, Call Return, Repeat Dialing/Redial, Extension Dialing, Anonymous Call Rejection and e-mail notification of voicemail, all at no additional charge. Clients are also able to choose additional features include: Multibox Voicemail, Music on Hold, Corporate Conference calling, Reassign Phone, Find me/Follow me, Auto Attendant, among others.

http://www.voxcorp.net

Arbinet Introduces Paid Voice Peering

Arbinet-thexchange, Inc., (news - alert) has announced a new service designed to allow voice over Internet protocol (VoIP) companies to get paid for telephone calls to their customers.

Local access charges, the fees for terminating a call to a customer, which total an estimated $14 billion annually in the United States alone, have been unavailable to a VoIP service provider until now because calls destined for a VoIP customer are typically sent through a local telephone company's network and that company retains the termination charge. Two key puzzles needed to be solved to enable commercial settlement for inbound Internet phone calls. The first was a way to match a traditional phone number to the IP address of a VoIP customer, and the second was the ability to charge for routing the call directly to the customer over the Internet.

"There are companies today who provide free termination of calls between VoIP service providers, but, in our view, that will not be of interest to large telecom companies," explains Curt Hockemeier, President and CEO of Arbinet. "Our new service commercializes this activity in such a way that we expect it will eliminate an expense and create a new revenue stream for VoIP service providers."

Arbinet's new commercial voice peering service provides a central telephone number to IP address mapping database and settlement system. An inbound call from an Arbinet buying Member will be checked against this database and sent directly to the selling Member's VoIP equipment. The seller can charge a termination fee for receiving the call. Arbinet will collect and settle the transactions with buyers and sellers every 15 days.

Arbinet operates a voice exchange service, which processes approximately 1 billion minutes of traffic monthly for 375 telecommunications service providers worldwide. Arbinet's new paid voice peering service will allow telephone calls to find their way through the Internet to the called party, bypassing the local, incumbent telephone companies.

http://www.arbinet.com

Keynote Launches VoIP Quality Benchmark Survey

Keynote Systems (news - alert) has launched the first study to benchmark the service quality of the leading providers of VoIP phone services, as perceived by end users.

The study is designed to assess the market readiness of the leading Internet phone service providers by comparing the call quality of those VoIP providers in San Francisco and New York to traditional PSTN. The study compares the quality of Internet phone service providers based on 10 performance factors to accurately benchmark typical scenarios including placing coast-to-coast VoIP and local PSTN calls between New York to San Francisco and back as well as calls from a VoIP-enabled phone to a traditional phone.

Additionally, the study ranks leading providers on every major factor that affects the end-user experience with Internet phone service, including service availability, outage minutes, dropped calls, audio delay, listening quality, consistency, and geographic uniformity. The study also analyzes various diagnostic metrics that constitute the service performance factors and evaluates the impact of the underlying network carriers on the VoIP quality perceived by end-users.

The six leading VoIP providers included in the study are: AT&T CallVantage (news - alert) , Packet 8 (news - alert) , Primus Lingo, Skype (news - alert) , Verizon (news - alert) , and Vonage (news - alert) . To understand the impact of underlying network performance on call quality, the VoIP telephone calls are carried on three T1 networks: AT&T, Sprint, and UUNET. In addition to measuring underlying network performance, the study captures the impact of the last-mile on call quality by measuring each of the six providers on residential DSL and residential cable lines as well.

http://www.keynote.com

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Acterna Launches VoIP Test And Management Solution

Acterna (news - alert) recently announced comprehensive VoIP support features for its NetComplete Service Assurance solution. Driving cost-effectiveness, efficiency, and helping to generate revenue for service providers, NetComplete's VoIP capabilities are designed to deliver end-to-end network coverage by enabling performance monitoring, turn-up test, and troubleshooting functions that meet the challenges of delivering successful, efficient, and reliable voice-over-IP service.

NetComplete's VoIP capabilities are a key strategic focus of Acterna's Service Assurance Solutions (SAS) division. Acterna recently launched SAS to provide "triple play" next generation network operators with a robust, modular portfolio of interoperable applications for end-to-end quality of service management across their converged broadband IP infrastructure.

NetComplete enhances the efficiency and reliability of the service quality management processes by enabling VoIP service providers to obtain visibility of the end-customer voice experience throughout the entire lifecycle of a call. It allows service providers to perform critical service-assurance functions such as service monitoring, rapid fault isolation, and tier-two troubleshooting so they can effectively address the unique challenges of delivering carrier grade VoIP. In doing so, NetComplete helps providers identify, isolate, and overcome common quality of service (QoS) issues, such as poor call quality, dropped calls, and slow dial tone response.

As components of NetComplete, Acterna's NetAnalyst Test Management and NetOptimize Performance Management applications work in conjunction with the QT family of distributed test heads. Together, they provide VoIP demand testing and proactive monitoring by means of active call generation or passive call monitoring.

http://www.acterna.com

Serverless Peer To Peer VoIP Meets WiFi

By Johanne Torres

Popular Telephony Inc., (news - alert) a serverless peer-to-peer technology and Peerio provider, made news recently when it announced that Clipcomm (news - alert), a Korean manufacturer of VoIP solutions and supplier to Korea Telecom, licensed the Peerio middleware technology to enable end-to-end serverless operability in a range of WiFi and fixed-line IP terminals.

"Clipcomm's WiFi and Bluetooth technology coupled with Popular Telephony's serverless Peerio middleware creates a fully featured SIP- or H.323-based PBX system for enterprises that requires neither servers nor wires," said Dmitry Goroshevsky, CEO and founder of Popular Telephony. "As a result of this new partnership Clipcomm is able to bring a new approach to wireless telephony technology to the North American, European and Asian markets.

Popular Telephony explained that by embedding Peerio into the Clipcomm terminals Ethernet cables and switching hardware become a thing of the past, as the user just plugs in the terminals and the power cord to deploy a Peerio enabled Clipcomm telephony system. Clipcomm's built-in WiFi and Bluetooth capabilities provide the interconnection between fixed and mobile phones. The company claims that security and safety have also been designed into the system from conception, allowing gateways to drive phones using the power from the PSTN in case of catastrophic power failure in the office. Clipcomm phones also support a specialized high-performance DSP technology, as well as a range of different encryption systems including DH key exchange and various DES options.

¨We think this relationship between Clipcomm and Peerio will be a great opportunity for the market in the near future, and we are sure that this combination of the technologies will bring tremendous success to both Clipcomm and Peerio." says Randy Ahn, Sales Director at Clipcomm.

Peerio core technology eliminates the need for any centralized server and allows any IP phone, PC or other terminal to interconnect and materialize into a bespoke communications system that is self-servicing, self-healing, redundant, and secure. A Peerio intelligent device or system can support a wide range of telephony features and services by delivering up to 250 features and scaling to over 4 billion lines.

http://www.populartelephony.com
http://www.clipcomm.co.kr

Nextel Offers WiFi Service For Business Travelers

By Robert Liu

In a recent announcement, Nextel (news - alert)said it has partnered with Boingo Wireless (news - alert) and Wayport to offer the Nextel WiFi HotSpot service at more than 7,000 airports, hotels, convention centers, retail stores and other hotspot locations throughout North America.

The new service addresses "key concerns of individual customers as well as IT managers," said Greg Santoro, vice president, products and services, Nextel.

The service is enabled by Nextel's Connection Manager software client which sits in the user's PC. Unlimited service is available for $39.99 per month and the first month is free.

But if the customer moves out of the Boingo or Wayport's service areas, Nextel Connection Manager can automatically connect users to Nextel's data network via its new im240 PC card for an additional fee. However, because that service uses Nextel's pre-existing data network, the access comes at a rate of only 24-40 kbps and, bundled with the WiFi service, costs $54.99 per month.

http://www.nextel.com
http://www.boingo.com

Telchemy Intros Wireless And VoWiFi Handset Monitor

By Johanne Torres

Telchemy, Inc., (news - alert) a provider of VoIP Fault and performance management technology, announced it introduced what the company says is the first performance management technology suitable for direct integration into cellular and wireless handsets. The VQmon Release 2.2 supports the measurement of performance parameters for Push-to-Talk and other near-real time applications, adds support for key 3G cellular vocoders such as AMR narrowband and wideband, SMV and EVRC, providing a wide range of performance metrics.

Telchemy explained that it designed the VQmon/EP 2.2 for direct integration into IP based mobile and wireless handsets, IP phones, media gateways and residential gateways. VQmon/SA 2.2 provides cellular aware midstream monitoring for probes and analyzers and can be integrated directly into routers and session/ border controllers.

VQmon/4G bundles in the company's new non-intrusive video quality monitoring technology to provide a single solution for IP based Voice and Video solutions. The company said the system is ideal for monitoring multimedia cellular handsets. Telchemy says the technology will help wireless service providers to deliver higher-quality advanced video applications including gaming, streaming video, and podcasting.

"Mobile and IP telephony are converging, which is opening up opportunities for service providers and enterprises to deliver multimedia over wireless handsets," said Alan Clark, CEO of Telchemy. "VQmon Releases 2.2 and 4G meets the growing need for improved service quality to the mobile marketplace."

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

Telchemy is making available the VQmon/EP Release 2.2 and VQmon/SA Release 2.2 in June 2005, VQmon 4G will be available in August 2005.

http://www.telchemy.com

PT Intros Higher-Capacity SEGway Signaling Gateway

By Johanne Torres

Performance Technologies (PT), (news - alert) a developer of integrated systems, platforms, components and software, announced today it introduced the SEGway 4300 Signaling Gateway. The company this is its most robust and highest capacity signaling gateway system to date.

The company explained that as global SS7 traffic continues to increase, carriers are seeking ways to increase network capacities. PT believes that in order to facilitate a gradual migration from legacy SS7 circuits to next-generation VoIP, carriers require devices capable of supporting both traditional SS7 protocols and IP protocols. The company says its gateway products such as the SEGway 4300 would provide the signaling bridge to allow traditional TDM and next-generation, IP-based SS7 networks to work together seamlessly.

Performance Technologies' signaling gateways use the latest Internet Engineering Task Force (IETF) SIGTRAN standards to ensure reliability for these demanding VoIP applications. Scalable to 96 signaling links, the SEGway 4300 ensures OEMs and carriers a solution that can expand to meet the demands of high-growth, SS7 circuit-based or IP networks.

"The SEGway 4300 is our highest capacity signaling gateway offering to date. Our field-hardened signaling software combined with our robust hardware platform offers the flourishing VoIP market a flexible, cost-competitive solution for next-generation networks," said Deb Brunner-Walker, signaling product manager for Performance Technologies. "As global IP convergence moves closer to the mainstream, our gateways are being deployed around the globe. This latest offering was created to meet the burgeoning demands of ever-growing IP networks and new application providers."

The SEGway 4300 is offered in a 7U CompactPCI form factor. It offers 1024 virtual point codes and supports both circuit-switched and IP-based SS7 protocols. Beta trials for the SEGway 4300 are scheduled for July with general availability scheduled for September 2005.

http://www.pt.com

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TI's Digital Media Processors Enable High-Def Conferencing

At a recent industry trade show, Texas Instruments (quote - news - alert) demonstrated full-feature, high-definition multipoint conferencing based on the TMS320DM642 DSP-based digital media processor.

According to Codian CEO David Holloway, "High-Definition videoconferencing is a reality. As the HD resolution of 1280x720 at 30 frames per second becomes the new standard for videoconferencing, we will use our investment in the DM642 architecture to quickly and cost-effectively deliver HD quality multipoint video, recording and streaming to our customers."

TI's DM642 digital media processor is designed to provide developers with an optimized platform for enterprise-class videoconferencing applications that scale easily from client endpoint devices to the multipoint control units/gateways using multiple DM642 devices that enable IP-based videoconferencing across several sites. Multiple DM642s can be seamlessly connected via the 66 MHz PCI bus interface for high-speed connectivity. The DM642 digital media processors have numerous integrated I/O and streaming capabilities — including on-chip HD-capable video ports, glueless 10/100 Ethernet connectivity, multi-channel audio, and 66 MHz PCI connectivity — to develop true high-definition systems.

TI officials maintain that they are committed to the high-definition videoconferencing market and will continue to provide the hardware and software support essential for driving new services and enhanced functionality. DM642 processors running at 720 MHz are shipping in volume production today, and TI plans future code-compatible devices that offer higher performance and additional system integration to continue to bring cost-effective and scalable high-definition systems to market.

http://www.ti.com

Surf Announces SurfRider-812

Surf Communication Solutions (news - alert), has announced the launch of SurfRider-812, a fully-integrated high-capacity PTMC DSP farm comprised of eight digital signal processors (DSPs). The SurfRider-812 provides simultaneous Triple-Play media-processing capabilities for developers of voice and video gateways, CTI applications, media servers, and a multitude of other voice, video, fax, and modem applications.

Based on Texas Instruments' TMS320C6412 DSPs, the SurfRider-812 can process voice, video, and data (fax/modem) simultaneously, on the same DSP. It integrates with PCI, Compact PCI (cPCI) and AdvancedTCA (ATCA) carriers.

The SurfRider-812 can be used in a variety of products, including Voice and Video gateway applications connecting mobile, broadband IP, wireless (WiFi/WiMAX), and PSTN networks; CTI Applications, such as Voice, Video, and Fax mail, Interactive Voice/Video Response (IVR) servers, announcement servers, unified messaging servers, and call recording servers for Voice, Video, and Fax; Inter-Working Function (IWF); and Stand-Alone Termination Applications, such as RAS, Conferencing Servers, Voice quality monitoring, and non-intrusive interception and security applications.

http://www.surf-com.com

AudioCodes Intros SDK For AdvancedTCA Platforms

AudioCodes (news - alert) has announced the availability of its TP-12610 Software Development Kit (SDK) for AdvancedTCA applications. This new product is a development system for application developers seeking a quick start for an ATCA compliant platform using AudioCodes technology.

The ATCA PICMG 3.x family of specifications targets the requirements for the next generation of carrier-grade communications equipment providers. The TP-12610/SDK is designed to enable short time-to-market for software developers requiring a VoIP building block compliant with ATCA standards. The new design kit features a high-capacity blade of 2,016 channels and is based on AudioCodes field proven API common to all AudioCodes boards. Customers can use the same API for future platforms and thus reduce their development cycle and time-to-market.

The TP-12610/SDK is based on an AudioCodes cPCI VoIP board (either the TP-6310 or the TP-1610) hosted in a cPCI to ATCA adaptor. The adaptor-based board enables GB Ethernet base and fabric interfaces. The platform contains a 5-slot / 4U chassis with an IPMI shelf manager, a fabric (PICMG 3.1) and base switch blade, and an optional application processor blade.

http://www.audiocodes.com

Alcatel, Intel Collaborate On AdvancedTCA

Alcatel (news - alert) and Intel Corporation (quote - news - alert) announced that they have signed an agreement to improve time to market of Advanced Telecom Computing Architecture (or AdvancedTCA) platforms for mobile service providers. Both companies will further work together on the design and development of AdvancedTCA subsystem building blocks to increase standardization and component interoperability across the mobile telecommunications industry. The new Alcatel solutions resulting from this collaboration are expected to be available in early 2006, and use an Intel Pentium M processor-based AdvancedTCA single board computer.

The companies will engage in joint development activities in such areas as system architecture definition and board and solution design based on Intel processors, with the resulting products to be available across the industry. Alcatel will participate early in Intel's standard product definition process, ensuring a tight match between requirements and customer usage models. In addition to this implementation effort, the two companies remain closely aligned in their participation with standards bodies defining the specifications for modular communications platforms, such as PICMG for AdvancedTCA, Service Availability Forum for high-availability middleware and OSDL for Carrier-Grade Linux.

http://www.intel.com
http://www.alcatel.com

Wind River Linux Available For Artesyn Blades

Artesyn Communication (news - alert) Products, recently announced the availability of Wind River Systems' Platform for Network Equipment for Artesyn's KatanaQp AdvancedTCA telecom blade.

Wind River's Platform for Network Equipment combines Linux 2.6 and OSDL Carrier Grade Linux with Wind River's Eclipse-based Workbench development suite and a rich set of networking middleware. The platform is designed to assist telecom OEMs to develop and deploy network infrastructure equipment based on Artesyn's KatanaQp ATCA blade.

The KatanaQp is a high-performance ATCA telecom blade that combines two PowerPC MPC7447A processors with four PTMC expansion sites, IPMI-based system management, and a PICMG 3.1-compliant ATCA interface. The KatanaQp's high-speed PowerPC processors, switched fabric ATCA interface, flexible mezzanine expansion, and integrated system management make it easy to configure for a wide variety of control and packet processing applications, including WAN access, SS7/SIGTRAN signaling, media gateways, traffic processing, wireless base stations and softswitches.

Wind River's Platform for Network Equipment provides a complete implementation of Linux 2.6 with CGL 2 extensions. The pre-emptive kernel features high-resolution timers, fast user-space mutexes, and a native POSIX thread library. CGL v2 extensions include an Intelligent Platform Management Interface (IPMI), Hardware Platform Interface (HPI), heartbeat monitor, hot-plug, and Ethernet link aggregation failover. Networking support includes IPv4/IPv6, SNMP management, and a comprehensive suite of open source networking protocols and applications, including DHCP, FTP, HTTP, NFSv4, NTP, PPP, SCTP, Telnet, VLAN, SSL, SSH, and IPsec.

The KatanaQp, equipped with a single MPC7447A processor and a four-channel Gigabit Ethernet fabric interface, sells in OEM quantities starting at $3,498. Wind River's Platform for Network Equipment is available directly from Wind River Systems.

http://www.artesyn.com

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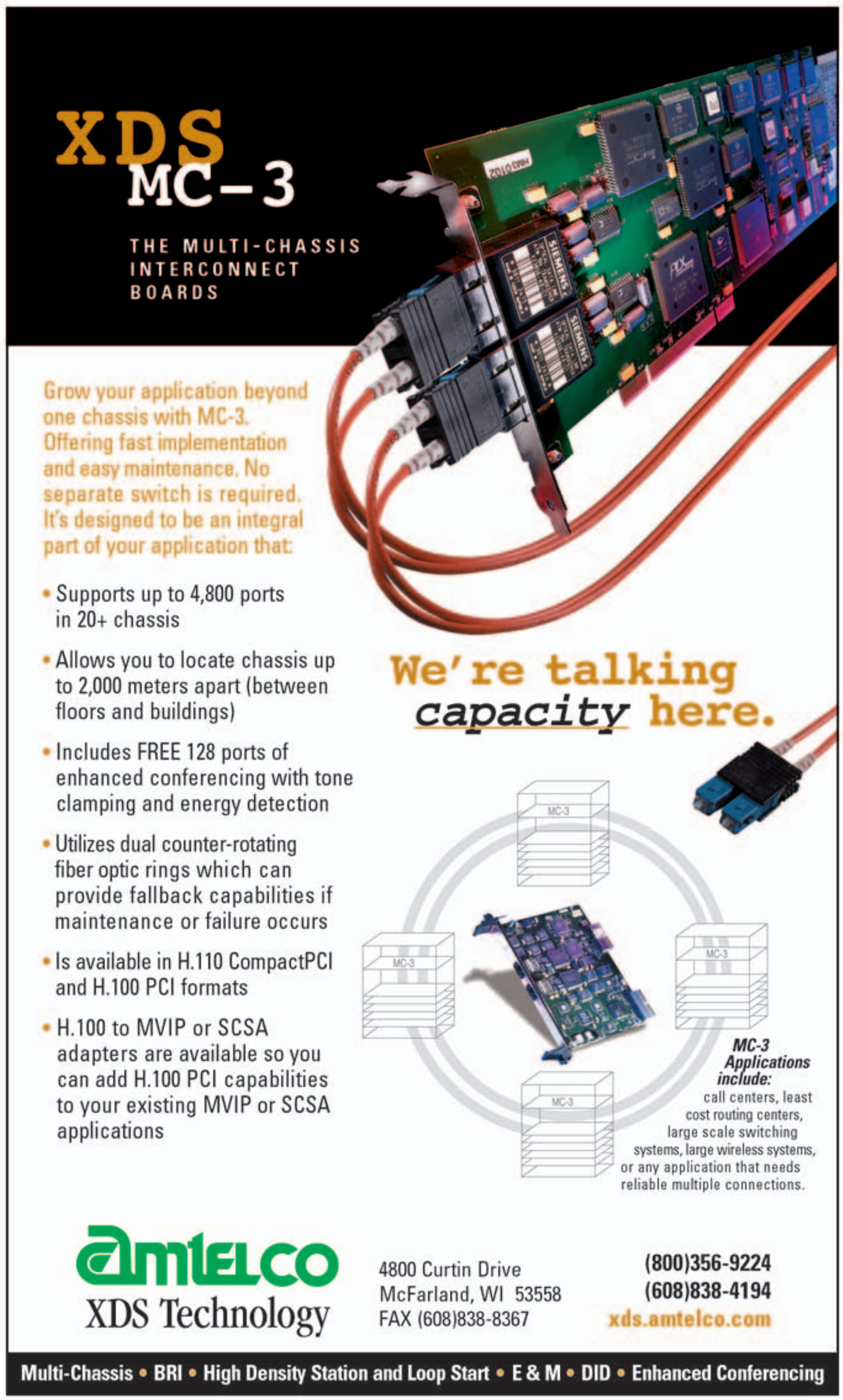

Ingate Intros Firewall 1180 And SIParator 18

By Johanne Torres

SIP security products provider Ingate Systems (news - alert) announced two new products: the Firewall 1180 and the Ingate SIParator 18. The company describes both as powerful tools that offer complete support for SIP-based IP communications, such as VoIP, to small businesses, branch offices, or home offices.

Ingate explained that both the Firewall 1180 and SIParator 18 have three ports and offer a 25 Mbit throughput. The company describes the products as small and versatile with no fan, making them virtually silent, therefore, eliminating the need for a separate server room.

“The Firewall 1180 and SIParator 18 bring VoIP, IM, and a host of SIP-based applications to the small business, branch office or home worker,” said Olle Westerberg, CEO, Ingate Systems. “With these products, any business, regardless of size, or location, can enjoy the benefits of increased productivity and the cost-savings associated with IP applications such as Internet telephony.”

Included with the 1180 are ten SIP user licenses that can be used for the registration of SIP user agents, such as phones and soft clients, on the SIP registrar. Five SIP traversal licenses also come standard, allowing up to five calls to traverse the Firewall at the same time. Additional SIP user licenses and SIP traversal licenses can be purchased at any time.

The Ingate Firewall 1180 offers control over SIP signaling, traffic, and network security. TLS encryption ensures privacy when communicating, making eavesdropping, call hijacking and call spoofing difficult.

The Ingate SIParator is a stand-alone device that connects to an existing network firewall to enable the traversal of SIP communications. Included with the SIParator 18 are ten SIP user licenses that can be used for the registration of SIP user agents, such as soft phones and soft clients, on the SIP registrar. Five SIP traversal licenses also come standard, allowing up to five calls to traverse the firewall attached to the SIParator at the same time. Additional SIP user licenses and SIP traversal licenses can be purchased at any time. The SIParator 18 can be configured as a part of the DMZ or in a standalone mode.

The Ingate Firewall 1180 and Ingate SIParator 18 are currently available for $900.

http://www.ingate.com

Genesys Announces SIP Contact Center System

By Johanne Torres

Genesys Telecommunications Laboratories, Inc., (news- alert) recently launched a new SIP-based contact center system. The system would provide entities with customer interaction control for any SIP-enabled infrastructure, regardless of vendor.

The company says it designed the Genesys SIP system to handle contact interaction control among SIP-enabled devices, such as gateways and endpoints. The system provides agent state tracking and monitoring functions, and delivers a set of interaction management functions needed in a contact center, including customer segmentation, call queuing, call routing, reporting, and call control. Furthermore, the solution takes traditional end-to-end IP calls and mediates them as a central IP server. Genesys explained that it is integrated into the open, standards-based Genesys Customer Interaction Management Platform for managing and tracking customer contacts from beginning to end.

"Our new SIP contact center solution extends Genesys' long-standing commitment toward platform-independence to the SIP world," explained Elliot Danziger, chief technology officer, Genesys. "This allows us to incorporate interactions from any SIP-enabled component and centralize the interaction data for improved reporting and management, delivering the full value of IP contact centers to our customers."

"From the beginning, Genesys has stood out in the contact center market by offering an open, software-based platform. By extending its open architecture into the IP contact center environment, enterprises are able to realize the significant cost and business benefits of IP, while utilizing a proven contact center solution," said Seema Lall, industry analyst, Frost & Sullivan. "Genesys has always been focused on open contact center software and this latest release demonstrates the company's commitment to be an industry leader in IP technology."

http://www.genesyslab.com

Nokia 770 To Support SIP-complaint VoIP

By Robert Liu

Nokia's (quote - news - alert) new 770 Internet Tablet is designed with VoIP capabilities, thanks largely to the combined efforts of fellow countrymen of the Finnish handset maker.

The Nokia 770 Internet Tablet, which connects to the Internet via WiFi or using Bluetooth with a compatible mobile phone, is built upon the DeviceLinux.org software applications platform, a license-free set of developer tools. Using this software environment, Helsinki-based Movial Corp. developed VoIP Connect — a third-party, SIP-compliant application designed to run on the 770's Linux environment.

But even though Skype also offers a Linux-based download for the desktop, Movial's CEO doesn't view the popular peer-to-peer (P2P) application as a competitor to his own product because his revenue stream is based on OEM agreements with handset makers and operators.

"For the operators, it's a lot better to rely on SIP than on peer-to-peer," Jari Ala-Ruona, Movial's CEO, explained.

According to Ala-Ruona, VoIP Connect supports both iterations of SIP: IETF and IMS. While the IETF's model for SIP is more widely used as the industry standard, cellular operators have adopted an infrastructure that uses IMS in order to better control access and usage of their closed networks.

The PC client is available immediately. VoIP Connect for the Nokia 770 will be available this summer. The product was never announced because Movial didn't want to pre-empt Nokia's launch of the 770.

Additional features of the Nokia 770 Internet Tablet include an Internet Radio, RSS News reader, Image viewer, and Media players for selected types of media. The Nokia 770 Internet Tablet is planned to start shipping in the third quarter of 2005 in selected countries in the Americas and Europe.

http://www.nokia.com

UCS Solutions Implements FrontRange Contact Center

By Johanne Torres

UCS Solutions (news - alert), an IT systems provider to South African businesses, announced that it implemented a contact center system provided by FrontRange (news - alert)that would enable the company to enjoy advances in the productivity and effectiveness of contact center staff. Thanks to the partnership between both companies, UCS Solutions and its 40-consultant contact center will now be able to provide support to about 15,000 users at more than 20 major customers.

Debby Webster, Customer Contact Center manager at UCS Solutions, explained that the company signed up as pilot implementers of FrontRange Contact Center in May 2004 and were fully aware of the challenges and risks associated with being the first HEAT customer globally to implement a new, mission critical application.

"We're also in the IT business. We knew that being the first with a new application would have its challenges. However, we drew on the strength of our five-year partnership, experience of their HEAT application and their knowledge of contact center business to deliver a great solution," Webster added.

It took UCS Solutions several months of testing and implementing the system. After this lapse of time, the company reported it was "now ready to respond to requests to visit the facility from people who are interested in seeing at firsthand how it works," stated Webster. "Quality monitoring and training is most important in any contact center. FrontRange Contact Center gives us the ability to ensure that consultants take calls on specific clients only when they are properly trained to do so. This ensures a quality service to our clients."

UCS Solutions is currently using the voice recording on demand functionality within FrontRange Contact Center, but plans to expand this to recording all calls soon.

UCS Solutions described the FrontRange Contact Center as a system that integrates with any other service and support application. The company found that the fast integration with the existing HEAT implementation a further benefit.

Future development for UCS Solutions could be to expand the use of FrontRange Contact Center to its Cape Town and Durban facilities, enabling the functionality of FrontRange Contact Center to be applied to the same queue of calls from any location. "Ultimately, we'll be able to exploit the potential of the idea of the virtual contact center, where individual consultants are able to take calls from home, for example. We plan to be ready for the growth and possibilities of VoIP by having the technology and the systems in place to make VoIP work effectively," explained Webster.

"It's not often that a totally new development such as FrontRange Contact Center is implemented first in South Africa, and we're especially pleased to have been able to extend and deepen our strong existing partnership with UCS Solutions during this implementation. Since this deal, FrontRange Solutions has closed a further 15 sites to date, thus vindicating the enormous potential we all see with this product," said Tracey Newman, managing director of FrontRange Solutions South Africa.

"The FrontRange Contact Center provides comprehensive telephony functionality in a single modular package that allows organizations to simply plug directly from Telkom into a SIP (session initiated protocol) compliant voice gateway and from there to a single server. It eliminates points of failure and extensive maintenance, makes upgrades easy and slashes implementation time and cost," concluded Newman.

Today's news follows FrontRange's recent extension of its global access of its new modular IT Service Management (ITSM) system with a new version and languages including English, German, Russian and Polish. The company announced this month that French and Chinese will be made available soon. The company said the release of ITSM 5.0.2 is designed to improve the performance of IT and support organizations for new customers as well as offer additional modules to increase functionality for HEAT customers.

Michael McCloskey, CEO of FrontRange stated: "We will continue to add significant new functionality and languages to our new offerings to extend user's access as well as our global reach. These solutions, to be introduced over the course of the year, will offer our customers an exceptional opportunity to obtain enterprise-class functionality at an exceptional value."

http://www.frontrange.com

Nuasis Integrates IP-Based Contact Center With Siebel CRM

By David Sims

Nuasis Corporation (news - alert), the IP contact center company, has announced that it has successfully integrated its software-only, IP-based contact center system with Siebel CRM (quote - news - alert). The Nuasis NuContact Center handles customer inquiries via the phone, e-mail, Web and fax. The integration of the Siebel CRM application with the software-only, IP-based Nuasis contact center system extends the CRM investment and should increase contact center efficiency and productivity.

When CRM applications are integrated with the NuContact Center, companies can “pop” existing customer information from the CRM database onto the contact center agent’s desktop, thereby providing the agent with detailed customer information so that the call/e-mail/Web chat can be handled faster and more efficiently.

Nuasis customers are reporting that the combination of the IP-based contact center integrated with their existing CRM applications allows them to shave 30–45 seconds from every customer contact.

“Our ability to quickly deploy integrations with enterprise applications such as Siebel is a result of our software-only, single network product architecture,” said Kevin McPartlan, vice president, product direction, Nuasis. “Our choice to use open standards such as VoIP, SIP, and SOAP rather than proprietary CTI protocols simplifies deployment and decreases implementation times from many weeks to days. The open standards, software-only, model dramatically lowers the cost of CRM integrations.”

Nuasis reports that nearly 100 percent of its customers are integrating the NuContact Center with CRM applications. For Nuasis customers, no costly hardware or middleware, nor CRM APIs are required. Furthermore, there is also no requirement for complex and costly professional services.

As a single, distributed system, the NuContact Center is designed to replace multiple legacy ACD systems. It intelligently routes and queues customer contacts across multiple geographically dispersed service centers taking full advantage of VoIP technology.

http://www.nuasis.com
http://www.siebel.com

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Siemens Announces Contact Center App Upgrades

By David Sims

Siemens Communications, Inc. (quote - news - alert) has announced Versions 6.5 of its HiPath ProCenter Agile and ProCenter Standard contact center applications, marketed towards enterprise customers small to large and across myriad industry sectors in more than 60 countries.

The Siemens HiPath ProCenter V6.5 portfolio is designed to provide faster call center administration, call processing and multimedia routing and reporting, including enhanced user and management visualization tools.

Presence-driven and permission-based collaboration tools have also been extended to reach enterprise-wide communication sources such as the telephone, e-mail and instant messaging.

The upgraded HiPath ProCenter Agile solution, with support for as many as 150 active agents, is designed for small and mid-sized enterprise contact centers or informal call handling groups. Its new features include e-mail management, scheduled callbacks and agents in multiple groups. It also provides integrated design capabilities for a basic IVR, and integration with Microsoft CRM.

As a ready-to-run solution, Agile is engineered to be easy to implement, configure, and use, delivering intelligent call routing, graphical reporting, and productivity tools for both call handling agents and managers.

Streamlined for call handling and communication using a smaller footprint, the HiPath ProCenter Agile solution includes an associate desktop that can extend intelligent call routing features to employees who serve as overflow agents during peak traffic periods.

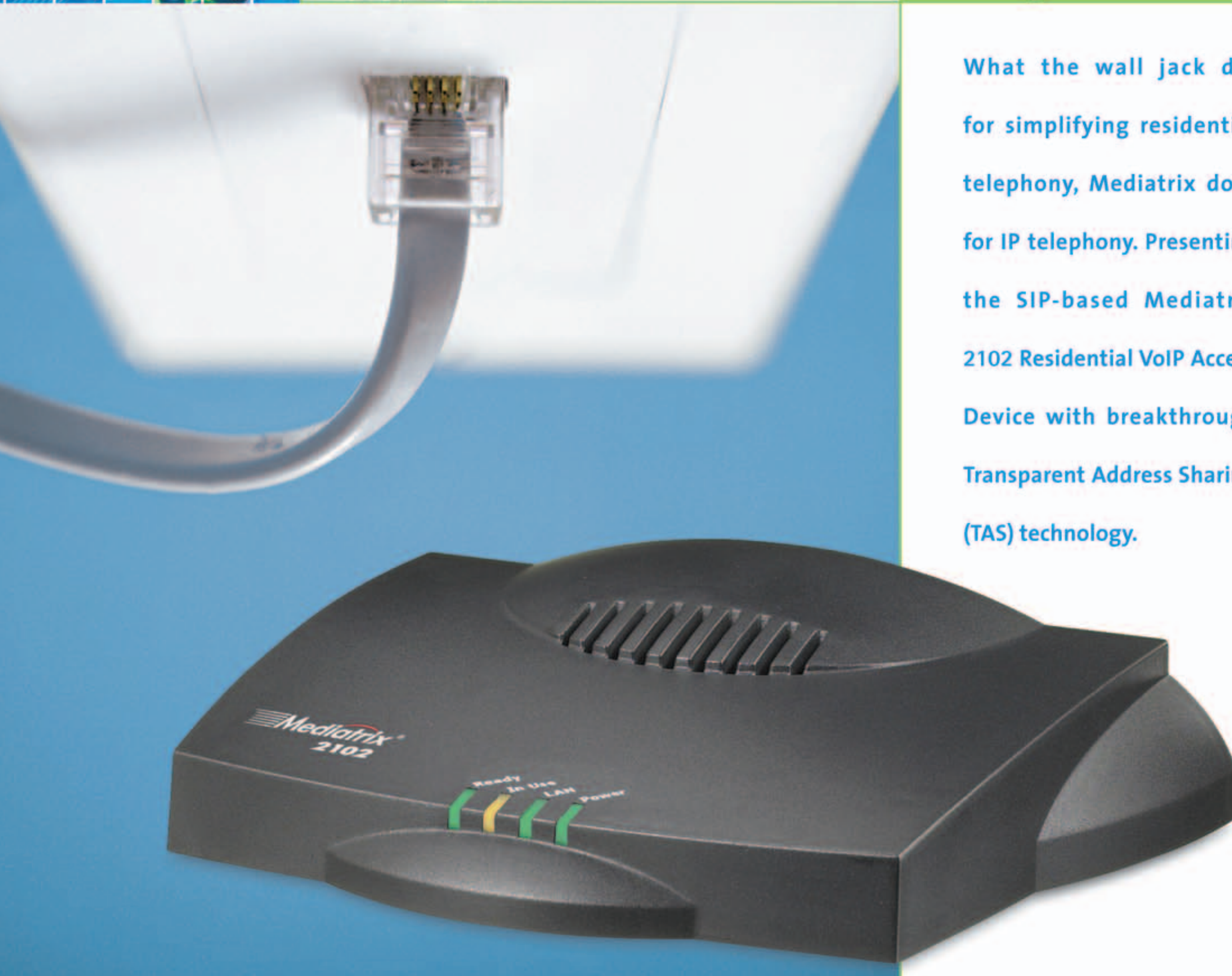
The upgraded HiPath ProCenter Standard solution, scaling up to 750 active users, includes enhancements such as full IVR support, multimedia skills-based routing and a software developer toolkit for vertical business process integrations.

As with the HiPath ProCenter Agile, presence tools such as Team List and Team Bar are integrated into the Standard's application desktop, allowing agents to visually monitor the real-time availability of other enterprise agents, managers and subject matter experts. As needed, agents can engage colleagues with the solution's one-click collaboration capabilities, whether they are in the next office or remotely connected to the enterprise via an IP network.

http://www.siemens.com

Residential VoIP... simplified

What the wall jack did for simplifying residential telephony, Mediatrix does for IP telephony. Presenting the SIP-based Mediatrix 2102 Residential VoIP Access Device with breakthrough Transparent Address Sharing (TAS) technology.



Keep it simple with Mediatrix

The Mediatrix 2102 does away with conventional wisdom to enable standard home phones, faxes and PCs to simultaneously access the Internet - using a single broadband connection and without the need for a router or a NAT.

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AltiGen To Partner With Commtech For Irish IP

By David Sims

AltiGen Communications, Inc. (news - alert), has announced it is partnering with Commtech, a distributor of Internet, networking and security products in Ireland, to distribute AltiGen's award winning IP phone systems and call center tools to the Irish market.

The partnership is another step in AltiGen's continuing strategic plans for growth in Europe.

"We see Commtech as a strategic distribution partner to support the unique needs of this emerging market," said Mike Plumer, AltiGen's Vice President of Sales. Commtech will be a specialist distributor of AltiGen technology and a key component in delivering complete VoIP and IP convergence solutions in the small and mid-sized enterprise market.

Commtech will also provide independent advisory and installation services to support the AltiGen reseller channel community.

Justin Owens, Managing Director of Commtech thinks AltiGen's "superior product offering" is "well suited to the Irish market."

http://www.altigen.com

Alcatel And Polycom Converge Voice, Video, And Data

By Johanne Torres

VoIP technology providers Alcatel (news - alert) and Polycom, Inc. (news - alert) announced that they have partnered to deliver converged collaboration systems to enterprises. The multi-year agreement will bring together Polycom's video conferencing systems with Alcatel's unified communications offering. The companies explained that they will now focus on delivering enterprise users integrated collaboration solutions that span voice, video, and Web conferencing.

With the new system in place, enterprises will now be able to instantly connect with people and workgroups through an instant messaging client or via the telephone to launch rich media collaboration sessions that include video. Users will also be able to escalate from one form of communication to another with just one touch, adding video to an Alcatel phone call or audio and video to an instant message.

"Alcatel is committed to delivering SIP-based collaboration solutions to enterprise users, enabling them to instantly initiate rich media sessions regardless of the device they are using and regardless of their location," said Jean-Luc Fourniou, senior vice president, Alcatel enterprise solutions activities. "Partnering with Polycom supports our strategy to deliver video as part of our converged collaboration solutions, improving both individual and workgroup productivity for enterprises."

http://www.alcatel.com
http://www.polycom.com

Nortel And IBM Sign Deal To Broaden VoIP Offerings

By Johanne Torres

Nortel (quote - news - alert) and IBM (quote - news - alert) announced that the companies agreed on a deal designed to jointly support customized products across a range of market segments. As the first step in this technology, research and services relationship, the companies will establish a Nortel-IBM Joint Development Center based in Research Triangle Park, North Carolina, to collaborate on the design and development of new products and services.

At the Nortel-IBM Joint Development Center, personnel from both companies will work to accomplish the following goals:

- Work together to enhance and extend current products, drawing from various divisions within Nortel and IBM, to drive new revenue growth while reducing R&D costs
- Collaborate on focused research on a project-by-project basis, enabling a new level of product creation by tapping the deep skills of each firm to introduce solutions more rapidly
- Work together on technology, initially, a new class of blade servers that would meet the specific and demanding data flow, reliability and security needs required by the network equipment marketplace embracing next generation network solutions.

http://www.ibm.com
http://www.nortel.com

By Marc Robins

The Telco IP-based Triple Play: Leapfrogging The Cable MSOs?

The telco triple play — the delivery of voice, data, and video services over a common network infrastructure — has become the rallying cry of an industry under assault from multiple fronts. Indeed, with their POTS (define - news - alert) revenue at risk from upstart VoIP (define - news- alert) providers and Cable MSOs with their VoCable offerings, and DSL uptake significantly trailing cable broadband, the RBOCs and the majority of local, independent telcos are rapidly facing their moment of truth: either upgrade their network infrastructure and expand service offerings or endure a slow and steady contraction of their subscriber base and revenue streams.

With the lines drawn in the battle, the strategy is clear: Telcos must add their own video services — including video telephony, broadcast video, and video-on-demand — to their current voice and data offerings in order to effectively compete in today's marketplace. To highlight the growing threat, consider that broadband cable providers have taken more than 38 percent of the broadband Internet market and 2.2 percent of the local telco market, according to industry sources, and this trend is expected to grow dramatically if left unchecked. In Omaha, Nebraska, the number one telephony provider today is Cox Communications, the city's leading cable MSO.

Different Paths, Same Destination

The big four RBOCs — Verizon, SBC, BellSouth and Qwest — are all devising different network architectures to deliver the triple-play service bundle. For example, while Verizon seems to be favoring Fiber to the Home (FTTH), the other RBOCs are focusing primarily on Fiber to the Node (FTTN), with new DSL-based delivery mechanisms that will allow them to leverage their existing DSLAMs and copper-based local loops, while reserving FTTH for mostly "Greenfield" deployments.

A FTTH triple-play architecture — with fiber traveling all the way to subscriber's households — offers virtually unlimited bandwidth. In contrast, FTTN architecture employs fiber up to the local loop distribution point. From that point on, the service would run over copper loops to the subscriber using DSL. The advantages of this approach are that it leverages the copper infrastructure as well as the telcos' sizeable investment in DSL technology. The downside is that copper local loops can't support anywhere near the same bandwidth as fiber.

The challenge for the telcos then, even with new, more efficient digital video-encoding algorithms like MPEG-4, and new higher-capacity flavors of DSL such as ADSL2+, VDSL, and VDSL2, is that there will still be a need for even greater capacity, considering that their infrastructure will have to support not only a multitude of standard and high-def video channels, but also broadband Internet and voice.

The Promise Of True IP-Based Triple Play

To solve this dilemma, IP is emerging as the telcos most potent weapon in their technology arsenal. Indeed, by leveraging existing Ethernet and packet-based technologies, telcos can efficiently deliver a triple-play service bundle over their existing DSL connections in a highly flexible and cost-effective manner.

From a network equipment perspective, we are seeing the arrival to market of a host of new, IP-centric products that essentially form the puzzle pieces for a complete migration from a circuit-switched to packet-based network infrastructure. Products such as softswitches for voice services, digital head-ends for IP video, and broadband loop carrier systems are all serving to enable the convergence of voice, data, and video on a single access network infrastructure. From a services perspective, an array of IP-based applications are also available, including VoIP, Internet access, virtual private networks (VPNs), and other data services, video telephony, and conferencing, and broadcast video (IP TV) and video-on-demand.

Perhaps the greatest benefit of the triple-play for telcos is the ability for them to provide a host of new enhanced services and applications that would have been impossible for them to offer before over their relatively low speed, inflexible legacy network infrastructures. Not only will a true IP triple-play allow telcos to enter the entertainment marketplace with video on-demand and IPTV, but the integration and bandwidth improvements will also help facilitate and promote a host of new and unique services, including video phone and video mail, e-learning, medical alerts, and SMS alerts on mobile devices of incoming calls to the home phone.

Telcos must add video services to effectively compete in today's marketplace.

Indeed, the potential even exists for telcos to be able to leapfrog over cable MSOs and satellite providers in the race to provide the most advanced service bundle in the marketplace. **IT**

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

By Tony Rybczynski & Cherif Sleiman

Where's The Wi In WiMax?

If you are like many in the industry, you're probably confused about WiMax (Worldwide Interoperability for Microwave Access). You know that 802.11 WLAN standards are focused on Ethernet and handle Ethernet frames up to 1,500 bytes. WiMax, on the other hand, or IEEE802.16 as it is formally known, is designed to seamlessly carry any higher layer protocol, such as ATM, frame relay, Ethernet, or IP, and supports frame concatenation and is fundamentally a higher power longer range technology. So what's the relationship between WiMax and WiFi? Will WiMax replace WiFi?

WiMax In Your Neighborhood

The 802.16 spec is full of optionality, only a subset of which is needed for typical deployments directed at specific markets. Adaptive burst profiles are used to increase the system capacity. The frame structure allows terminals to be dynamically assigned uplink and downlink burst profiles to allow trade-offs between capacity and robustness in real-time. In contrast, 802.11 is very much focused on one general environment: wireless LANs, whether in buildings or hotspots, or more recently across campuses through wireless mesh networking. Consequently, a WiFi-enabled device cannot talk to a WiMax base station; one is not a subset of the other.

WiMax (define - news - alert) provides a range of mechanisms targeting bandwidth efficiency. For example, multiple user frames can be concatenated into a single burst to reduce bandwidth overhead. In contrast, IEEE802.11 follows the KISS principle and addresses bandwidth capacity through new development such as 802.11n, which targets full 100Mbps bandwidth to the mobile user.

WiMax operates in various licensed and unlicensed bands between 2 and 66 GHz, including the unlicensed bands of 2.4 and 5GHz used by WiFi. This highlights a third major difference. The full potential of WiMax over tens of miles with aggregate bandwidth of 70Mbps (or 1 Mbps per user) is only achievable at the higher power levels supported in the licensed bands, often requiring line of sight. Licensed operation is only available to service providers and specific operators (e.g., public safety agencies). Operating in the unlicensed bands limits the applications of WiMax to a three-to-five mile radius, initially for fixed applications. WiFi coverage is typically for cells of 100 meter radius.

A WiFi-enabled device cannot talk to a WiMax base station; one is not a subset of the other.

IEEE 802.16 is initially targeted at fixed broadband wireless access systems employing a point-to-multipoint architecture, and seamlessly carrying protocols such as frame relay, ATM, Ethernet, or IP. The largest application is as an alternative to DSL (define - news - alert) and cable modems for public network access for consumers and small business users. It is not initially for mobile users, though it can be used as an alternative to wired solutions between buildings. Therefore, WiMax and WiFi are different technologies targeting different markets.

WiMax WiFi Convergence

WiMax and WiFi (define - news - alert) are converging in three ways: network convergence, voice convergence, mobility convergence.

Network convergence brings WiFi and WiMax technologies together to expand geographic scalability. In an environment in which wired Ethernet to each Access Point (AP) is cost prohibitive (e.g., across a campus or shipping yard), WiMax can be used to wirelessly backhaul APs to a central wired base station. This is an alternative to Wireless Mesh Networks, whereby WiFi APs dynamically learn about each other and wirelessly route packets reliably across this mesh. WiMax provides better reach, but mesh solutions typically have lower operational costs (via plug-and-play) and higher reliability (via dynamic routing).

Voice convergence expands the application fit of wireless solutions through QoS. WiFi is doing this through a new standard called 802.11e. WiMax has developed a number of WiMax specific mechanisms for QoS. This includes frame fragmentation, a self-correcting bandwidth request/grant scheme that eliminates the overhead and delay of acknowledgements, and polling mechanisms. IP telephony can therefore run over both WiFi and WiMax networks.

Mobility convergence is inherent in WiFi across APs in a single IP subnet, has been extended to enterprise-wide roaming, and is being extended to seamless roaming for voice and data between enterprise and public 2G/3G cellular networks. WiMax is moving from fixed broadband access to mobility

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through an extension called 802.16e (with initial rollouts in 2006). While mobility in a WiFi environment supports mobility at ambulatory speeds, in a WiMax environment, it is intended to support speeds that are an order of magnitude greater. Once developed and mature, these could compete against 3G wireless solutions. We can foresee multimode mobile devices that support WiFi, WiMax, and 3G wireless roaming.

So where's the Fi in WiMax? WiMax is not a replacement for WiFi, nor is it interoperable with WiFi. In fact it's more complementary to WiFi by providing reach and more competitive with third-generation public cellular technologies by providing wireless bandwidth over long distances. But these two technologies are converging with respect to network solutions that integrate both, with respect to supporting voice and with respect to mobility, including convergence in mobile devices.

WiMax and WiFi are converging in three ways: network convergence, voice convergence, mobility convergence.

What WiMax now needs is acceptance by a major economic power. GSM (define - news - alert) was propelled by Europe, CDMA (define - news - alert) by the U.S. and Korea. But what would happen if the emerging giants of India or China adopted it as their standard? IT

Tony Rybczynski is the director of strategic enterprise technologies for Nortel. He has over 30 years experience in the application of packet network technology. Cherif Sleiman is technology strategy leader for Nortel's Enterprise business unit, responsible for technology roadmaps associated with multimedia networks. For more information, please visit the company online at http://www.nortel.com.

By John Cimko

Bringing Affordable Broadband Services To All Americans

The federal government wants everybody to have high-speed, affordable broadband service. And the government is banking on competition to make it happen.

President Bush announced last year that he's "talking about broadband technology to every corner of our country by the year 2007 with competition shortly thereafter." NTIA Administrator Michael Gallagher is pushing to reach this objective: "We're going to drive hard to meet that goal [by] removing the regulatory underbrush so that a competitive environment will deliver on the President's goal."

Good luck.

If the government's plan is to rely on the marketplace to bring affordable, high-speed broadband to every corner of America, then many low-income and rural consumers may find themselves on the wrong side of the "digital divide."

The government's broadband strategy appears to hinge on encouraging "intermodal" competition (cable companies against regional Bells, for example) while also discarding policies that promote "intramodal" competition (regional Bells against competitive local exchange carriers, for example). MCI's Richard Severy argues that recent FCC decisions substantially restrict intramodal competition. "They've essentially bet the country on intermodal competition."

The government argues that cable competing against the regional Bells' DSL (define - news - alert) services is sufficient to deploy affordable, high speed broadband. The government is also counting on market entry by fixed wireless, satellite, and fiber services, and by broadband over power lines. Meanwhile, the "regulatory underbrush" (mainly rules enabling competitive LECs and others to provide broadband by requiring access to incumbent carriers' facilities at regulated prices), must be removed because it distorts the market and dampens incumbent carriers' investment in broadband infrastructure. The government also opposes promoting intramodal competition by requiring cable companies to give other Internet service providers non-discriminatory use of their broadband pipes.

How are the government's policies doing so far? Not so great.

According to International Telecommunications Union data for 2004, the U.S. ranks sixteenth on the broadband penetration chart (with a rate of 11.4 subscribers per 100 population). Business Week criticizes the FCC for creating "a cozy duopoly of broadband providers: the Bells and the cable-TV companies," and recently reported that about 20 percent of Americans have no way to get broadband, and another five to 10 percent have only one choice, their local cable company.

The picture doesn't get any better when you look at speed and price. FCC Commissioner Michael Copps, who says that broadband is "the most central infrastructure challenge facing the country," notes that Japanese consumers get 8 Mbps for $10 per month. Most Americans must pay $30 to $40 per month for 1.5 to 2 Mbps. According to Commissioner Copps, South Korean consumers get 10 Mbps for the same price US consumers pay for 1.5 Mbps. South Korea, incidentally, tops the broadband penetration chart at 24.9 subscribers per 100 population.

SBC's recently announced DSL price cuts suggest the picture could change. SBC will offer 1.5 Mbps for $15 per month, and 3.0 Mbps for $25. But Broadcasting & Cable reports there are catches. Subscribers must take service for at least one year and also subscribe to SBC's phone service. After the first year, DSL rates go up to $50 per month for 1.5 Mbps and $60 for 3.0 Mbps.

Everyone agrees that broadband is a powerful engine. Mark Cooper of the Consumer Federation of America says broadband is critical because the Internet is "a technological revolution that promises to enhance productivity… and increase the standard of living for all those who use it." Former Congressman David McIntosh says the U.S. could create more than one million high-tech jobs through universal deployment of broadband. But many Americans are missing the Internet's full potential because of steep broadband prices and lagging broadband deployment in less populated areas.

The government has set the right goal in aiming for universally available and affordable broadband. But are we following the best path toward that goal? The problem with depending on intermodal competition as the driver for deployment of cheap, fast broadband service is that a market dominated by two players — cable and the regional Bells — may not produce the best results. The current picture underscores this concern.

Other countries are outpacing the U.S. in delivering inexpensive, high-speed broadband. If we want to move up the broadband chart, then the FCC and NTIA should reexamine the policies needed to achieve the government's goals.

One place to look is Japan. Ikeda Nobuo, a Japanese economist who has studied government efforts to promote DSL broadband in Japan and the US, reaches these conclusions. First, unbundling and co-location requirements may not be enough, but they are essential prerequisites to making DSL widely available and affordable. Second, the requirements must be vigorously enforced to encourage entry and counter the obstructionist tendencies of incumbents.

Third, based on Japan's experience, if unbundling and co-location requirements are adequately enforced, these policies will not only encourage entry but will also promote investment by incumbent carriers. Finally, the deregulatory, market-based approach taken by the U.S. may be problematic. "The FCC's new policy to put an end to unbundling," Mr. Nobuo says, "is likely to make things even worse."

Time will tell whether President Bush's broadband goals are met by 2007. But there's a risk that the regulatory underbrush now being cleared away by government agencies could have been a fertile seed-bed for achieving these goals. 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.

Enterprises, Contact Centers Embrace VoIP

By Greg Galitzine

VoIP is perhaps the fastest-growing segment of telecommunications and perhaps even the hottest segment of technology in general. Is that a bold statement? Maybe, but a quick perusal of trade magazines and mainstream publications alike is all that it takes to realize that VoIP (define - news- - alert) is on everyone's mind.

Several segments of the VoIP market show more promise than others. Enterprises and contact centers are at the forefront of those companies making the move to VoIP and are deploying IP-based phone systems in ever increasing numbers. The first thing most people realize about VoIP is that it can save their business money by reducing or eliminating the toll charges for long-distance and even local calling. Indeed the cost of calling between and among far-flung branches of an enterprise can be completely eliminated. However VoIP is much more than simply an opportunity to lower your company's phone bill. There are many so-called "soft" benefits enabled by VoIP, such as increased worker productivity, the ability to collaborate among multiple branch offices, lower operational expenditures as a result of simplified management schemes, and more.

One oft-overlooked key benefit of VoIP in the enterprise is the ability to leverage IP to integrate tightly with the applications and business processes already in use at your company. Imagine the advantages of being able to integrate your phone system to work hand in hand with your CRM or contact center applications. What about the benefits associated with tying your voice platform to your company's audio or video conferencing platform? And what about deploying these advantages to your remote offices and home-based employees? It seems obvious, doesn't it?

CRM

Combining a company's VoIP system with CRM (define - news- - alert)and contact center applications can lead to a number of benefits, including the ability to better deal with customer queries and the ability to provide superior customer service. Access to a customer's history and other pertinent information is critical, and IP helps facilitate the transfer of that information direct to the agent in both formal and informal contact center settings. When integrated with CRM solutions, VoIP empowers every employee to deal with any and all customer inquiries.

CRM delivered on an IP Platform allows companies to follow a more distributed call center environment, scaling up or down headcounts in many different geographically disparate centers, rather than operating a few large centers, allowing companies to take advantage of the labor cost benefits in more rural areas, or the time zone spread for more extensive customer care hours, while still sharing the same workforce management system, call recording facilities, e-learning system, database and CRM apps among all employees.

Skills-based routing can be vastly improved; both call and customer data can be transferred to the most appropriate agent in the most appropriate center much more seamlessly, and invisible to the customer. Better skills-based routing improves time-to-answer, first-call resolution, and customer satisfaction metrics.

Many of these advantages were simply heretofore not possible. And by leveraging IP phone systems that are integrally tied to CRM applications and other business processes, the benefits are indeed enticing.

Conferencing

Working from home used to be a frustrating experience, fraught with slow connection speeds and a lack of truly integrated applications that would enable people to be productive. Today, with broadband pipes extending to most any place an employee might set up shop, and the availability of tools such as presence and conferencing and collaboration applications, working remotely has taken on a new meaning.

Isn't that what VoIP is all about? Better business decisions? Better results?

Market research tells us of an increase in remote, distributed and mobile workers and therefore there is a concomitant need to fully integrate them and enable them to access all the applications they're used to in an office environment — including integration with voice. Increasingly we're seeing these products and we're seeing companies integrating them to offer solutions to enterprises.

According to Taylor Collier, senior director for product management at Microsoft Corp.'s Real Time Collaboration business unit, integrated conferencing solutions, such as Vonexus' Live Conference and the Microsoft Office Live Meeting product, "…will help organizations better communicate and collaborate with geographically dispersed employees, partners, and customers - leading to better business decisions and better business results."

And after all, isn't that what VoIP is all about? Better business decisions? Better results? VoIP enables enterprises to better integrate business processes with voice with the end goal of becoming more efficient and more productive.

Greg Galitzine is the editorial director of Internet Telephony magazine.

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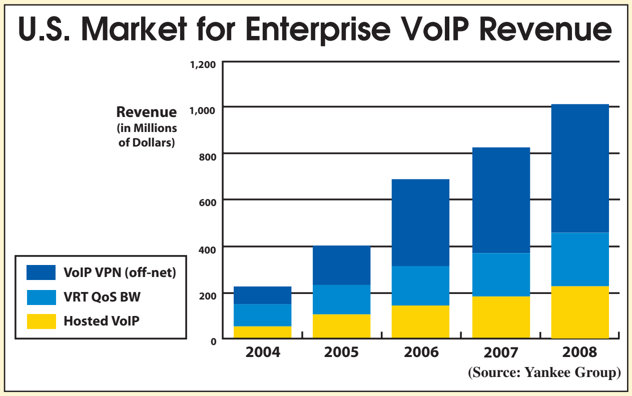
Building Upon the Value of Microsoft's Integrated Approach

Microsoft Doesn't Offer an IP Phone System. Finally, However, Enterprises and Contact Centers Can Integrate IP Telephony to Their Microsoft Platform.

Read the name Microsoft and "data" is probably the first thing that comes to mind. Or talk to an IT director and you'll likely hear "integrated," as in Microsoft's integrated approach to managing data, using solutions from the software maker's integrated server platform and distinct product families.

What you don't normally read or hear in mainstream Microsoft circles, though, are "phone system" and "IP telephony" — even though Microsoft has long been a proponent of the Session Initiation Protocol (SIP) that supports voice over IP, and continues to boost VoIP functionality in its Windows CE operating system for IP-enabled client devices.

Now, however, Interactive Intelligence has made phone calls possible on the Microsoft Windows Server System platform with an IP contact center application suite and open software architecture that welcomes Microsoft integrations. Furthering this effort, Interactive Intelligence launched its Vonexus (news - alert) subsidiary in June 2004 to introduce the only 100 percent Microsoft-based IP phone and communications system available today, the Enterprise Interaction Center (EIC).



Any contact center or business that relies on Microsoft's solutions can therefore get a single pre-integrated server to manage voice interactions as well as data using a SIP-driven network for VoIP. The same server also gives them plenty of room and power to integrate their existing Microsoft applications, along with any new ones down the road. Better still, this single-server approach is applicable for corporate offices as well as branch locations and remote and mobile workers.

Moreover for the most critical part of any business — the bottom line — building upon the value of Microsoft's integrated approach with Interactive Intelligence and Vonexus lets you:

- Leverage your current Microsoft investment. Rather than deploying a PBX that doesn't integrate with anything else or is soon to be obsolete, simply build upon the Microsoft investment you've already made by integrating IP functionality to your Microsoft server system.

- Add telephony to your Microsoft applications. Desktop Business Client integrations from Interactive Intelligence and Vonexus let you add call management capability to Outlook, Microsoft Business Solutions — Great Plains, and Microsoft CRM with no need to add new applications. A .NET "thin" client strategy also lets you universally deploy Client functionality to distributed and remote users with zero-effort while reducing bandwidth requirements and eliminating the need to install software at each desktop.

- Utilize Microsoft's full range of products. The IP solutions from Interactive Intelligence and Vonexus integrate with Microsoft Business Solutions applications, products from the Windows Server System, and integrate Exchange Server and Outlook for unified communications. EIC also integrates to Windows XP, Office Online applications, and Windows Mobile devices.

- Reduce customization and implementation. Integrations to these broadly deployed Microsoft applications are out-of-the-box, meaning there are no long, drawn-out, expensive integration projects. Implementation and deployment times also are reduced, and your return on investment improves dramatically.

- Stick with what you know. Or more specifically, what your employees know. Reduce training costs by letting workers continue to use the Microsoft applications they're familiar with, and make training time almost non-existent with intuitive point and click telephony capabilities for calls, Web contacts and other interaction functions.

- Reduce overall costs. By allowing your business to deploy fewer desktop applications while increasing functionality, Interactive Intelligence and Vonexus give your business the advantage of an established technology partnership plus ongoing integration efforts between Microsoft and two companies that bring unmatched innovation, experience and value to your Microsoft platform.

Moreover, Interactive Intelligence and Vonexus let you deploy a proven telephony solution with the backing and cooperation of the largest software company in the world. So mention Microsoft now, and say "phone system" and "voice over IP" right along with it.

Peggy Gritt is Senior Director, Product Marketing for Interactive Intelligence Inc., a global developer of software for contact centers and the enterprise since 1994. Interactive Intelligence integrated out-of-the-box IP functionality into its lineup of business communications software solutions in 2002, and launched its Vonexus subsidiary in 2004 to offer the only Microsoft-based IP PBX available anywhere.

By Hunter Newby

Cutting Through The VoIP Confusion

In this rapidly growing and changing market there still exists much confusion. The absence of a clear picture being communicated by service providers is having an impact on the adoption rate of a basic VoIP implementation model. Understanding what logically comes first could accelerate the development and acceptance of the other services not yet ready for prime time.

In the next few years it will be made clear that there are three major service segments:

- Layer 1: physical layer network interconnection points;
- Layer 2: transport providers (Ethernet and wavelength);
- Layers 3–7: application service providers.

Aside from services there will be hardware and devices. Service providers will be best served themselves if they realize this and focus on their strengths, which probably sit within one of those sections, but not try to do too many. This will only cause them to dilute their message, complicate their branding, and potentially lose one piece of business that they could have won because it was bundled together with another piece that was inferior, or too costly.

VoIP (define - news - alert) is not Voice over the Public Internet. It is Voice over Internet Protocol. VoIP may ride over the public Internet, but it does not have to and in many cases the users (enterprises) specifically do not want it to. The Internet and IP are two different things. Similarly, it is not necessary to have a VoIP phone, or handset to have calls connected using IP whether on the public Internet, or on a private IP network. VoIP does not require a VoIP device at each far end point in order for it to work. Additionally, the hard dollar savings in VoIP is realized from the trunking side (PBX out) and simply IP enabling an existing TDM PBX gives the user access to many wonderful benefits of VoIP related to reducing costs for minutes and other capabilities. This fact seems to elude some journalists and it is evidenced in their reporting... "VoIP (A.K.A. voice on the Internet) is changing the world…" or "...The business case for VoIP is not clear when you factor in all of the costs for the handsets…" These inaccuracies perpetuate misleading information about many important aspects of VoIP, namely security and a CFO hot button: total cost of ownership (TCO).

Rallying around the elimination of toll calls is a sound business strategy.

While this is most likely not intentional and probably due to the speed at which the industry is moving, the lack of investigation and, or comprehension of the way it all works keeps the industry and particularly certain service providers from moving forward at a faster pace because of other things that get lumped together and slow them down. It is easier to understand if we separate the issues and break them up in to smaller pieces. First: the public Internet versus private, purpose-built Internets. Second, VoIP economics — hard dollar savings versus soft savings, or another way to say it, operational expense savings versus functionality and features that improve business process.

Many enterprise IT directors are evaluating VoIP. That in and of itself is telling as the responsibility for voice has shifted to the IT department. For many in IT, voice is a new application, so they read up on the subject and perhaps assume that in order for VoIP to work it needs to be end to end. In most cases they are looking at a combination of equipment including an IP PBX and handsets along with a VoIP service provider. Usually, due to the cost of the handsets, the initial scope of work includes only one or two sites to "see how it goes." For those that know they want to link their offices and run voice over IP between them to eliminate toll calls they all prefer private (non-public Internet) connections. Here we can already see the great business case divide between not wanting to do too many handsets because of expense versus trying to connect every site and run VoIP because of savings. All of this is falling under the VoIP umbrella.

Let's take a closer look at the savings side of the issues. VoIP is generally perceived to be a new type of service that saves money. This is evidenced in the retail side by companies like Vonage that offer flat rate, unlimited calling plans. Basically you pay a port fee and you're in, you are on-net. In the enterprise market this perception is validated by the "shocking" articles being published that say that VoIP isn't "cheaper" when taking in to consideration all of the costs (TCO) associated with it. Well, if the CFO has to factor in throwing out the big fat TDM PBX they just bought three years ago with a five-year payback and then the cost of the new IP PBX and all of the nifty phones that just have to go with it for it to work to its fullest value then, yes, it's probably going to be a while before the ROI happens.

What the enterprise buyer thinks and wants are what the service provider should be saying and selling to them. The question is which service provider can offer them the right solution? It depends on the enterprise and their specific application of course, but for sure, capital expense in the form of a new IP PBX and phones will be examined and delayed, barring some "must-have" features that improve business process

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whereas clear savings realized from IP enabling an existing TDM PBX (define - news - alert) and phones is almost a no-brainer. The service provider segment in the best position to enable the immediate savings are the Layer 2 transport providers working in conjunction with a hardware vendor that IP enables the edge of a legacy PBX and phone system. Rallying around the elimination of toll calls is a sound business strategy because it is what makes the immediate business case for the buyer. This is simple, clean, and the first logical step to any VoIP analysis.

Now let's look at the network side. The enterprise buyers want private Layer 2 VoIP networks for security reasons. They have heard of SPIT (Spam over Internet Telephony), want no part of it, and have the ability to build their own network to avoid it. Once they have made the business case and seen the savings of VoIP over Layer 2 they want to buy data pipes and link their sites. Once the VoIP WAN is built they are in a position to be sold on additional services such as VoIP enabled IXC (long distance "off-net") minutes, public IP transit, and VoIP Peering with other enterprises. The provisioning of these services can be very easily accomplished from a major interconnection point, such as a carrier hotel. This is another sales opportunity for the transport providers. Educate the buyers about what is available to them from these wholesale marketplaces and show them the savings and they will add a node on their WAN at the local carrier hotel.

The enterprise buyers want private Layer 2 VoIP networks for security reasons.

It is at this point that the Layer 2 service providers can begin to refer business to the Layer 3 providers. It is best that they do not try to be all things to the buyers, but rather deliver them to the shopping mall and let them shop.

Hopefully this paints a clearer picture of what's under the VoIP umbrella. It's not that there is anything wrong with IP phones on the desk, but a recent article indicated that IT directors are apprehensive about putting an IP phone on the CEO's desk. This is out of a perceived fear that it won't work and they'll get fired. Maybe it's just that some of the aspects of VoIP need a little more development. That shouldn't keep them from saving money by using VoIP trunking though. Carriers have been doing that for years. Don't let VoIP confuse you. It can be as easy as 1, 2, 3–7. IT

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

By Richard Zimmermann

The Value Of Integrated Conferencing

Many enterprises that proactively deployed IP telephony (IPT) systems are finding that, several months after the implementation, they have not realized the anticipated expense reduction that fueled the ROI analysis that initially justified the deployment of an IPT system. Other companies are looking to aggressively reduce operating expenses to achieve target profitability levels in an increasingly competitive economy. The answer for both camps may lie in integrated conferencing.

Integrated conferencing falls into the broad group of applications classified alternately as real-time communications (RTC) or rich media communications (RMC). It combines traditional and emerging real-time applications such as data, audio, video, and instant messaging (IM) into a single communications vehicle designed to provide an enhanced communications experience and allow geographically disparate business units to collaborate more efficiently.

The emerging market for these components represents a significant opportunity for manufacturers, value added resellers, and service providers. Revenue in this market space, although difficult to measure accurately, is conservatively estimated at $4.5 billion. The lion's share of this revenue is in traditional audio-conferencing solutions, followed by videoconferencing ($600 million), Web conferencing ($500 million) and instant messaging ($300 million). According to Mike Gotta (*Real Time Collaboration: Making the Enterprise Decision, Content & Collaboration Strategies*), this market is expected to grow to over $10 billion by 2008.

RMC solutions fall into three categories, each with its own risks and benefits: externally-hosted solutions, on-premises solutions, and managed service solutions.

Externally-hosted solutions generally make sense for organizations with sporadic needs for event-type conferences with large numbers of participants (e.g., quarterly all-employee conference calls). In this type of environment, it is not economical for an organization to buy and maintain a premises-based system that will only be used at full capacity once a quarter. However, there are security, compliance, and privacy risks associated with hosting IM, audio, and data traffic over the public network. In addition, actual monthly costs for hosted solutions may be higher than expected, due either to one-time setup and overage charges, or to the fact that many departments establish their own contracts with service providers, which erodes the economies of scale across a single, enterprise-wide agreement.

Premises-based solutions are typically more economical for organizations with high-volume conferencing requirements across geographically dispersed locations. A clear advantage of premises-based systems is that the enterprise can place all the facets of a conferencing solution behind its corporate firewall, and apply its own security policies, allowing off-net users access through the firewall. Although premises-based solutions require an upfront capital investment, many companies have been able to finance the upfront purchase and actually lower their monthly operating expenses by reducing or eliminating monthly payments to service providers of externally-hosted solutions.

The ROI is often less than twelve months for companies that shift from an externally-hosted to a premises-based solution, calculated purely based on hard costs (i.e., the costs of deploying the new system minus the costs avoided by canceling contracts for hosted solutions). When soft costs, such as savings from travel avoidance, are considered, the savings become even more compelling. But ironically, many companies find that they are not realizing the expected ROI after the implementation of a premises-based solution. In many cases, this is not a technological issue, but a social one. That is, employees are slow to embrace the new system, or the company fails to change its business policies to efficiently exploit the advantage of the new system. For instance, the contracts with the former hosted service provider are not cancelled, travel policies are not adjusted to encourage integrated conferencing as an alternative to face to face meetings, or adequate end user training is not made available to employees to allow them to realize all the benefits of the new system.

Once an RMC system is deployed within an enterprise and embraced by its employees, the benefits gained through increased employee productivity can far outweigh the expense reductions that justified the initial system deployment. The benefits of horizontal applications are obvious: many more people from diverse backgrounds can participate in continuing education, open forums, and other meetings, giving companies the benefit of much more diverse input on critical issues. However, vertical-specific applications may hold the greatest potential for improving business efficiencies. For instance, real-time document sharing during contract negotiation can eliminate multiple rounds of passing revised documents back and forth between the parties. Similarly, advances in virtual whiteboard technology can allow system engineers and clients to jointly design proposed solutions in real time, cutting down on both the time and expense associated with multiple face to face meetings.

RMC is unlikely to replace face-to-face meetings in their entirety. For the establishment of new business relationships, it is difficult to replicate the interpersonal and non-verbal communication that is essential to establishing a level of comfort and trust with prospective business partners. However, for organizations savvy enough to judiciously blend face-to-face meetings and integrated conferencing as appropriate, RMC holds intriguing possibilities for dramatically increasing productivity and lowering cost. IT

Richard Zimmerman is Forsythe's director of network solutions marketing. Zimmermann also serves as the Chairman of the Board of the Enterprise Communications Association (ECA), an industry forum promoting the deployment of voice, video, and data communications solutions in the enterprise. For more information, please visit http://www.encomm.org.

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Toshiba Makes A Splash At Niagara Water

Water. It's the essence of life. More than half of the human body is water, and it makes up three-quarters of our planet. We depend on it for our very existence, as do the thousands of customers who purchase Niagara Water for their everyday water needs. One of the leading bottled drinking water companies in Southern California, Irvine-based Niagara Bottling LLC turned to Authorized Toshiba Dealer **smpl**solutions of Lake Forest, California, for a reliable, technically advanced telephone system that is as essential to Niagara's business as water is to the human body.

Mission: Network and Centralize the Telephone System, Add VoIP

Niagara (news - alert) needed to network its three facilities together — its headquarters and call center agents in Irvine and its bottling plant and main offices in Ontario, California — while allowing for future growth. In addition, Niagara's management wanted to maximize the investment in its existing Toshiba Strata DK280 telephone system by improving internal and external business communications.

Solution: One System for Three Offices, Remote Workers

David Case, president of **smpl**solutions (news - alert), met Niagara's needs using a single Toshiba Strata CTX670 business communications system plus Toshiba IP (Internet Protocol) telephones at the company's other locations. Case explained, "We replaced two Toshiba telephone switches with one switch at Niagara's Ontario facility. Then we added ACD (Automated Call Distribution) for customer service call-ins at the Ontario system and routed the calls to the Irvine call center agents. By putting in IP telephones at the Irvine office, where call center agents are located, ACD calls are actually done over the Internet."

To make this work without changing any of the telephone numbers that Niagara's customers were accustomed to dialing, Niagara had the phone company direct the Irvine telephone lines to ring at the Ontario facility. As a result, when customers dial the Irvine office telephone number, the call is routed to Ontario and goes into the auto-attendant for customer service. It is then automatically placed into a queue for the next available CSR (Customer

David Case of **smpl**solutions, right, shows Andrew Wirtjes of Niagara how to program his Toshiba telephone.

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Moving to IP enabled the elimination of telephone lines and allowed Niagara to consolidate all its bills.

Service Representative) who resides back at the Irvine office. When a CSR becomes available, the call is routed over the Internet back to the Irvine Office.

Moving to IP telephones at the Irvine facility enabled the elimination of 80 individual telephone lines and allowed Niagara to consolidate all its bills, resulting in a savings of $5,000 to $6,000 per month. "This allowed us to negotiate a pre-set amount for our outgoing calls," Andrew Wirtjes, director of Technology & Information of Niagara, said. "We never again have to wonder what our phone bill will be!"

Result #1: Remote Workers Plug In Anywhere

Of Niagara's 300 employees, only about 100 of them are on-site. Many of the others, particularly in sales and tech support, work remotely from their homes, using IP telephones.

In fact, Niagara's owner Andy Peykoff, recently named Entrepreneur of the Year by Ernst & Young, is one of the company's remote workers. He uses his Toshiba IP telephone to stay connected with customers and employees. Working predominantly from home, he can even contact employees at the office by dialing their four-digit extensions. Eliminating long-distance costs between his local residence and out-of-state residence has saved the company nearly $500 per month.

Wirtjes also uses an IP telephone at his home office or at hotels when he travels. "People just don't realize how easy it is for me to take my IP telephone and plug it in wherever I am, and it's just like getting and making calls at the office."

Result #2: Remote Management With Complete Control

Wirtjes remotely manages the system directly from his computer. He said, "I can control everything right from my laptop without having to travel to the office." With the IP telephones, moves, adds, and changes are easy and don't require the dealer's assistance or even for Wirtjes to go out to the facility. He explained, "To move a phone, it's just as easy as this: unplug the IP telephone, move it, and plug it in. The system automatically finds it and logs it in."

Wirtjes also appreciates the ACD system's extensive reporting capabilities. He said, "We use it to track productivity and usage and to make plans to improve efficiencies as our customer base and employee needs grow."

Result #3: Toshiba Delivers on Migration Path

In choosing the new Toshiba Strata CTX670 system, Wirtjes said Toshiba's migration path was a strong motivation. "Toshiba's strong migration path meant that we could meet our goals without having to replace all of our existing Toshiba equipment."

By upgrading from their existing Toshiba system, Niagara saved at least a third of the hard equipment costs over buying a completely new Toshiba system, according to Wirtjes. He added that a bid from a well-known IP-only equipment supplier was more than double the cost of migrating the Toshiba system.

As a result of the Toshiba System upgrade, Niagara has also decreased its overall calling costs. Case said, "We have eliminated long-distance calls between the Irvine and Ontario facilities, saving an average of $1,350 per month for the client, making calls between Niagara's facilities virtually free. In addition, we improved Niagara's outgoing long-distance charges by having long-distance calls originate solely from Ontario."

The Bottom Line: A Centralized, Network System That Saves Thousands

"Together, Toshiba and **smpl**solutions met our goal of having a centralized telecommunications system that would work at all three locations and appear completely seamless to callers," Niagara's Wirtjes said. "It also allowed us to retain our investment in our original Toshiba system and delivers cost savings of nearly $10,000 every month. You could say that Toshiba helps us manage our flood of calls and saturate the market with Niagara Water." **IT**

Andrew Wirtjes of Niagara, left, and David Case of **smpl**solutions congratulate themselves on installing a Toshiba system that helps save Niagara more than $10,000 per month.

E911 Ruling: What It All Means

By Yaron Raps, Bruno Codispoti & John Pfaff

The Federal Communications Commission recently took steps to protect consumers by mandating that all providers of Voice over Internet Protocol (VoIP) phone service supply Enhanced 911 (E911) emergency calling capabilities as part of their standard service. The order has the intent of minimizing the likelihood of situations like recent incidents in which users of VoIP dialed 911, but could not connect with 911 operators. The May 19, 2005 order, which requires compliance within 120 days of the published date, marks an important step for maturing VoIP technology as a means of delivering voice calls — but one that is not without its challenges.

Basic 911 Versus E911

In fact there are major differences between basic 911 services and E911. With basic 911, the address information of the calling party is not transferred to the public safety answering point (PSAP) or local emergency centers; the calling party has to inform the operator of his/her location, requiring, therefore, that the customer be conscious and in a condition to speak. Also, basic 911 calls are routed to the PSAP as regular calls via the public switched telephone network (PSTN) (define - news - alert) without priority status and may not be handled in the same manner as calls made via the E911 network.

By contrast, with E911, a customer's physical address is stored in a database known as automatic location identification (ALI). During an emergency call, the customer's service provider routes the call to a dedicated network that has been built to address E911 call needs from a reliability, capacity and technology standpoint. The calling party's phone line generates an automatic number identification (ANI) signal to the network. The E911 system then reads the ANI and ALI from databases and routes the number to the appropriate PSAP emergency center. Along with that, the caller's address information is displayed on the dispatch screen, enabling the dispatch of emergency personnel to the site of the emergency. But when companies began offering VoIP (define - news - alert) , some service providers could not provide E911 services like this because of technical issues and / or business constraints; moreover, some providers would not even attempt to route or complete 911 calls if the customer did not initialize the 911 service.

However, differences in 911 service and its limitations notwithstanding, most customers choose VoIP because of more attractive pricing. Generally, customers have neither knowledge of nor reason to understand the differences between E911 and basic 911 in terms of operation; customers act on the expectation that the new technology has the same capabilities as the existing technology — and that this voice product works the same as their traditional telephone service.

The FCC order has the effect of imposing E911-like rules on VOIP providers in the near future and outlines the following requirements:

• VoIP providers must deliver all 911 calls to the customer's local emergency operator. This must be a standard, rather than an optional feature of the service. This means that it will not be up to the customer's discretion to activate the service; all customers, new and existing, must have this capability.

• VoIP providers must provide emergency operators with the call back number and location information of their customers (e.g., E911) where the emergency operator is capable of receiving it. Although the customer must provide the location information, the VoIP provider must pass to the PSAP the emergency physical address, and also afford the customer a means to update this information, whether he or she is at home/office or away in a different location.

• By the effective date, VoIP providers must inform their customers, both new and existing, of the 911 capabilities and limitations of their service.

• The four major incumbent local exchange carriers, (ILECs) must provide access to the network to any requesting service provider. This includes access for competitive carriers to trunks, selective routers, and E911 databases.

Impact On VoIP Providers

Today VoIP service providers fall into three categories: cable providers such as Cox Communications and Time Warner Cable, wireline such as AT&T and Verizon, and emerging or pure play VoIP providers such as Vonage and Packet8. Some will not need to make modifications to their infrastructures to support the new E911 ruling. Others will face challenges not only related to supporting E911 within the given time, but also to comply with the recent order at the same time that they strive to keep their networks and cost structures intact.

Cable Service Providers: The vast majority of cable providers are already supporting E911. Cable VoIP service is delivered through a customer premise device that is associated with an IP address and a customer's static physical address, and cable providers use this address as the emergency location. Also, the cable providers are either a licensed competitive local exchange carrier (CLEC) or partner with a CLEC that has direct access to the 911 network. Further, cable companies typically do not offer customers multiple phone numbers with area codes that differ from their physical addresses, or authorize the use of the VoIP service at locations other than the registered service address. As a result, most, if not all, cable companies will require little to no action to comply with the FCC's ruling.

Wireline Service Providers: Wireline companies have the infrastructure that provides connection and support to the 911 networks. However, in order to stay competitive, most wireline companies are supporting virtual numbers and mobility. For example, customers can choose a New York City number while living in Dallas today and tomorrow they can physically install their VoIP "line" in Chicago. Because the customers are responsible for activating their 911 service, the operator and the authorities would not automatically have knowledge of the customer's relocation. The wireline companies' challenge is to enable these new 911 capabilities without being disruptive to customers. Further, to comply with the FCC order, they may also be required to change their sign-up and provision processes if they support multiple virtual numbers.

Emerging/Pure Play Service Providers: The FCC order may have the greatest impact to this segment. The FCC mandate now requires these service providers to connect to 911 networks via their incumbent local exchange carrier (ILEC). Up to this point, the FCC did not mandate the ILEC to accommodate an emerging/pure play's request to utilize the E911infrastructue. While this requirement will provide these companies with the enabling advantage to afford their customers E911 capability, it comes with the burden of significant additional operational cost. Put simply, some emerging service providers must start from scratch to connect to the network. They must provision and install dedicated 911 circuits. Also, providers will have to wrestle with E911 costs – both operational and 911 surcharges, passing these through to the customer or absorbing the costs depending on their strategy. These additional costs if passed to the customer, potentially make it more difficult to attract and retain customers and if absorbed, further challenge the provider's profitability.

A Value Add, But Questions Remain

If VoIP is to reach the mass market, it must perform like plain old telephone service and the FCC ruling goes a long way to accomplishing this and establishing VoIP as a mainstream phone service. This is indeed a plus for the cable and wireline VoIP providers who may begin to see a more level playing field in terms of price competition from the emerging or pure play VoIP providers. Similarly, these service providers will, in turn, realize some beneficial effect in their attempt to gain equal footing with these other multiservice communications companies and in their ability to gain access to not only the E911 network but also to the trunks, selective routers, and E911 databases that are required to provide the enhanced emergency service.

But how will the ruling impact the wireline offering of virtual number, mobility, and the overall industry focus on a person not a place? And, can the pure play VoIP providers add the E911 service and absorb the additional infrastructure cost and still offer new customer's bundled packages at a lower cost — let alone deal with the price commitments and terms of their existing contracts? Will theses emerging/pure play providers absorb the additional infrastructure and regulatory costs, or will they be forced to pass along the cost to customers in the form of rate increases — a contrast to their initial marketing, which was all about price for features offered?

The FCC's E911 ruling was not a shot across the bow of VoIP providers. Headlines in the mainstream press took that shot. But one thing is certain: VoIP will not become mainstream until it can solve the E911 issue. In that process, there may be some winners and some losers. Only time will tell. But with the clock ticking, VoIP takes another step towards becoming a mainstream telephone service. IT

Yaron Raps and Bruno Codispoti are solution partners, and John Pfaff, a senior consultant, in the communications practice of BusinessEdge Solutions, Inc., an industry-focused business and technology consulting firm offering operational strategy, business process and systems integration solutions to clients in the communications, life sciences, financial services and insurance industries. For more information, please visit http://www.businessedge.com.

Addressing Security In VoIP

By Greg Galitzine

Security is the Achilles' heel of IP networks. At least, that is the perception held by many people, in and outside of the industry alike. Indeed, reports abound of security breaches and the threat of a coming wave of security disruptions targeting VoIP networks and VoIP (define -news - alert) users has security experts scrambling to meet the danger head on.

For example, Broadband Reports posted news in 2004 regarding Cleveland-based Broadvox Direct, and a potential security breach there that could have resulted in the theft of unsecured customer data.

And, a ZDnet report from last month underscores the fact that VoIP (among other technologies) must be hitting the mainstream, "when it becomes the focus of pharming and other security attacks."

(According to Wikipedia.org, Pharming is defined as the exploitation of a vulnerability in the DNS server software that allows a hacker to acquire the Domain Name for a site, and to redirect traffic meant for that Web site to another Web site. DNS servers are the machines responsible for resolving Internet names into their real addresses — the "signposts" of the Internet.)

The report goes on to explain: "One of VoIP's flaws is that it is inherently vulnerable to hackers because, like e-mail, VoIP calls find their way by locating an IP address, a unique set of numbers assigned to each device connected to the Web. Yet while scores of commercial VoIP providers have quickly expanded to take advantage of the growing interest in the service, many have not implemented even basic security measures, such as encrypting phone calls."

And with consumer VoIP subscriptions on the rise, expecting to reach up to 25 million over the next few years, it is truly fertile ground for hackers and other ne'er do wells looking to wreak havoc.

There are a number of organizations that are looking to address the issue of VoIP security. Perhaps best known is the group called VoIPSA (for Voice Over IP Security Alliance). The Alliance was conceived to fill the void of VoIP security related resources through a unique collaboration of VoIP and Information Security vendors, providers, and thought leaders. VOIPSA's mission is to promote the current state of VoIP security research, VoIP security education and awareness, and free VoIP testing methodologies and tools.

According to a news release announcing the organization earlier this year, "The growing convergence of voice and data networks only serves to exacerbate and magnify the security risks of today's traditional prevalent cyber attacks. Successful attacks against a combined voice and data network can cripple an enterprise, halt communications required for productivity, and result in irate customers and lost revenue. As VoIP increases in popularity, so does the potential for harm from a cyber attack. As VoIP deployments become more widespread, the technology becomes a more attractive target for hackers. The emergence of VoIP application-level attacks will likely occur as attackers become more familiar with the technology through exposure and easy access."

According to a recent article in *Internet Telephony* magazine, authored by Joel Pogar, National Practice Manager Secure Network Services at Siemens ICN, there are a number of specific threats and best practices to consider when addressing VoIP security in the enterprise.

Threats

Among the more significant concerns relating to VoIP are:

- **Denial of Service (DoS) Attacks:** IP phones, and VoIP gateways can be bombarded with malicious packets in an attempt to disrupt communications.
- **Call Intercept:** Unauthorized monitoring of voice packets.
- **Signal Protocol Tampering:** Malicious users could monitor and capture the packets that set up the call. This would allow them to manipulate fields in the data stream and make VoIP calls without using a VoIP phone. This could be especially problematic, when used to make expensive calls and make the IP-PBX believe it originated from another user.
- **Presence Theft:** Impersonation of a legitimate user sending or receiving data.
- **Toll Fraud:** The ability of a malicious user or intruder to place fraudulent calls.
- **Call Handling OS:** The call handling software of many IP-PBX systems relies on operating systems, or operating system components, that may not be secure. Once compromised, this could be an avenue into other connected systems and information stores.

If VoIP traffic will be traversing a firewall, make sure your firewall is capable of direct support for SIP or H.323.

Best Practices

To minimize the security risks in a VoIP environment, the following best practices are recommended:

Virtual LANs

Keeping voice and data on separate VLANs is a good idea for increasing performance and security. What's more, the best practice for securing a voice VLAN is to control the traffic between the voice and data VLAN using filtering and/or firewalls. This can prevent DoS attacks and spoofing as well as providing general filtering that limits malicious footprinting.

Encryption

Wherever possible and practical, implement encryption through VPNs (define - news - alert) or any method available to you. On one hand, encryption potentially can delay voice packets and adversely affect the performance of VoIP on your network — especially with multiple encryption points. On the other hand, if a network is operating efficiently, the overhead of the encryption should have little impact the performance of the VoIP system.

Direct Firewall Support

If VoIP traffic will be traversing a firewall, make sure your firewall is capable of direct support for SIP (define - news - alert) or H.323. If you have to "open" a port to allow these protocols through, then your firewall does not adequately support VoIP.

As VoIP increases in popularity, so does the potential for harm from a cyber attack.

Secure OS Of Call Handling Software

Use a commercial scanning tool to "probe" the call servers in your VoIP system. If any critical or high-level vulnerability arises, contact your vendor to have them corrected as soon as possible. Care should be taken to allow only necessary services to run and to limit the number of listening ports that could be attacked. This might warrant placing core VoIP devices in a "safe zone" behind a firewall or a router with access filters.

Routine Monitoring

Managed services are a good idea for firms without the resources to keep an eye on their networks. It also makes sense when your VoIP system becomes mission critical. You should establish daily, weekly and quarterly milestones of activity to watch for.

Sound Security Practices

If already in place, a good data security program — strong passwords, anti-virus protection, reliable backup and so forth — gives firms that much of an advantage when implementing VoIP and should be maintained rigorously at all times thereafter.

Greg Galitzine is the editorial director for Internet Telephony magazine.

Putting Security Concerns at Ease

Spurred by the promise of greater productivity and efficiency offered by innovative applications that improve organizational processes and significantly lower costs, organizations of all sizes are quickly adopting voice over IP (VoIP) technologies. In fact, Gartner predicts that by 2009, 90 percent of all new telephony systems sold will be IP enabled. However, while CIOs and network administrators appreciate the benefits of VoIP, they also have concerns that their organization is implementing a solution that may be vulnerable to a range of security threats.

As many security experts have expected, the increase in VoIP adoption has caught the interest of hackers looking for a new challenge. VoIP involves the digitizing of voice communications, converting the digital signal into data packets and then transporting the information across an IP network, including the public Internet. Since VoIP delivers voice over the same lines as data, phone calls are now threatened by the same security challenges as data including viruses, spoofing, eavesdropping, and denial of service (DOS) attacks.

For example, Internet hackers with the capability to intercept voice packets could remove, edit, or add parts of a conversation unbeknownst to the sender. Even simple eavesdropping, which is a much more likely threat, carries the potential to cause significant damage.

In industries such as healthcare and financial services, where organizations face strict security mandates, the ramifications of a security breach are especially severe. Companies in closely regulated industries are expected to protect and monitor their systems at all times and face very strict financial and legal penalties if their systems are compromised.

Fortunately, strategies to address security issues for data networks translate very well to VoIP. By employing a series of comprehensive measures to ensure that both voice and data information is secure, organizations can minimize the risk from myriad security challenges. An effective, holistic approach to security requires constructing initiatives, protocols, and advanced monitoring technologies designed to vigilantly protect an organization's internal network infrastructure, the perimeter and the transportation method.

The first step to securing an organization's voice and data network is conducting a rigorous security audit and network assessment. A thorough security audit designed to find vulnerabilities in an existing network ensures that external forces will not compromise an organization's network.

Next steps to secure the network include implementing stringent monitoring and prevention tools and services. Existing firewalls should be reviewed and strengthened, if need be and organizations should also consider implementing intrusion detection and prevention systems as well as network monitoring services.

Companies must also consider how its employees will be efficiently authenticated when trying to access secure areas of the network. Increasingly, organizations are seeing the benefits of biometric authentication such as retinal, fingerprint, and voiceprint scans to provide a stronger defense against unauthorized access than more traditional methods such as passwords or digital certificates. While upfront costs of biometric devices are higher, the added benefits plus long-term cost-savings from reduced maintenance provides CIOs and network administrators with a compelling reason to invest.

In addition to these steps, organizations should also evaluate other tools that are available to provide additional layers of protection. While no solution can absolutely guarantee safety, a well-planned and implemented security strategy helps to ensure the reliability of the VoIP network by significantly decreasing the risk from threats such as viruses and unauthorized attacks.

With a secured network, organizations are able to enjoy the benefits offered by VoIP to concentrate on achieving their fundamental business objectives.

To learn more about NEC's VoIP Security solutions, please visit http://www.necunified.com.
Mark Nagiel is Manager of Security Consulting with NEC Unified Solutions, Inc.

VoIP Developer Conference Set For Triumphant Return

Still reveling in the afterglow of 2004's successful debut, the VoIP Developer Conference is gearing up for a repeat performance this August 2–4 in San Francisco.

TMC VoIP DEVELOPER CONFERENCE™

VoIP (define - news - alert) is one of the hottest sectors in technology today; it is changing the way the world communicates, and the world is taking note. The second annual VoIP Developer Conference — taking place at the South San Francisco Conference Center — is the only conference in the United States where developers can gather to meet and network and to learn how to quickly develop new VoIP applications that are in high demand.

Delegates to the VoIP Developer Conference can expect the usual high-quality conference proceedings that TMC has become famous for. The advantages of attending this conference are many:

• Learn how best to leverage the available development tools on the market.

• Discover what you need to know about today's hottest technologies: SIP (define - news - alert), WiFi telephony (define - news - alert), DSP (define - news - alert)development, Presence, Peer-to-Peer, Linux telephony, and more.

• See hands-on demos and compare products from industry leaders.

• Take advantage of unmatched networking opportunities with your peers in the VoIP development community.

• Acquire the knowledge needed to develop new applications for Wireless/ WiFi telephony, perhaps the most promising aspect of the VoIP explosion.

• Acquire Tools for Development in the Entire Spectrum of VoIP Products: video over IP, SIP, cable telephony, wireless/WiFi telephony, software development and hardware/device development.

Keynotes

The show has already established a tradition of presenting a fantastic lineup of keynote speakers, and this year is no different. The development community will hear expert commentary and insightful thoughts from industry leaders AudioCodes (news -alert), Avaya (quote - news -alert), and Aculab(news - alert).

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> "Every session I attended was interesting and valuable."
>
> **– Rick Ringel, Inter-Tel**

Alan Percy

Mr. Percy is Director of Business Development at AudioCodes, a leading provider of Voice over IP telephony enabling technology and systems components. Mr. Percy joined AudioCodes in July 2001, bringing over 15 years of experience in the telecommunications, networking and wireless equipment industries. Mr. Percy is a well respected and frequent speaker at industry conferences.

Scott McKechnie

Scott McKechnie is Director of the Application Enablement Services in the Converged Systems Division at Avaya. He is responsible for leading the company's efforts in developing a powerful new Application Platform for third-party ISVs (Independent Software Vendors), SI (System Integrators), and Corporate Application Developers to create the next generation of converged telephony and business solutions for our Avaya customers. McKechnie has been in the communication industry for over 23 years. He has extensive experience in the digital PBX technology, software development best practices and leading small rapid response development teams.

Mike Donoghue

Mike Donoghue, Aculab's Vice President of Sales, is responsible for leading the growth of Aculab's sales and customer support activities in the Americas Region. Mike brings over 18 years of experience to Aculab, growing both established and emerging companies in the communications industry. His initial experience was gained at Brooktrout Technology where his responsibilities over a 13-year period included National Sales Manager, Managing Director, Europe, Vice President of Worldwide Sales, and General Manager. Prior to joining Aculab, Mike was the Vice President of Sales & Business Development for iMPath Networks, where he also served as interim CEO.

Conference Program

Rick Ringel, director of engineering at Inter-Tel, said of last year's conference, "Every session I attended was interesting and valuable." Once again the editors of Internet Telephony magazine have worked hard to create a conference program that is at once unique and second to none. Broken into five tracks spanning three days, the conference will cover the hottest topics in the VoIP development space, from hardware and software development, to SIP and wireless development trends, and other essential issues.

Here's a sample of just some of the cutting-edge session titles that will be presented at the conference this August:

- Voice Quality — Challenges Facing VoIP Carriers
- IMS & Wireline to Wireless Convergence
- Real-Time Performance and Voice Quality on Single Core RISC Devices
- Utilizing SoC for Building Carrier-Grade Fault-Tolerant Hardware
- How to Construct World-Class VoIP Applications on Next Generation Hardware
- Designing Applications Using DSPs
- Developing a Next-Generation Roadmap for VoIP Migration
- Developing Applications Using Host Processing Instead Of DSPs
- Developing SIP Compliant VoIP Solutions for Mobile
- Voice over WiMAX Development Trends
- Voice Over WLAN and IP Phone System Architecture

Travel

The conference is taking place at the South San Francisco Conference Center, which is centrally located just one mile north of San Francisco International Airport, 15 minutes from downtown San Francisco, and just a short jaunt from the Silicon Valley. The proximity of this event to the vast population of developers of all stripes bodes well for the opportunity to meet and network with the best and brightest that the industry has to offer. The official show hotel is the Four Points by Sheraton Hotel & Suites San Francisco Airport. For more information, and to book rooms, please call the hotel directly at 650-624-3700. IT

CONFERENCE SESSION HOURS

Tuesday, August 2
12:00pm–6:00pm
Wednesday, August 3
8:30am–6:00pm
Thursday, August 4
8:30am–5:15pm

EXHIBIT HALL HOURS

Wednesday, August 3
6:00pm–8:00pm
Thursday, August 4
10:00am–4:00pm

EXHIBITOR LIST
(as of May 31, 2005)

- Aculab
- Avaya DeveloperConnection Program
- Brooktrout Technology
- CT Labs
- Data Connection
- Electric Cloud, Inc.
- Excel Switching
- Intel Corp.
- LignUp Corporation
- LSI Logic
- Metreos Corporation
- NMS Communications
- PIKA Technologies Inc.
- Sangoma
- SIPFoundry, Inc.
- Surf Communications
- Telco Bridges, Inc.
- Testing Technologies
- Trinity Convergence
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Fax: 516-829-2923
Web http://www.phnxaudio.com

Price: $299.95

Whether it's using your desktop phone's speakerphone mode or a professional conferencing unit sitting in the middle of a conference room table, just about everyone at one point has used some sort of speakerphone to hold a conference call. But what about VoIP conferencing? Sure, if you own a Cisco 7900 series VoIP phone (~$200–$300) you can simply use the speakerphone on the Cisco unit itself. However, most desktop phones — including desktop VoIP phones — do not have the highest quality speakerphone nor very good echo cancellation. Further, what if you are part of the growing trend of VoIP softphone client users? If you use a VoIP softphone client such as Skype, Xten SIP client, or the Vonage soft client, how do you engage in a VoIP conference then?

Phoenix Audio Technologies Duet product "speakerphone-enables" any PC-based audio application including any VoIP softphone application. The Duet works via USB and can not only act as your VoIP (define - news-alert)speakerphone, it can also work with your telephone, IP telephone, cell phone, or PDA connecting via the your device's headset jack to the Duet's RJ11 (telephone connection) , thus adding speakerphone capabilities to these devices as well.

In addition, the Duet can work with both your computer and your telephone simultaneously, thus enabling three-way conferencing. For instance, we were able to start a Skype call (computer), then dial a PSTN (define - news-alert) number (using your telephone) and then conference the two — essentially "bridging" a VoIP connection with a PSTN connection — our own little mini VoIP gateway if you will. This VoIP-to-PSTN conferencing feature is a unique feature for the Duet that no doubt devoted VoIP users (such as Skype users) will enjoy.

Simply plug this device into your USB port and Windows automatically installs the proper USB audio classes and drivers to be able to communicate with it. It is powered via the USB port and a cool blue LED lights up indicating power. Importantly, it can also be powered via a power connector for when the USB connection is unavailable while using it exclusively with your telephone or cell phone. Also worthy of noting is that when muting the device via the mute button, the blue LED flashes. The unit also features two other buttons — Volume Up and Volume Down.

By default, the Duet utilizes its internal microphone and speakerphone for high-quality, hands-free speakerphone communication. You can override this default behavior simply by plugging your headphones or high quality desktop speakers into the 3.5mm speakers/headphones jack. This will only disable the internal Duet speaker — but still use the built-in microphone.

The Duet also has a 2.5mm headset jack for connecting a standard telephone headset. By connecting the telephone headset to your Duet you automatically override both the Duet's internal speaker and microphone.

Although the main purpose of the Duet is for VoIP conferencing, it can simply be used as a desktop speaker replacement. Thus, you can play your favorite iTunes or MP3 files with no problem. Although, the speaker seems to be designed more for voice than music since it seems a bit on the tinny side with minimal bass. The conference quality is the result of Phoenix Audio's proprietary technologies which include acoustic echo cancellation, noise suppression, Line Echo, and voice level compensation algorithms. Overall, the sound quality of the built-in speaker was pretty good, especially for voice online, and the microphone sensitivity was just about right. The echo cancellation seemed to work very well, although if we jostled or moved the unit we did get a tiny bit of feedback. We asked Phoenix about this and they told us that this was only a problem in the pre-production units and that it was corrected in their software and works just fine in their production units.

Conclusion

Simply put, using the Duet enables more than one person to engage in a VoIP call or online conference headset-free using the Duet's high-quality microphone and speakerphone. It's the first product of its kind that we've seen that brings audio conferencing to VoIP softphone clients and we enjoyed its ease of installation, portability (fits in a laptop case), and very good sound quality. IT

RATINGS (0–5)

Installation: 5
Documentation: 5
Features: 4.5
GUI: N/A
Overall: A-

PROS	CONS
Conference both VoIP and PSTN calls Speakerphone-enable VoIP softphones Excellent echo cancellation Plug and play installation	A bit pricey

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TMC Labs Internet Telephony Innovation Awards 2005: Part I

VoIP (define - news- alert) continues to dominate technology news as one of the most innovative technologies. TMC Labs wishes to recognize the companies that are providing innovative solutions within the Internet telephony industry. This is our sixth installment of the TMC Labs Innovation Awards and we should point out that we grant these awards based solely upon how unique or innovative a particular product or service is and not based upon sales numbers or how large a company is.

Once again, our task in picking the most innovative products and services was quite challenging. It seems that each year brings more and more innovative VoIP products and services. This year we had quite a few VoIP testing and QoS products that perform tasks such as measuring bandwidth utilization, maintaining QoS, measuring voice quality, and performing VoIP load testing. Well, it certainly was difficult to judge which products within this same genre were more innovative since they all did something similar but in a unique and different way.

TMC Labs proudly bestows 25 Innovation Awards, which will be published in two parts in order to accommodate our in-depth write-ups for the winners. The complete winners list will be published in both issues, however we will write the detailed write-ups in alphabetical order beginning with AnchorPoint this month and ending with Go2Call. Next month, we start

2005 TMC Labs Innovation Award Winners (Full List)

COMPANY	PRODUCT NAME
AnchorPoint	AnchorPoint Business Analytics
AudioCodes Ltd.	TP-260 SIP Gateway
Brooktrout Technology	SnowShore IP Media Server
CallWave, Inc.	CallWave Mobile Call Screening/Mobile Call Transfer
Covad Communications	COVAD VoIP with Voice-Optimized Access (VOA)
EagleACD	EagleACD
Edgewater Networks	EdgeView
Empirix Inc.	Hammer VoIP Test Solution for Enterprises
FiberTower Corporation	Backhaul Services for Wireless Carriers
FrontRange Solutions	IP Contact Center (IPCC)
Global IP Sound	VoiceEngine Mobile
Go2Call	Hosted VoIP Global Platform
Inter-Tel, Incorporated	*Inter-Tel 5000 Network Communications Solutions*
LignUp Corporation	*LignUp Communications Solution*
Millenigence, Inc.	*DashPhone CXP*
Nortel	*BCM50*
Pandora Networks	*Worksmart*
RingCentral	*RingCentral*
SER Solutions, Inc.	*TSP500 Outbound Dialer*
Toshiba America Information Systems, Digital Solutions Division	*Toshiba Strata CIX*
Tripp Lite	*SmartOnline Expandable Rack/Tower UPS System*
VegaStream	*Vega 400*
Witness Systems	*eQuality ContactStore for IP Solution*
Xten Networks, Inc	*Xten Pocket PC SIP Softphone*
Zultys Technologies	*MX30 Enterprise Media Exchange*

**companies appearing in italics will appear in our August 2005 issue with a full description.*

with Inter-Tel and end with Zultys.

AnchorPoint

AnchorPoint Business Analytics

http://www.anchorpoint.com

Telecom managers who audit bills, make purchasing decisions, manage VoIP, PBXs(define - news- alert), and wireless usage, as well as negotiate vendor contracts have a difficult task, especially when they are managing Fortune 1000 companies that can have multiple locations with multiple PBXs that can reach in the dozens or even hundreds. Equally, business unit executives who are accountable for the costs consumed by their departments and CFOs who need data and trend analysis for budgeting, forecasting and cost-saving initiatives; as well as CIOs who need to contain IT budgets and align their costs with business objectives and need real time access to monitor budgets and costs also have a difficult task.

AnchorPoint (news - alert) Business Analytics aims to simplify these tasks with their set of business intelligence tools designed to help companies mitigate risk and increase the efficiency and agility of their communications management initiatives. When implemented with AnchorPoint's suite of Telecom Financial Management (TFM) applications, Business Analytics provides real-time communications data to end-users through its dashboard, benchmarking, and ad-hoc reporting tools. Integration with the AnchorPoint TFM application suite (including the Invoice Management, Asset Management, and Usage Management modules) is made possible through the AnchorPoint Communications Intelligence Engine. Data from each module is collected in a centralized database and processed by the AnchorPoint Communications Intelligence engine. Information is then delivered to the user via the Business Analytics reporting tools to allow companies to re-allocate or eliminate costs, improve operating efficiencies, and increase ROI from their telecom/IT investments. With Business Analytics, users can view information from a single screen in multiple methods — charts, reports, meters, dials, etc. — that represent data from the AnchorPoint Communications Intelligence engine in a "business ready" format.

To develop Business Analytics, AnchorPoint partnered with the well known business intelligence technology market leader, Business Objects. This partnership, unique in the industry, allows AnchorPoint customers to benefit from Business Objects' leading reporting and analytic tools.

AnchorPoint's Business Analytics and TFM solutions help companies meet five key business objectives: minimize communications expenses; maximize ROI on communications assets; improve operational efficiencies; enable internal staff to focus on core competencies; and provide a baseline of information to make decisions on new technology purchases (such as VoIP).

AnchorPoint targets companies who have annual telecom expenses starting at $2M and up. Currently, 20 percent of AnchorPoint's customer base is in the Fortune 500 and 80 percent is in the Fortune 2000.

AudioCodes Ltd.

TP-260 SIP Gateway

http://www.audiocodes.com

Every once in a while, TMC Labs hears about a product or reads a product description and we say to ourselves "Huh? How the heck does that work?" Indeed, it is a rare occasion that we are stumped when it comes to technology, but AudioCodes (news - alert) had us stumped when we read the description of their TP-260 SIP Gateway.

The description explained that the TP-260/SIP is a complete "plug and play" Media Gateway on a PCI board that uses the PCI bus for power supply only and that no software drivers are needed. No drivers? Whoever heard of a PCI board with no operating system drivers? It just didn't fit the norm for how any hardware board works and this certainly intrigued us. Upon further investigation we learned that the TP-260 can be installed in any PCI server, no matter what the operating system. Further, it doesn't use the PC's CPU, so it doesn't affect applications on the PC at all.

We weren't the only ones stumped by AudioCodes product. AudioCodes told us, "We found out that many VoIP SIP gateway users do not like to put an external box with a different vendor's name in their solution. They prefer to have the gateway as an embedded solution inside their box. This solution is so different from what the market is used to — when you talk about a SIP to PRI VoIP gateway, or transcoding entity, we have discovered that users don't believe there is such a product. Once they've heard that such a product exists, they don't want to use an external VoIP gateway any more."

It can be used for several VoIP applications. For example, it can be used as a PRI to VoIP gateway with up to eight E1/T1s per single slot board. Secondly, it can be used as a transcoding entity, from almost any LBR coder to other LBR coder, controlled by SIP. The list of codecs supported include G.711, G.726, G.727, G.723.1, G.729A/B, MS-GSM, GSM-FR, GSM-EFR, AMR, EVRC. It can also be used to develop an IP PBX, IP IVR, and next generation switches.

According to AudioCodes, "The TP-260/SIP board is the first (and as far as we know — the only) stand-alone Media Gateway on a PCI board. All other gateways are either an external box, which is less suitable for the market we are addressing, or it's a board controlled by API running on the PC host (with an external SIP stack, and that means lots of development by the end user)."

For configuration, the user can configure the board in three ways (via the IP, from a remote PC, or from the local one), SNMP, and Web (the board has an internal Web server that the PC can connect to with a Web browser). In addition to having a Web server onboard, it also is a fully functional media server with voice prompts that can be stored on the board's memory (up to 10MB of compressed voice). The voice prompts can also be stored on a remote server, like an HTTP server. The board can connect

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to this HTTP server, take a prompt and play it to the IP or the PSTN (define - news- alert) side. (If required, the board transcodes the prompt.)

The TP-260 supports SNMP, G.168-compliant echo cancellation, T.38 Real-Time Fax over IP, as well as a wide selection of in-band and out-of-band tone detection and generation. The TP-260 has a wide selection of TDM interfaces for easy integration with other third-party CTI boards. The TP-260 complies with standard network control protocols including MGCP, MEGACO (H.248) as well as AudioCodes' proprietary TPNCP. In summary, users — especially developers and product designers — can take AudioCodes' innovative board, install it into their PC box, and with no extra development needed users can sell their product as one complete solution in a single box.

Brooktrout Technology

SnowShore IP Media Server

http://www.brooktrout.com

Brooktrout's (news - alert) SnowShore IP Media Server is an open, carrier-class IP media server designed to deliver advanced media processing for SIP (define - news- alert)applications, which include basic IP messaging and prepaid services to conferencing and video messaging.

According to Brooktrout, the SnowShore IP Media Server is the first "pure SIP" media server, which was written by Eric Burger, SnowShore's CTO. Its patented Web content integration engine enables the SnowShore IP media server to provide high-performance RTP to HTTP conversion — giving it a unique capability to support high scale VoiceXML-based messaging services in the carrier environment. Brooktrout SnowShore IP media server is the first media server deployed to power 3G wireless video messaging with its support of the H.263 video compression codec.

Brooktrout pioneered the use of SIP and VoiceXML for control of its SnowShore IP Media Server. Brooktrout's suite of media server products are open, carrier-class IP media servers designed to deliver advanced media processing for communications applications, from basic messaging and prepaid services to conferencing and video mail. With standard network and programming interfaces such as SIP, VoiceXML (define - news- alert), and MSCML, Brooktrout's solutions provide interoperability with a broad range of third-party Application Servers in next-generation wireline, wireless, and broadband networks. As proof of its open architecture and ability to interoperate, Brooktrout demonstrates an IBM eServer BladeCenter concurrently running BayPackets' Prepaid application, Ubiquity's Ringback Tone application, and the SnowShore IP Media Server.

Another unique feature of this solution is their Web content integration (RTP-HTTP conversion) that enables the high-performance scaling of VoiceXML messaging services. Their software architecture runs on IBM BladeCenter and Advanced Telecom Computing Architecture (AdvancedTCA) platforms — giving carriers an open, robust, and scalable SIP-based media server. In fact it delivers up to 500 sessions per blade and up to 7,000 sessions in a 7U chassis and is also available in a NEBS-compliant version.

It also supports an optional TDM interface, which extends IP media services to PSTN networks. The SnowShore IP Media Server allows customers to bridge TDM-originated calls, with reliable SIP and RTP interoperability between the media server and media gateway. Customers now have the flexibility to deploy flexible IP-based media processing resources that also connect to TDM environments.

CallWave, Inc.

CallWave Mobile Call Screening/Mobile Call Transfer

http://www.callwave.com

CallWave's (news - alert) history in the convergence of voice and the Internet goes back to 1999 when they launched their first product, the Internet Answering Machine, which currently has more than 800,000 paying subscribers today. Now CallWave has a new convergence product that leverages VoIP, call screening, and mobility via CallWave Mobile Call Screening/Mobile Call Transfer. Essentially, CallWave provides subscribers with a new, local phone number that rings on their existing cell phone. When a caller uses that new CallWave number, the subscriber will be able to screen calls and even transfer them to a regular landline phone, if they so desire.

With Mobile Call Screening, a CallWave user can perform call screening similar to that of a home answering machine. As a caller is leaving a message you can listen to it in real time and then press '1' on your cell phone to take the call, or simply let the message go to voicemail. Mobile Call Transfer allows the user to seamlessly transfer a live call from their mobile phone to a designated landline phone by simply pressing '2' on the handset. This feature helps stretch mobile minutes and is also convenient when the user arrives at work and wants to switch to the office line, or when they get home and the cell phone reception is poor.

Unlike customers of other VoIP companies, CallWave subscribers don't need to buy a Pocket PC or lug around a laptop to gain mobility. CallWave takes advantage of VoIP technology that is unique from any other VoIP provider. CallWave is using VoIP to separate applications from transport to provide value-add services to customers and allowing them to take advantage of these new services on the communications devices they already own. CallWave has a proprietary softswitch and they claim to be the only company that's currently using softswitch technology to offer these mobile phone features to the mass market. According to CallWave, "The technology behind allowing voice calls to be moved between landline, mobile and Internet actually requires extensive relationships with technology partners like Pac West and Level 3."

These applications are sold on a subscription basis and you can be live in seconds once you've ordered the service. While this product utilizes the traditional PSTN as the last mile to the consumer and not VoIP, CallWave responded, "As the cost of phone service continues to plummet, the VoIP transport layer is becoming irrelevant as a differentiator. Consumers are also becoming more sophisticated, leading them to demand more from their mobile and Internet-connected devices. The real market opportunity lies in the development of value-add applications such as the ones CallWave currently offers."

Covad Communications

COVAD VoIP with Voice-Optimized Access (VOA)

http://www.covadvoip.com

COVAD (news - alert) VoIP with Voice-Optimized Access (VOA) targets businesses with 10–200 employees per site. Covad VoIP with Voice Optimized Access (VOA) establishes two permanent virtual circuits (PVCs)-one for voice, one for data. This enables the network to allocate bandwidth dynamically and prioritize voice traffic over data traffic at the ATM layer (Layer 2) rather than the IP layer (Layer 3), eliminating the need for packet fragmentation, resulting in reduced jitter, improved bandwidth utilization, improved reliability, and superior voice quality — an absolute necessity in the business world.

Covad claims to be the first and only access provider in the industry to offer this type of prioritized service. It is available now on Covad T1 and symmetric DSL (SDSL) lines to offer customer an array of access mediums. Covad's VOA is available in 384 kbps, 768kbps, 1.1 Mbps and 1.5 Mbps SDSL, and full 1.5 Mbps T1.

The company leverages its nationwide broadband network to provide a managed VoIP service through their VOA technology. Covad told TMC Labs that they are the first company to offer nationwide business-class VoIP with managed quality to ensure that businesses get high quality, reliable communication services. According to them, "Before Covad VoIP with VOA, VoIP users lacked the ability to prioritize and segregate voice traffic from data on a single broadband line while automatically being able to dynamically adjust bandwidth between voice and data so that VoIP phone users never experiences jitter, echoes, etc." Covad added that, "Call quality is one of the top concerns keeping businesses from adopting VoIP, with this major issue resolved more businesses can choose Covad VoIP with peace-of-mind that they will be getting a true telephone company replacement product."

Covad VoIP with VOA is a unique VoIP service because it can overcome the barrier of last mile QoS by not only segregating and prioritizing voice traffic from data but also dynamically adjusting bandwidth according to real-time usage. Covad claims to be the only provider that can do this nationwide. The product also features Web administration and the ability to call contact with a click of a mouse. TMC Labs was quite impressed with the innovative features of Covad VoIP with VOA.

EagleACD

EagleACD

http://www.eagleacd.com

Contact centers are continually looking for alternatives to help preserve capital investment and outsource only telecom and data infrastructure. IP Hosted Contact Center solution providers are one arena where call centers can outsource their telecom and datacom infrastructure to reduce expenditures. EagleACD is a product of EagleIP, headquartered in New York City — that has been designed to eliminate high entry-cost barriers that organizations are faced with when deploying new products and services.

EagleACD's (news - alert) unique approach allows organizations of any size to offer voice ACD with skills-based routing, Web chat, and e-mail-ACD over either PSTN or VoIP — at no entry cost. In fact EagleACD makes it very easy to try their service at virtually no risk. Most of their services can be deployed within a matter of days, and without any up-front capital costs.

Early on, EagleACD realized the "utility-based" model for contact centers would be successful. The basic idea is to move computing/networking toward a "utility pay-for-use" model in which users can tap into vast pools of computing/networking power and use only what they need, only paying for the bandwidth processing/network connectivity and applications that are actually used. Eagle developed the industry's first truly innovative "pay-as-you-go" pricing scheme. According to EagleACD, "The idea that you pay for what you use is a big change from the call center norm. Moving from a fixed-cost, asset-based payment plan to a variable non-asset based payment for minutes used is a huge marketing innovation in the industry. We have worked very hard to create a very simple relationship with our clients. You only pay for minutes used. There is not a lot of extra billing for different items, no fixed monthly expenses per agent seat, no minimum. That is a big innovation to provide true-metered services for call centers."

Another truly unique innovation is what while there are a number of hosted contact service providers, all of them require minimum monthly payments. However, EagleACD does not. There are "no expenses" if there is "no business." Essentially, expenses are closely tied to the revenue stream. Under their business model, customers do not have any fixed monthly expenses for contact center agent seats. This is a strict interactive usage based business model where customers only pay for what they need. EagleACD told us, "This is widely used concept amongst utility companies. We are the world innovator for implementing utility concept in the hosted contact center industry."

Rather than lump all the features under one pricing scheme, EagleACD allows for a la carte selection of features. EagleACD has developed what they call the EagleACD Utility Grid to serve the contact center market, which currently has these prices:

- Online session — Web chat - $0.03/min

• Live agents — voice; $0.06/min
• Outbound call — Predictive; $0.08/min

One of EagleACD's references stated, "EagleACD's skill-based routing finds the right agent among all the agents connected to the network. This allows us to optimize staffing and increase service quality for better financial results — creating a more efficient company and more productive call center. No overstaffing is required due to this high reliability."

EagleACD's unique pricing model makes it a truly innovative solution. Add to the fact that it has a plethora of features including VoIP, Web chat, skills-based routing, multi-media ACD, IVR, predictive dialing, e-mail response management, call recording, CTI-like integration, and more, and TMC Labs can unequivocally state that contact centers looking for a hosted contact service provider should seriously consider EagleACD.

Edgewater Networks

EdgeView

http://www.edgewaternetworks.com

The EdgeView (news - alert) NMS allows network operations and field technicians to better troubleshoot problems that impact the quality of VoIP calls for business customers. Along with detailed call quality statistics including MOS, jitter, latency, and other measurements, EdgeView provides advanced diagnostics linked to an online knowledgebase that network operators use to obtain troubleshooting tips. This capability dramatically reduces the effort and time required to identify the root cause of poor quality calls. The EdgeView product is primarily targeted at service providers, while it provides remote monitoring, testing, and repair for SMB managed VoIP and video applications.

In addition to facilitating problem resolution, EdgeView provides visibility into the overall call quality performance of the network. Trend analysis and proactive notification of poor VoIP call performance enable the operator to identify and resolve issues that would otherwise impact VoIP service delivery. EdgeView also provides several administrative features that enable the scalability of the VoIP service including centralized management, node database backup, and restore and group upgrades. According to Edgewater Networks, "To the best of our knowledge, the EdgeView was the first to couple these measurements with an online database that explained them and provided troubleshooting guidance."

EdgeView takes common passive call quality monitoring measurements and presents them in a unique way that helps network operators save money, improve service, and increase customer satisfaction. EdgeView has several other features including identifying call quality issues with alarms, active call count reporting, and configuration backup and restore.

A customer reference stated, "The EdgeView enables our NOC technicians to become familiar with the customer experience without a truck roll. The EdgeView not only allows us to troubleshoot, but also provides us with the ability to rectify problems remotely. This saves us and our customers time and money."

Having good tools to test VoIP and ensuring the best voice quality is obviously a no-brainer. TMC Labs was impressed with the innovative trend analysis and database comparison to put the VoIP testing results in perspective. What they have done is make VoIP testing and analysis virtually dummy-proof. TMC Labs commends Edgewater Networks for simplifying the rather complex task of VoIP testing.

Empirix, Inc.

Hammer VoIP Test Solution for Enterprises

http://www.empirix.com

Empirix (news - alert) has been a leader in telecom testing for quite some time, so when they added datacom/VoIP testing to their testing arsenal, there was little doubt Empirix would have an innovative winner on its hands. Empirix is uniquely situated within the VoIP testing arena because it already has TDM testing expertise which they leveraged when developing their VoIP testing products. The Hammer VoIP Test Solution for Enterprises reduces risk and speeds the rollout of VoIP services and IP telephony applications (such as messaging, speech self-service, conferencing, and CTI) by accurately assessing how their infrastructure and applications will perform live, in production. It involves driving a high load of virtual callers, performing real transactions, into a VoIP environment and then measuring how the network and applications perform. Problems are flagged, and users can drill down to determine the likely source and fix them before the system goes into production. The solution is unique because it can simulate a real-world mix of callers, across both IP and TDM lines.

The solution is made up of three components:

• Hammer FXT, a feature test platform spanning IP and TDM that generates synthetic VoIP calls (signaling and media) and evaluates voice quality in real time;
• Hammer CallMaster, a graphical scripting and reporting tool for creating test call flows; and
• Hammer Call Analyzer, a diagnostics and troubleshooting solution that enables users to visualize and debug signaling and voice quality problems in VoIP and TDM networks.

Hammer VoIP Test Solution for Enterprises provides a comprehensive approach to testing enterprise communications environments that covers both networks and applications, across IP and TDM (define - news - alert) . According to Empirix, "This is the first solution to apply Empirix' patented Hammer technology to enterprise communications networks (versus carrier and service-provider networks)."

This product is uniquely suited to look at both voice quality (both packet and call level measurements) as well as testing the correctness and performance of applications such as IVR, routing, voicemail, etc. Empirix uses speech recognition technology to assess the correctness of these applications, voice quality measurement technology to connect the network characteristics with the performance of the applications, and a breadth of protocol support (SIP, H.323, MCGP, etc., as well as all major TDM signaling protocols) to cover a breadth of equipment types and interoperability scenarios.

FiberTower Corporation

Backhaul Services for Wireless Carriers

http://www.fibertower.com

Optimizing and increasing backhaul capacity and reliability has become a critical element especially as voice, video, and data application usage continues to climb. FiberTower (news - alert) is one of only a handful of companies to address backhaul requirements, particularly those associated with growing 3G capacity demands. While backhaul is the most critical component of the wireless network for providing sufficient capacity and scalability, it is also the portion of the network that has historically been most limiting and most expensive to wireless carriers. Until now, the backhaul space has been completely dominated by the ILECs, creating a monopoly market that offers no alternative for wireless carriers. Considering that backhaul is one of the fastest growing cost of service (COS) items and accounts for over 60 percent of cell site outages, lack of a viable backhaul alternative has rendered wireless carriers helpless to improve a major weakness in the reliability and performance standards of their networks.

With FiberTower wireless carriers now have a choice in their backhaul service and an opportunity to significantly improve customer service, network management, and cost effectiveness. FiberTower enables carriers to scale to meet next generation capacity demands — commonly referred to as 3G — and improve their current service standards. With over 170 million Americans using cell phones today and that number expected to increase substantially with the arrival of multimedia convergence devices, performance levels, and cell site outages have become of paramount concern to carriers.

FiberTower's backhaul solution was built expressly to scale to meet the growing demand for wireless voice, data, and video applications. By inventing a new way to conduct backhaul traffic and eliminating the legacy copper T1 land lines that carriers traditionally employ, FiberTower not only improves today's standards of service reliability, but also positions carriers to handle future capacity demands.

FiberTower's solution employs a model currently used in Asia and Europe, which utilizes a hybrid solution of fiber and digital radio links (DRLs) to provide optimal backhaul service. Until recently, backhaul in the U.S. has been dominated by aged legacy copper links, which suffer from limited capacity, and are also responsible for frequent network outages, sub-optimal repair cycles and an inability to scale quickly to meet escalating bandwidth demands. FiberTower leverages a combination of fiber and digital radio links to alleviate the risk of outages resulting from land obstructions.

When we asked for any other unique aspects of their solution, FiberTower responded "We have managed to extract entirely new levels of performance and reliability out of microwave digital radio links by marrying the proven capacity benefits of microwave technology with the enhanced reliability inherent in a fiber infrastructure. Existing legacy copper backhaul suffers from shortcomings in both capacity and reliability, and FiberTower's hybrid solution takes the best of both worlds — microwave and fiber — to address these shortcomings." Proven microwave technology replaces the aged legacy local copper loop, dramatically enhancing capacity by offering up to 155 Mbps (megabits) of storage versus the 1–3 Mbps provided by a copper loop.

As high-speed data applications, including VoIP, increase in popularity wireless carriers are under increasing pressure to scale and deliver bandwidth in a reliable, cost effective way. Through its facilities based approach, FiberTower claimed that it reduces backhaul costs by 10–30 percent based on current tariffs. We commend this unique and innovative hybrid solution for helping to solve the bandwidth problem while simultaneously reducing costs.

FrontRange Solutions

IP Contact Center (IPCC)

http://www.frontrange.com

IP Contact Center (IPCC) is a SIP-based contact center solution with an advanced skills-based and data directed routing engine with text to speech, automated speech recognition and common data speaker fully integrated. Because the solution is server-based, there is no need for specialized or proprietary hardware. IPCC integrates seamlessly into existing infrastructure. IPCC is the first IP-based contact center to provide telephony integration with FrontRange's (news - alert) core CRM/service management products, HEAT and GoldMine. It is a cost-effective and seamless way for these users to integrate queuing and routing solutions (among the other product features) without transitioning to and paying for proprietary architecture.

When asked about some unique aspects of IPCC, FrontRange responded, "Call recording is a good instance of a technology that we've replaced. Because we are software in a SIP-based solution, we can offer call recording without any additional hardware or costs." FrontRange also explained that on the CRM side, with GoldMine, they have an integrated SIP soft phone that will speak with multiple vendors' proxy providing us interoperability with service providers and customer premise equipment solutions. "With the tight integration of our VoIP (SIP) solution to our Service Desk and CRM applications, we are creating the VoIP 'Killer Applications' through advanced unified communication and application integration that customers will look to as the benchmark for return on investment analysis."

IPCC features some powerful self-service applications including IVR. Within the IVR, FrontRange Solutions'

IPPC has the ability to integrate text to speech which provides a human-sounding voice for names, addresses, and product descriptions. Self-service is also enhanced by automatic speech recognition and "common data speaker" where dates, times, money amounts are all built-in and delivered on a multilingual vocabulary-based basis.

Another unique feature comes from the integration between IPCC and GoldMine for customer relationship management. IPCC has the ability to blend inbound and outbound calls. Outbound campaigns are automatically put "on hold" when a matching inbound call takes place. TMC Labs was impressed with the tight integration of VoIP with one of the most popular CRM applications on the market today, as well as the advanced feature-set including TTS and speech recognition — making IPCC truly a unique and innovative solution.

Global IP Sound

VoiceEngine Mobile

http://www.globalipsound.com

Global IP Sound (news - alert) is renowned for its excellent Voice over IP codecs. Several years ago, before Global IP Sound was well known, TMC Labs tested their voice codec on a Pocket PC and used Shunra's network emulator to induce jitter and latency into the VoIP call. Long story short, the Global IP Sound codec handled the extra latency and jitter with no problem. We were very impressed then, and today Global IP Sound continues to impress. Many VoIP softphones embed Global IP Sound technology including Skype and Teleo. Now Global IP Sound has decided to go after the mobile handset market, specifically the "cool" dual mode WiFi/3G handsets which are coming to market.

VoiceEngine Mobile is embedded voice processing software that enables application developers, service providers, and OEMs to create mobile products that are both easy to use and that provide substantially better voice quality than previous generation products. The solution is optimized for devices that run on the Symbian operating system, such as smartphones or dual-mode cellular phones, as well as for Windows-powered personal digital assistants (PDAs).

They have very advanced speech processing algorithms that provide the high-quality voice and this is what sets Global IP Sound apart. Global IP Sound was quoted as saying, "It is the first product to bring such a high level of quality (better than PSTN and current cellular telephony) to mobile communications, and is on the cutting edge by combining VoIP with current mobile technology." Further, they also claimed, "VoiceEngine Mobile not only provides, hands down, the best possible voice quality available on the market, but also enables applications developers to focus on development without having to worry about integration. The result is a superior product that gets to market more quickly." After witnessing the superb voice quality for ourselves firsthand in their earliest product years ago as well as embedded in Skype, we certainly do not dispute their claim.

The packaged solution handles all of the necessary voice components for VoIP to achieve superior voice quality, even under adverse network conditions. VoiceEngine Mobile manages all of the most challenging problems encountered in mobile IP applications, including packet loss, delay, and jitter. Using the solution with its comprehensive and easy to use API, vendors can quickly develop products that give the end user greater mobility and that incorporate excellent voice quality without in-house expertise.

Go2Call

Hosted VoIP Global Platform

http://www.go2call.com

The Go2Call (news - alert) Global Platform enables customers to become VoIP service providers — that is if you want to be the next Vonage, you can use Go2Call's hosted solution. As a fully hosted, turnkey solution, the Go2Call Global Platform incorporates all of the necessary elements to create, manage, and deploy feature-rich VoIP service offerings. The Go2Call Global Platform enables customers to target solutions for the residential, enterprise, mobile, and call shop end user.

Go2Call's comprehensive hosted platform includes global call termination and origination, an OSS providing Web-based business management tools and pre- and post-paid, multi-tiered billing flexibility in real time. Go2Call claims that their platform is the first to place all of these elements in one hosted package that can be completely customized-from the branding to the service options by a customer.

One of the hottest new trends in software applications is allowing customers to customize the look and feel of the application through the use of "skins." The Go2Call Global Platform has recently incorporated a customized SIP Dialer V9 into its offerings that leverages SIP and HTML, and it provides fast and reliable PC calling. It works in a wide variety of Internet environments and custom skins permit Go2Call customers to modify its appearance and branding.

Go2Call's Global Platform includes several unique features. The real-time billing and provisioning systems support multi-level distribution, allowing service providers to effectively manage and sell VoIP services through their distribution networks. Go2Call's platform supports provisioning and billing for any SIP-based voice service, including PC-to Phone, Device-to-Phone, or traditional calling card services. Go2Call's advanced yet easy-to-use Web-based tools include powerful features for managing all aspects of deploying a VoIP service, including DID provisioning, rate management, and account management systems.

While offering the customer the benefits of a hosted solution, the Go2Call Global Platform simultaneously empowers the customer with the ability to customize and control the business at multiple levels. Go2Call integrates white- or private- label branding into all layers of a business. With the Go2Call Global Platform, the customer has capacity to customize everything from end user tools, such as the SIP Dialer V9 and IVR, to distributor materials, such as the multi-tiered administration tool and marketing documents. IT

Unified Communications Meets VoIP:

A Marriage Made In Heaven

By now we know that IP systems and networks are being implemented at record pace, as enterprises of all sizes and in all types of industries recognize the value of IP. But contrary to popular belief, the real value of IP and voice over IP (VoIP) is not in inexpensive transport, cheap phone calls, and simpler management of a single network. Rather, the true value is in the new applications IP enables.

Looking beyond network efficiency, VoIP enables a range of applications either not previously available or requiring significant integration and customization, such as the multimedia contact center, integrated Web and audio conferencing, and most notably, unified communications (UC). Converged applications such as UC can take advantage of IP and enable companies to not only increase revenues and decrease costs, but enhance productivity and improve customer service.

But if you're not ready to take the plunge to IP technology, don't worry — you can implement a UC system today on a legacy TDM (define - news - alert) system, benefit from the UC applications immediately, and still reap the additional benefits when you migrate to an IP network. UC applications are supported by both TDM and IP-based communications systems, and newer UC systems will enable your organization to implement a UC system now, use it with your traditional phone system, and then easily migrate to VoIP later, when you're ready.

Unified Communications — Beyond Messaging

You're probably asking yourself, isn't unified communications a fancy name for unified messaging? Let's be clear — Unified Communications builds upon UM, but adds much, much more.

Unified messaging delivers user access to voice, fax, and e-mail messages through a single interface, typically via the telephone, desktop PC, or mobile device. Unified communications adds real-time call control, collaboration, media handling, and further integration of voice and data applications.

Additionally, UC systems provide real-time call control and call completion applications including find-me/follow-me services, automated call return capabilities, missed call log with call return capabilities, and outbound dialing from groupware contacts or personal address books.

With UC, users can access their groupware calendar to accept or reject meeting requests via a speech interface over the telephone, set up new appointments or meetings, as well as check to see what is on their schedule.

The latest generation of UC systems includes "personal assistants" with speech recognition technology to help mobile employees manage their communication, including messaging, call control, groupware, and other enterprise data.

The above UC applications are all available today for traditional circuit-switched PBX (define - news - alert)environments, as well as for IP-PBX environments. However, additional capabilities will emerge with the move to VoIP.

Adding VoIP To The Picture

As enterprises implement IP voice and data networks, they are looking for applications that can run on these converged networks, and UC is one of the key applications that can take advantage of convergence.

When the voice server becomes a node on a network, there are more

By Tom Minifie

opportunities to integrate voice communication and messaging with other network applications. VoIP and UC have a natural synergy, and with the convergence of voice and data across one network, the barriers to unifying the different communications are immediately lowered. By utilizing standards-based UC and IP-based systems on a single voice and data network, there is little integration required.

We can look at the benefits of uniting UC and IP in two different ways — in terms of the features and functionality, and in terms of the system and network architecture.

The newest generation of IP-enabled UC systems provides opportunities to integrate instant messaging (IM), presence awareness, features such as click-to-call, click-to-conference, Web and voice conferencing, Web or multimedia chat, and document collaboration, all via a common user interface. By integrating these various technologies, UC systems can do more intelligent routing based on what's on the user's calendar, their presence status, and personal rules. UC systems can send instant messages to users to notify them of received messages or other communication events of interest.

When looking at the architectural benefits, IP-based UC systems are simpler to integrate, easier to expand and scale, and less expensive to manage. As VoIP is very standards-driven, the integration of converged applications such as UC is more straightforward and cost-effective given the removal of proprietary interface software and hardware components. When implementing new IP PBXs, it is relatively easy to add UC and it does not require some of the integration efforts that were needed to integrate UC on TDM platforms. The time to deploy UC solutions is greatly reduced in a standards-based, single-protocol environment.

What To Look For In A Unified Communications System

When looking for a UC system, here are some things to consider:

• Even if you're not deploying IP-telephony today, be sure that your UC solutions can work with IP-telephony when you eventually go there. Look for a UC system that can support both IP and TDM, particularly in a mixed environment. You shouldn't have to change your switching fabric when moving to a UC system. You will want to move from your current voice mail system to UM/UC while being able to migrate to IP when you're ready.

• Make sure that the system lets you start with voice mail and "turn on" UC capabilities on a user-by-user basis. There is no reason for you to have to deploy all UC capabilities to your entire user community. Not every user has the same need for UC so you shouldn't have to pay for or support UC for every user.

• The ability to support multiple switches will ensure that you don't get locked in to a single vendor as you move forward with your UC and VoIP deployments. Look for a UC system that can work on a variety of switches, and even support both TDM and IP switches on the same server.

• Flexibility is a key so support for multiple e-mail or groupware systems, including Microsoft Exchange, Lotus Notes, and others, is a must.

• If migrating from a legacy voice mail system, look for a UC system that lets you use the same or similar user interface (both GUI and TUI) to what you're used to. Some UC systems can replicate the user interfaces of legacy voice mail systems, eliminating the need for users to learn new commands to navigate the system. IT

SIP Is The Key

Session Initiation Protocol (SIP) has emerged as the protocol of choice within the VoIP market, especially within IP PBX solutions. Because SIP is a signaling protocol for Internet conferencing, telephony, presence, events notification, and instant messaging, it is the ideal protocol to be used by UC systems. Companies like Avaya, Nortel, Siemens, and even Microsoft (with its Live Communication Server) are all embracing SIP as a standard of choice for their VoIP solutions.

According to Blair Pleasant, President & Principal Analyst at COMMfusion, "UC is expected to experience additional market adoption because of the addition of presence capabilities, made possible in part by SIP. Presence will bring about a new set of applications that will further enhance our ability to interact with one another."

When integrating UC and presence, UC systems can offer real-time, presence-based access to people, calendars, and files. We will see "personal communication portals" that provide a unified way to perform various communication tasks, such as handling voice calls, e-mails, and instant messages, based on a personal rules engine. More advanced systems will allow the presence engine to change a user's status based on their calendar (i.e., in a meeting, out of the office, etc.). The UC system will be able to play the appropriate response to callers based on the user's status.

Expect to see presence-enabled UC become the real-time collaboration component of enterprise application software.

A Graceful Migration

Migrating to an enterprise unified communications solution doesn't necessarily mean replacing current phone systems or jumping headfirst into the world of IP telephony. Instead, an enterprise can layer a UC solution across all existing office locations and phone switches — both IP and legacy. This can reduce the management burden, provide all users with a consistent set of productivity-enhancing services and present callers with a unified company image while allowing the enterprise to transi-

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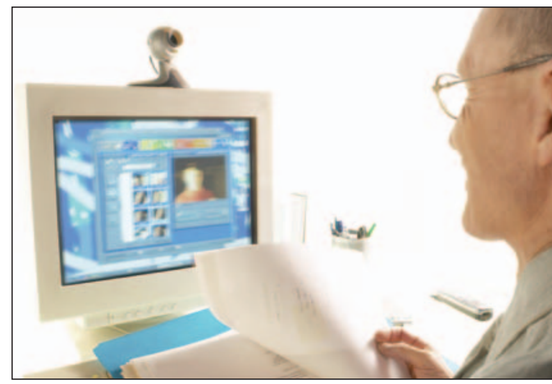

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tion its legacy telephony switching environment at its own pace.

Even if customers aren't deploying IP-telephony today, they need to be asking questions about compatibility, and be sure that their UC solutions can work with IP telephony for when they eventually move to IP. In many cases, UC systems are the first step toward a company's eventual migration to IP. Companies can start the migration process by implementing a UC system on their TDM switches, and continue to use the same UC system with their new IP-PBXs when they're ready.

When deploying a UC solution, an enterprise should be able to upgrade its existing phone switches, or migrate to IP telephony, with virtually no disruption to employees and the organization, and no need for retraining. There should be no need to replace the UC solution when a company changes its phone switch or migrates to an IP-based switching solution. The UC solution should simply integrate with the new IP switch. This means that users don't have to learn new interfaces and companies get to keep the same robust set of features and level of reliability they have become accustomed to even as they move to an IP-based telephony environment.

Unified communications systems provide a multitude of benefits to companies of all sizes and across all industries by enhancing communications and productivity. For those companies that are ready to dive into the world of VoIP, unified communications is the ideal convergence application to get started with. For those companies that want to wait a little longer before replacing their legacy PBXs, by upgrading from a legacy voice mail system to a UC system, they can reap the benefits of this leading-edge enterprise communications technology while gradually migrating to a full-blown converged network. **IT**

Tom Minifie is the vice president of product management for Applied Voice & Speech Technologies, Inc. (AVST), a developer of communications solutions for businesses of all sizes. For more information, please visit http://www.avstgroup.com.

Enterprise IT: Friend Or Foe Of Unified Communication Systems?

By Grace Tiscareño-Sato

Flexible and unified communication technologies, along with a slew of work-focused mobile devices, are now abundantly available to help drive greater enterprise productivity. Such systems and tools encourage employee commitment and more diligent work, wherever an employee may roam. Yet, organizations have been slow to fully integrate these solutions into the workflow mainstream.

So, what's holding organizations back?

It may be that the organizational teams most responsible for championing new technologies — enterprise IT departments — are now the stumbling blocks for enabling enterprises to take full competitive advantage of communication tools and solutions. Hiding behind old attitudes and fears about new collaboration tools and potential threats to security, enterprise IT seems to be more adept at disabling rather than embracing emerging communication solutions.

Just take instant messaging as one example. According to a new 2005 Nemertes Research study, while 76 percent of IT executives say employees use IM on the job, only 37 percent say they have standardized on an enterprise IM application. This compares to 90 percent of IT executives who reported employee IM use in 2004.

"We are seeing a knee-jerk reaction that deals with IM security and compliance problems by just shutting it down altogether, which is of course not something we recommend as a viable long-term strategy," said Melanie Turk, a principal at Nemertes.

How entrenched are the misgivings about the adoption of unified communication tools, in general? Take a look at some recent seminar titles at this year's RSA conference in San Francisco:

- Airborne viruses;
- Remote instant messaging: vulnerable anytime, anywhere;
- The next generation of messaging threats.

From an IT perspective, it's clear that there are few warm-and-fuzzy feelings about unified communication. This has tremendous business consequences in a global economy where knowledge is the only scarce resource and one that only has value at the moment the business needs it. Blocking the flow of knowledge to any degree can be detrimental for a business. This is especially so for an enterprise whose competitors are embracing emerging technologies that accelerate the flow of business knowledge and therefore make faster, better-informed decisions.

Said one professional working mom who works from home 75 percent of the time: "I didn't realize I was being an enterprise renegade when I purchased a PDA with a built-in phone. My IT shop wasn't buying mobile devices it so I bought myself, thinking I could use it to keep myself more connected to my team in the many places I work."

Hers is a common complaint. A recent Siemens survey found that despite the growing number of communication tools and devices available, most workers remain dissatisfied with their ability to reach colleagues and obtain critical information when it's needed. Among the top complaints: 67 percent of U.S. workers say they must leave multiple messages in different places when seeking immediate responses; and 65 percent say that decisions are delayed because colleagues fail to respond in a timely manner. The problem is that various communication tools and devices continue to exist on isolated islands, with nothing tying them together.

Millions of slick new mobile devices are sold each year to consumers looking for a personal competitive edge. However, only a fraction of these tools' capabilities are ever realized. "My smart phone could have made me more available when

I'm between locations and tied me into critical enterprise applications and databases. Instead, it's an expensive phone/organizer completely unsupported by IT. Plus, it's making me work beneath my potential — it's their loss."

For IT, the fact is that mobile or nomadic workers are here to stay and, in fact, their numbers are growing. According to analysts at IDC, the number of mobile workers in the United States will reach 105 million by 2006, representing roughly two-thirds of all workers. This mobile workforce will continue to demand highly flexible communication tools that deliver greater efficiencies — at, around and outside of the corporate campus environment. The influence of this group will also continue to grow, whether IT is a willing participant or not. Today, more than half of U.S. companies actively support at least one wireless network and 22 percent plan to deploy wireless technology by the end of the year, according to Jupiter Research.

Enterprises do not have to accept communication chaos and communication limitations. Better solutions and tools exist, but enterprises and IT leaders need creative new strategies to turn unified communication and mobility into competitive advantages.

Start By Listening

An organization-wide communication assessment, from the end-user community's point of view, will begin to shed light on collaboration and productivity enhancement needs.

Begin with the most valuable workers: revenue generators, customer-facing experts, and knowledge workers (the people you'd never want to lose to your competitor). Ask them what applications they have at work that they would like to access when mobile? What systems and devices can you put in place to make critical decisions faster? Do they need wireless access on campus? Do they need more presence-driven and permission-based collaboration tools? Would speech access to workflow applications help? Think globally too. What tools do workers need to be more successful collaborating across time zones and across domains? A soft client on a laptop, integrated with a corporate IP/PBX, for example, may be the perfect solution for global workers in different time zones from colleagues and customers.

Be Creative

Stretch beyond issuing a laptop or desktop to every employee. Seriously explore the costs and benefits of other tools and devices. In a bold move, Volvo Cars, a division of Ford, recently announced that it would completely do away with desk phones for 5,000 of its most mobile employees and equip them all with wireless phones, unified messaging and a one-number service solution. By packaging together a large solution, Volvo was able to negotiate a fixed, predictable rate for mobile expenses and, at the same time, equip high-value employees with better unified communications tools that are more suited to how they work.

Maintain Control

This is where IT can maintain control of its voice and data infrastructure. IT staffs must methodically think through how devices will be procured and supported with the right applications needed to maximize user and team productivity. They must also think through how these valuable devices will ultimately be returned to the enterprise if an employee leaves. A great deal of valuable enterprise knowledge is generated on communication tools and devices used by high-value employees. Formalizing, documenting, and communicating policies are critical to retaining information such as customer contacts, future product roadmaps, sales strategies, competitive knowledge, etc. In fact, this is a major benefit of enterprise-issued gear. How likely is it that an employee will leave behind information on a mobile device they purchased with their own money?

Keep Up

Stay abreast of and ensure foundational enterprise support for the next generation of unified communication tools (presence, IM Push, soft client support, and more). This includes complete IT support and planning — across secure platforms — of all tools that make new business process speeds achievable (including wireless, fixed, premise-based, and service-provider solutions). Competitive advantage goes to the company with the most equipped and informed users, who are able to apply their knowledge and expertise at the moment it is needed. Otherwise it is lost.

The communication choices that workers need will become clear, and they will also be the choices that are most important for the organization, as long as the evaluation process starts with users, business processes and communication needs that drive workflow.

Wainhouse Research shows a tremendous difference in results when planning and deployment starts with business process versus technology. When IT planning and support begins with studying daily processes and user behavior and then applies the right technology tools to achieve improved workflow, organizations achieve as much as 30 percent cost savings, in addition to accelerated business processes. On the contrary, when IT plans and deploys technology because it's exciting, cool, and new, then tries to find the right users and processes for it, organizations typically experience as much as a nine percent cost increase.

While the majority of organizations, according to Larstan Business Reports, believe that getting mission-critical information to mobile employees is vital to strategic objectives, only 22 percent have enterprise-wide mobility initiatives in place and only 24 percent say their current infrastructures are optimized to support these initiatives. Fewer still appear to have put much strategic thought into how mobility may change the fundamental operations — and therefore, the competitive rules of engagement — in their industries, according to the report, "Managing the Mobility Imperative."

It's time for IT and security managers to work with enterprise executives to actively adopt real-time unified communication strategies that give workers a complete set of voice, data, video, and Web-based collaboration tools that operate across time, distance, and network barriers. Simply put, it means strategic support of anywhere, any device communication. **IT**

Grace Tiscareño-Sato is global marketing manager, Unified Communications, at Siemens Communications, Inc. For more information, please visit http://www.siemens.com.

Security For Service Providers

As we stand on the threshold of entrusting our communications to a global IP-based telephony network, it seems the perfect time to discover just how secure these networks will be and explore the issues associated with their security.

Phone system security concerns started in the 1960's. Initially it was a pastime amongst enthusiasts who discovered that playing certain tones through telephone receivers made the telephone companies' switches believe a call had ended, and thus got free phone calls — hence the term 'phreaking'. Colorful stories abound regarding 'Cap'n Crunch' who discovered the free whistle in a cereal packet could play just the right tone to get free calls.

From Circuit To Packet

Since then telephone networks have undergone several evolutions and the security of those networks has evolved with them. With each evolution two things have remained constant; the telephone network is based on circuit-switched technology, and it is a 'closed' network.

• A circuit-switched network creates a dedicated path between the two parties for the duration of the call. This means that a call ties up resources throughout the network for the whole call, even if the call is silent.

• A 'closed' network is one for which the telecommunications operator controls how users communicate across the network. This should mean that handsets cannot access individual elements within the carriers' network.

However, the advent of voice over IP (VoIP) changes all this in quite a dramatic way; the network is packet-switched and is, in effect, an 'open' network. Packet-switched means there is no dedicated path set up for a VoIP call. Individual packets are routed through the network to the destination as they appear. This allows for a more efficient network design carrying more calls over a cheaper router-based network.

Being an IP-based network, it is 'open.' Both terminals and network elements communicate using public protocols, meaning that potentially anyone can access and disrupt any network-based elements. In such an open network, the challenge is to allow legitimate traffic to flow freely whilst maintaining the necessary level of security.

What Is Security?

Security of a public voice service covers a wide range of areas, these include:

- Securing the network from other connected networks.
- Protecting the subscribers' from attack.
- Protecting the subscribers privacy.
- Protecting the infrastructure from attack or misuse.
- Complying with regulatory requirements for legal interception.

Protecting The Borders

In traditional circuit-switched networks, each carrier hands off a call to a peering carrier at a distinct demarcation point. At this point, it is possible to limit the internal information passed to the neighboring network and monitor the volume of calls passing between the networks.

With the advent of global VoIP services, IP networks must be interconnected and protected in a similar way. When multimedia services cross a border, each call is actually made up of a number of streams. Devices need to understand the linkage between these streams in order to handle them correctly. Hence, a new class of equipment — session border controllers — have been designed to fill this role.

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By David Gladwin

The session border controller acts as a proxy for both signaling and media as they cross the border into neighboring networks and therefore acts as a defined demarcation point for the network. A session controller acts as a ‘pinhole firewall’, opening and closing paths for media under the control of authorized signaling sources. Unexpected or unauthorized traffic is simply discarded, and true carrier-class, hardware-based session controllers will perform this at wire-speed with no loss of performance for authorized traffic. In this way, IP-level Denial of Service (DoS) attacks can be compartmentalized, thus limiting the scope of their disruption.

Topology Hiding

One feature of VoIP calls is that certain packets can accumulate information about the network elements that it has crossed, thus providing anyone snooping this information with a roadmap of the network — valuable information if you are planning to disrupt that network. The session controller eliminates this problem by removing all internal network informa-

As carriers evolve and expand their service offerings, the adopted underlying network of choice remains IP.

tion and presenting itself as the originator of the call — this is called topology hiding.

Protecting The Subscribers

Public Presence

In order to be able to both make and, most importantly, receive calls, the subscriber must be identifiable within the public network. The subscriber's IP phone registers with a server, which advertises the subscriber's presence. This means the subscriber's phone is an easily identifiable target. A second valuable role carried out by session controllers is to act as the public point of presence, or proxy, for the subscriber. This provides privacy for the subscriber as their real address is known only to the session controller.

In this way, any attack directed at the subscriber is now handled by the session controller, a dedicated hardware device built to the exacting standards of carriers. The session controller offers a high level of protection by handling attacks directed at individual subscribers and also by preventing a loss of service caused by attacks designed to saturate the access network with rogue traffic.

Denial of Service attacks need targets, the real addresses of subscribers are never revealed within the IP packets crossing the network, thus making it considerably more difficult for hackers to identify potential targets. Some types of attacks, known as Distributed Denial of Service (DDoS) attacks, cause disruption by generating large amounts of traffic from many sources: the session controller identifies and blocks these attacks before they enter the access network.

Managing Access & Service Theft

Access networks are usually dimensioned to handle typical Web browsing, downloads, and e-mail traffic. These types of traffic are not particularly sensitive to variable delays and remain relatively unaffected by over-booking. However, multimedia traffic is sensitive to both delay and variations in delay. As a result, over-booking of access networks can lead to a significant degradation in the quality of multimedia services. This problem can be minimized by policing the multimedia traffic that is admitted into the access network. This maintains the quality for all calls.

The session controller can also police individual calls to ensure the bandwidth being used in a call complies with the requested resources, thus reducing the opportunity for service theft. Excessive SIP signaling rates can also be restricted thus limiting attempts to overrun a device.

Protecting The Infrastructure

Many elements within a VoIP network could be subject to attacks both at the IP level and at an application level. The actions taken to protect the borders and the subscribers also protect vital infrastructure elements. Since a subscriber registering with the VoIP service is given the address of the session controller by the DNS, it is the address of the session controller that is advertised not the softswitch. Thus any malicious signaling traffic directed at the softswitch can be policed and if necessary, rate limited by the session controller. For example, invalid or inappropriate packets are simply discarded, while surges in apparently valid signaling can be rate limited, thus relieving pressure on the softswitch.

Regulatory Compliance

Agencies such as FCC and ETSI are starting to insist that VoIP networks should comply with the same regulatory requirements as their circuit-based counterparts. These requirements can be diverse and country specific, but in general encompass Lawful Intercept (LI) and Emergency Call handling (ECH).

Lawful Intercept capability dictates that given the correct authorization, a user's communications must be intercepted. Within any network it must be done without the user being aware. All calls that the user makes traverse a session controller and this device can be used to duplicate the relevant signaling and media streams. As Lawful Intercept requirements evolve, the same methods can be used to encompass the interception of Instant Messaging, video, etc.

Emergency Call Handling requires that when a subscriber dials 911, their call is routed to the most appropriate Emergency Call Center, generally based on the location of the caller. If the subscriber is not able to convey their location, due to illness etc., then the Emergency Call Center must be able to trace where the call originated. This is done by interrogating the Broadband RAS to map the user's IP address to their actual details.

The Global Multimedia Net

As carriers evolve and expand their service offerings, the adopted underlying network of choice remains IP. Standards for deploying and interconnecting multimedia in both wireless and wireline networks such as IMS (define - news - alert) and TISPAN (define - news - alert) are being developed and are gaining momentum. Running through all these standards is the need to build a secure, reliable and controllable network.

So, perhaps the days of worrying about a phreaker with plastic whistle are gone, but as networks have evolved so have the threats. Security remains a cornerstone of good network design, and session controllers are establishing themselves as the key component for ensuring IP networks maintain the familiar standards associated with the PSTN. **IT**

David Gladwin is marketing manager at Newport Networks. For more information, please visit http://www.newport-networks.com.

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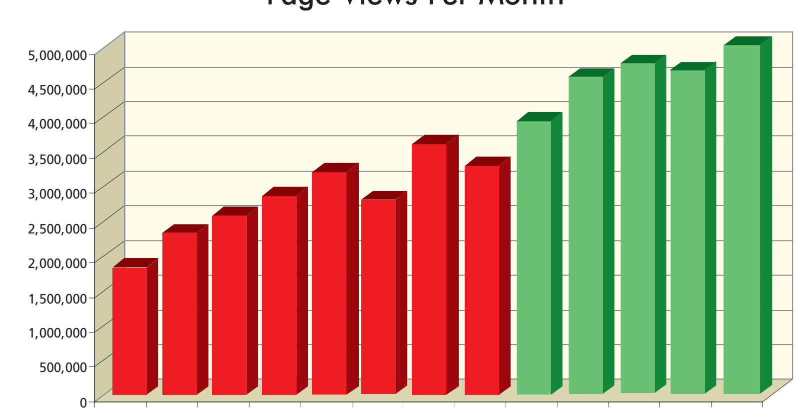

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Enabling And Accelerating Channel Revenue

A few years ago, voice over IP (VoIP) sounded like a great concept on paper. Today VoIP is no longer just a concept, but a firm reality with millions of residential and business customers, and the numbers are growing. In fact, according to Atlantic-ACM, consumer VoIP has grown from about 15,000 users at the end of 2003 to approximately 1.2 million users at the end of 2004. VoIP providers got a huge assist in the form of government deregulation and a growing number of customers who are accepting the idea of gathering all their information (voice, video, and data) over an IP network rather than traditional mediums.

With barriers to VoIP being lifted, a path is cleared for service providers to promote the technology and its wide range of benefits to customers of all sizes. In fact, Yankee Group estimated that 20 percent of all new business phone systems installed in the United States last year utilized some type of VoIP (define - news - alert) technology.

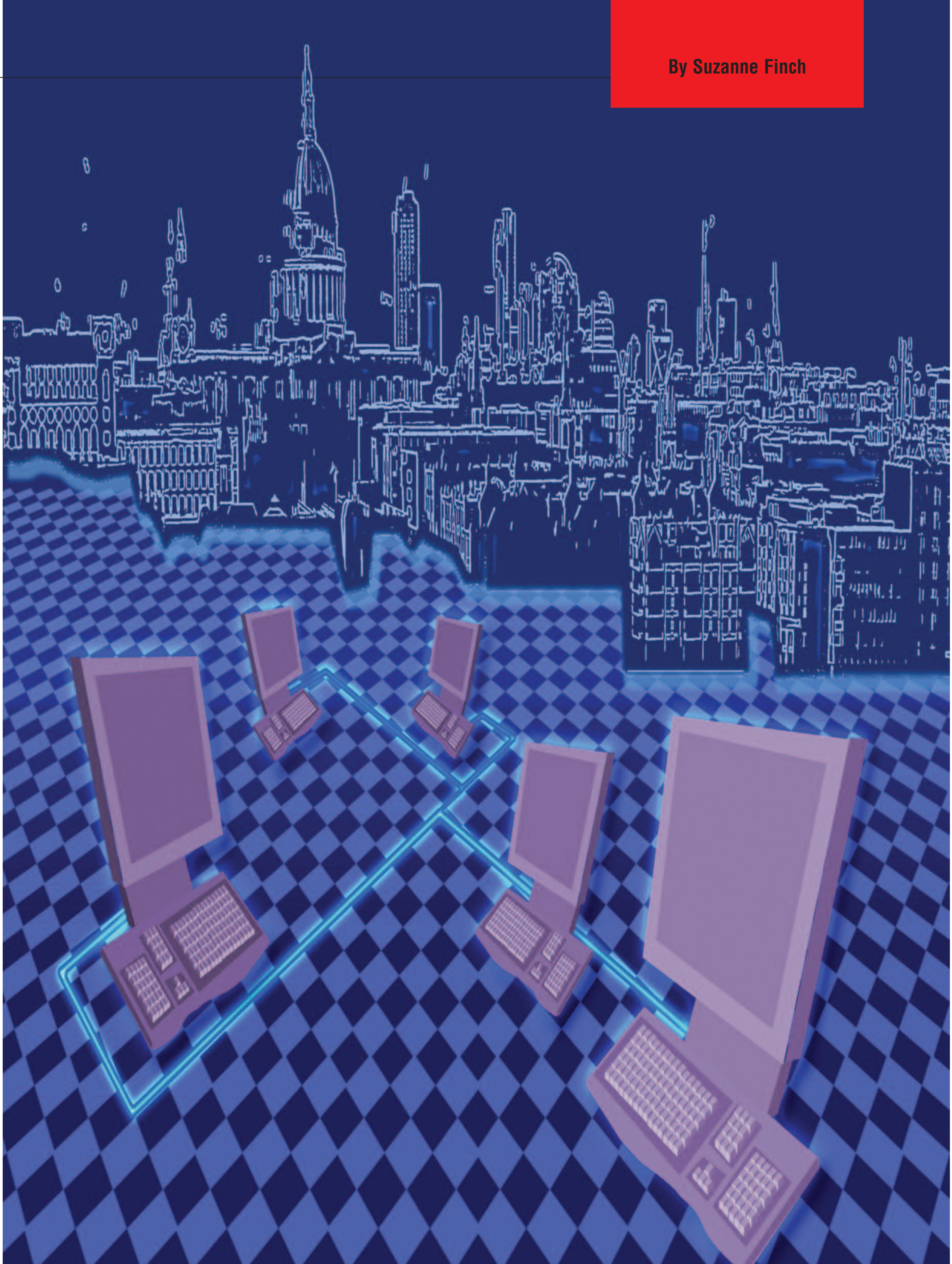
One of the benefits of VoIP is that it goes hand-in-hand with other services such as e-mail, domain name registration, digital rights management (pay-per-view), streaming video, and other services that add value for the customer and revenue for the provider. Services can be converged and bundled to offer attractive packages to potential customers simply by virtue of the fact that instead of several invoices, they now have one vendor and one bill to deal with each month. But as competition for customers increases, so does the need to build a broad resale channel sales force that can contribute to creating a large end-user client base.

Enter channel marketing. For IP service providers, channel marketing includes the use of partners, sometimes known as channels who market directly to the customer as consultants, re-packagers, agents, or resellers. While direct sales is the highest dollar-cost approach to selling and delivering services, indirect channels can often sell VoIP services most efficiently.

The new channel marketing model now includes resellers that identify with a specific market or ethnic audience and has the advantage of being able to customize their messages and service bundles to fit their targeted market. Other resellers may have expertise in the residential market and still others focus on specific types of businesses. The vast assortment of service resellers enables the wholesaler/provider to market to an audience they might not otherwise be able to access through traditional methods.

With all the advantages offered by utilizing the channel marketing model, wholesalers and providers are then tasked with the burden of trying to find a method of effectively bundling and delivering those services to their resellers. To do this, wholesalers/providers need a well constructed set of marketing support tools to help the sales effort, including those that track commissions, offer reliable reporting and organizes sales and marketing information. Again, according to Atlantic-ACM, resellers are also looking to

By Suzanne Finch



As competition for customers increases, so does the need to build a broad resale channel sales force.

streamline their service offerings with Web-based sign up, online billing, and online customer service. The requirements of both the wholesaler/provider and the reseller can be fulfilled with new generation operations systems and software (OSS).

Known mostly as billing and provisioning software, many OSS applications have the ability to enable bundled services and offer online signup and activation. Many providers may already have legacy OSS software that has these features built-in, but are not currently using them. Other systems have these capabilities, but require a low-cost module to enable these features.

Once these features are put into play, resellers can take advantage of them through a branded portal. Branded portals map back to the provider, but can display the look and logo of the reseller. A transparently branded portal can generally be configured in a matter of hours and allows the reseller to roll out and sell services within a day. The roll-out process may take a little extra time if the reseller wants to load their entire customer base and pricing structure into the providers OSS platform, however, a Web-based application will allow them to upload their information and make edits from any Web browser.

Oss Application In A Channel Market

In addition to bundling VoIP services such as find me/follow me, call screening, and unified messaging, OSS allows wholesalers/providers the ability to bundle non-VoIP services like the previously mentioned e-mail, domain name registration, streaming video, and others. Most OSS systems can enable bundled services that are either outsourced or internally hosted. This allows resellers to offer end-users an even greater number of services with the specific service bundles that fit their market demand. Also, a greater number of services tend to make the bundle "sticky" in that customers now deal with a single provider for all services rather than different providers for each service.

In addition to enabling service bundles, Web-based OSS offers a convenient way for resellers to sell services that are provisioned and activated online, allowing them to sell 24/7 rather than from just 9 to 5 and without regard to time zones. Online billing also allows end-users the opportunity to view their account from any browser and the OSS platform can be configured to provide up-to-the-minute account information — especially important for users of services with usage sensitive rates or time increments.

Resellers can also take advantage of online customer care features offered by many OSS systems. Features such as online service tickets and customer discussion boards allow end-users the ability to service many problem issues on their own without burdening either the reseller or the wholesaler/provider with commonly asked questions or with uncomplicated maintenance issues.

While OSS brings a wealth of features that the reseller can use to their full advantage, the wholesaler/provider can utilize the commission calculations and reporting elements that allow them to track sales and reseller performance. OSS systems that feature Online Analytical Processing (OLAP) compliance are generally capable of providing this functionality which vastly improves the wholesaler/provider's ability to track data, maintain information and make educated decisions based on reseller activities. OLAP tools provide analysis of data stored in the OSS database and enable users to analyze different segments of multidimensional data which make up the OSS functionality. For example, OLAP can provide time series and trend analysis views of the reseller's sales activities.

Once sales are made, both the reseller and the wholesaler/provider want payment for their services — that's where payment gateways come in. Payment gateway modules may come inherent to the OSS system, but are most often purchased separately. It is the payment gateways that enable end-users to pay for IP services with a credit card and deposit the funds in the provider/wholesaler's account. Payment gateways operate by batching transactions from the OSS and then sending the batched data to the payment processor for verification and authorization. The wholesaler/provider arranges with a banking institution for merchant ID, account number, and a password, all of which is transacted between the payment processor and banking institution. The payment is then deposited into the wholesaler/provider's account. Integrating the OSS with the payment gateway AND automatically calculating commissions makes financial arrangements with resellers a far less painful process.

As VoIP technology gains acceptance and providers become more competitive in the marketplace, the need for a fully Web-enabled OSS system that can enable VoIP features and functionality will become a major factor in ramping services through resellers quickly and effectively. While there are hundreds of OSS systems that enable VoIP services, either finding the right system or utilizing the full functionality of a legacy system is clearly becoming the make or break of a successful channel strategy. **IT**

Suzanne Finch is director of marketing communications at Rodopi Software. For more information, please visit http://www.rodopi.com.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Managing Convergence —

The Next Operational Leap

VoIP is the single biggest thing to happen to enterprise networks since the telephone. IP Telephony will make communications cheaper, easier, and more flexible. Convergence will fundamentally change how we work and how we communicate. In short, VoIP will save your life.

You've probably already heard all of this before, as well as dozens more hackneyed platitudes and sound bites. And you're probably already quite sold on the *importance* of upgrading your network to VoIP (define - news - alert), but you have questions and concerns. And while many of these can be addressed by the many VoIP vendors and system integrators vying for your business, there may be a few they're not so able to answer.

One question, that often comes up toward the end of a deployment: "Once I upgrade my enterprise communications network to VoIP, how will I manage it?" Of course, a better question to ask: "How do I keep management in mind while I plan and execute on my VoIP network upgrade so I don't need to worry about disconnects, poor sounding calls, application failures, and other problems?" Unfortunately, few are very interested in detailed conversations around the need to adequately manage the "easy-to-use" equipment they're trying to sell you.

The truth is that VoIP has already started a major transformation in how we communicate, and things are going to get bumpier before they get easier. Yes, one day you'll plug into the wall and have dial tone, video, data, and more, in much the same way we enjoy electricity and water today, but despite the best efforts by many service providers around the world, that day is a long ways off.

The good news? Your organization can very likely achieve significant cost savings, tangible productivity increases, and move to an underlying technology with almost limitless room for growth and extension — all while accepting modest risk, modest cost, and (relatively) stress-free nights. How? A strong management solution, so that whatever vendor equipment you deploy, no matter its configuration, and no matter which SI you use, you'll know the VoIP application is available, meeting customer needs, and secure. And, if there is

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By Jeremy Bloom

VoIP has already started a major transformation in how we communicate.

a problem, you'll know immediately, and have the tools to drill-down and find both the problem and solution quickly and efficiently — hopefully without end-user impact.

In order to understand what management system must be built, let's first review the five major challenges of VoIP management. Keeping these five challenges in mind, as well as challenges unique to your organization, will help you build the right management solution, whether you build your own, buy commercial off the shelf products, or outsource to a managed service provider.

Challenge 1: Anything Can Break

Chances are quite good that your data network is complex — very complex. You probably have a headquarters, at least several distributed sites, and at least a few telecommuters. And if you're anything like your competition, you've been adding more and more business critical features and functionality to your network, from things as simple as e-mail and instant messaging to sales force automation, workflow applications, and ERP systems. Now you're going to add VoIP, which runs on the same network, but adds a whole new level of application fickleness.

Consider an example where Boris wants to communicate with Natasha. If he sends an e-mail, and due to transient network congestion it takes five minutes to arrive, there's no issue (let's assume for the moment that Natasha is not eagerly awaiting news on Bullwinkle's whereabouts). But now if Boris picks up the IP phone to call Natasha, and a link is temporarily out of service or even just has a considerable delay, they can't connect. Bottom line: the data network you're running today may or may not be ready for VoIP. If you're like most companies, you're going to need to do some upgrading of both equipment and management systems. That's healthy and natural: you didn't need this level of resiliency before, but VoIP needs it now.

Challenge 2: Understanding the Data

Most data equipment can be rather chatty when there's a problem (or even when there isn't!) — from Simple Network Management Protocol (SNMP) traps to database records to Element Manager Systems (EMS) reports to anything else. Sometimes this information can be hard to understand, let alone act upon. Do you really know how much of a problem it is if a certain router is running at an 82 percent CPU usage? Adding VoIP-specific and VoIP-enabled equipment makes this more complex. Consider the simple case of an IP PBX (define - news - alert). A pure IP PBX has probably only been in existence for a few years. Even an IP-enabled PBX has many features (certainly the IP ones) that are relatively young, even by technology standards. In the never-ending quest to get your business, vendors add new features and apply patches all of the time, many of which result in changes to event formats and even meanings. You need a system to understand these events and adapt to changing ones. That doesn't just mean it will work today — but you need to be comfortable that when something changes tomorrow, such as a new feature or an IP PBX being demoted from production to a test server, that you have a way to alter how you act upon its events.

Challenge 3: Understanding the User Perspective

While managing the underlying network components and services is critical, it's all moot if your users experience poor quality calls. The key word here is "experience." If a user picks up the phone and doesn't get dial tone, or makes a call and hears an echo, low volume, or choppy speech, she's going to formulate a negative opinion about the phone service at the time. It doesn't matter that there were no problems reported on the network, that twenty other users made calls at the same time with no issues, or that someone was downloading several new mp3s for their iPod and caused network congestion.

If you're lucky, she'll complain, so you can log a ticket and research the error. More likely, she'll just form a negative opinion of the service, use it less (perhaps using her mobile phone instead), and not complain until she's experienced a number of problems with a clear and demonstrable effect on productivity. Only then will you get a complaint, and it won't usually have exact times and dates attached as much as it will have an understandable dissatisfaction and unhappiness with the service.

One way to handle this is to encourage users to report bad calls, perhaps even with a simple to use Web form. An even better way: leverage technology that allows actual monitoring of user calls, or at least a representation of them — this is generally called "passive monitoring." Related and also quite useful, "active monitoring," allows synthetic calls to be placed on a network to generate periodic and on-demand calls as well. A number of vendors have passive and/or active monitoring tools available today, and most are evolving these technologies to work even better. Virtually all allow this information to be passed "northbound" so it can be viewed in the context of other network information.

Challenge 4: Correlating User Experience with Network Information

So you're collecting data from all the equipment on the network, as well as actual service quality data from both your SQM management tools and your actual users. Now, how do you get this to work together, so you can quickly resolve issues, preferably before they become major? The first step is to make sure all of your information is going to a central system, so that it can be analyzed in the context of all the available information. Next, you need to start prioritizing events so you know where to focus first. Even if the supply closet's

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phone has failed three times over the last week, even a single failure in the receptionist's phone can be much more disruptive for business operations. Finally, you need to do the hard part: start looking for patterns of failures, especially between user reported or SQM software reported events and actual network problems.

For example, you may have a user who likes to download mp3s over his lunch hour. A misconfiguration on your network may result in those mp3s receiving the same priority as VoIP calls, which can result in network congestion. This congestion will be reported in SNMP (define - news - alert) traps and Syslog messages, and can be used to help further the diagnosis.

While this can be done by hand, it's a lot easier to use off the shelf software to help automate many of these problems. Some of these packages will do totally independent automation, others will allow arbitrary service correlation and modeling (you map out the service it is to manage), and the best will do combinations. Also look for strong visualization capability, so that the entire network and all important services can be viewed in multiple ways. For instance, sometimes you'll want a layers 1–3 view, other times you'll want a layer 7 application view, and other times you'll want something else. Remember that no single paradigm will find every problem, so look for software that has a good variety of capabilities.

Challenge 5: Baselining and Performance Management

Once your VoIP network is running optimally, you need to baseline it's performance, use trending tools for capacity planning, and compare the past results to current results when problems occur. A strong performance management solution is critical here — whether you build your own or buy off the shelf. While your VoIP deployment may be complete, you'll likely be deploying extra phone features, unified messaging capabilities, and more IP-based services that will increase both the bandwidth utilization and the complexity of your network. By tracking the network over time, you can determine needs before they become critical, such as a new IP PBX, increasingly low disk space in a voice mail server, or an increasingly slow backup cycle in your customer database.

With all the complexity of managing a VoIP network, many enterprises are looking to outsource their operations. This is especially true of companies that are already comfortable with outsourcing, or are small enough that they simply can't afford to have a dedicated technical team. This approach can be very beneficial, as it allows both the enterprise and the managed service provider (MSP) to focus on their core competencies. Clearly, there are a lot of factors to consider when picking an MSP. Many enterprises look for the safety and security of a larger MSP, while others like the personal and customized services of a small MSP, perhaps even leveraging the systems integrator they may have used to help with the original installation. Whoever you choose, make sure that you're both working with someone you trust as well and you establish clear Service Level Agreements so that if problems occur the MSP will feel the pain almost as much as you do. If problems occur, you want a rapid fix, not finger pointing and denial. Also, don't be shy to ask for a hybrid environment, where perhaps you'll manage some of the equipment but they're responsible for the remainder as well as the total service. Most MSPs are quite eager for your business, and are willing to work within your constraints to get you what you need.

For those that choose to manage their own VoIP deployments, either exclusively or with an MSP assisting, the right management tools are critical. As is often the case in technology, no single vendor will have every solution you need, but the best will have most of the right components and have no trouble working with your other best-of-breed selections. You need to build the right network management solution both for today and the future — you certainly don't want to train your team on one solution and replace it in one year. As is always the case with vendor selection, you'll need to find the right blend of several factors, including innovation and stability. Sometimes you'll want the safety of a large vendor, other times you'll require the innovative approach of a smaller one. Whatever direction you choose, look for good technology and a strong customer base as prerequisites.

Finally, remember that your network will almost certainly grow quickly and dramatically, even if your core business is only experiencing modest growth. Metcalfe's Law states that the value of a network increases exponentially with the number of connections. Every time you add a user, a new application, or a new service, you're adding complexity. And, unfortunately, every time you add network complexity, management complexity also increases. With the right planning, people, and tools, you can minimize that complexity and focus your efforts on building the right applications and services for your user base, rather than on trying to determine what failed and why. **IT**

Jeremy Bloom is Director of Product Management at Micromuse, a provider of scalable, real-time business and service assurance software solutions. For more information, please visit http://www.micromuse.com.

Critical IP Infrastructure Issues:

Navigating VoIP From Fad To Mainstream

The move to Internet protocol (IP)-based communications, including voice over IP (VoIP), has accelerated existing pressure on voice revenues for communications service providers. To date, much of the attention has been on the opportunity to generate innovative new services using SIP, presence, and enhanced mobility capabilities as a way to build new revenue streams.

Several important, and often overlooked, infrastructure issues that deserve attention and are critical to the mainstream success of VoIP (define - news - alert) include:

• Enabling IP-to-IP interoperability while preserving full PSTN (define - news - alert) interoperability;

• Implementing new provisioning solutions optimized for VoIP services in the face of declining voice margins while continuing to support a parallel PSTN infrastructure; and

• Ensuring VoIP security. These issues must be addressed for VoIP services to become mainstream and realize their revenue-generating potential.

The fast-changing industry landscape further complicates these issues. For example, next-generation service providers like Vonage break the telephony service provider mold; this is only getting more interesting with parties like AOL (quote- news- alert) and Earthlink (quote- news- alert) entering the voice market. These new models compound the already complex problem of seamless migration from, and interoperability between, the circuit switched world and the evolving IP world. Additionally, cable companies are looking to bite off a piece of the VoIP pie, introducing yet another layer of complexity.

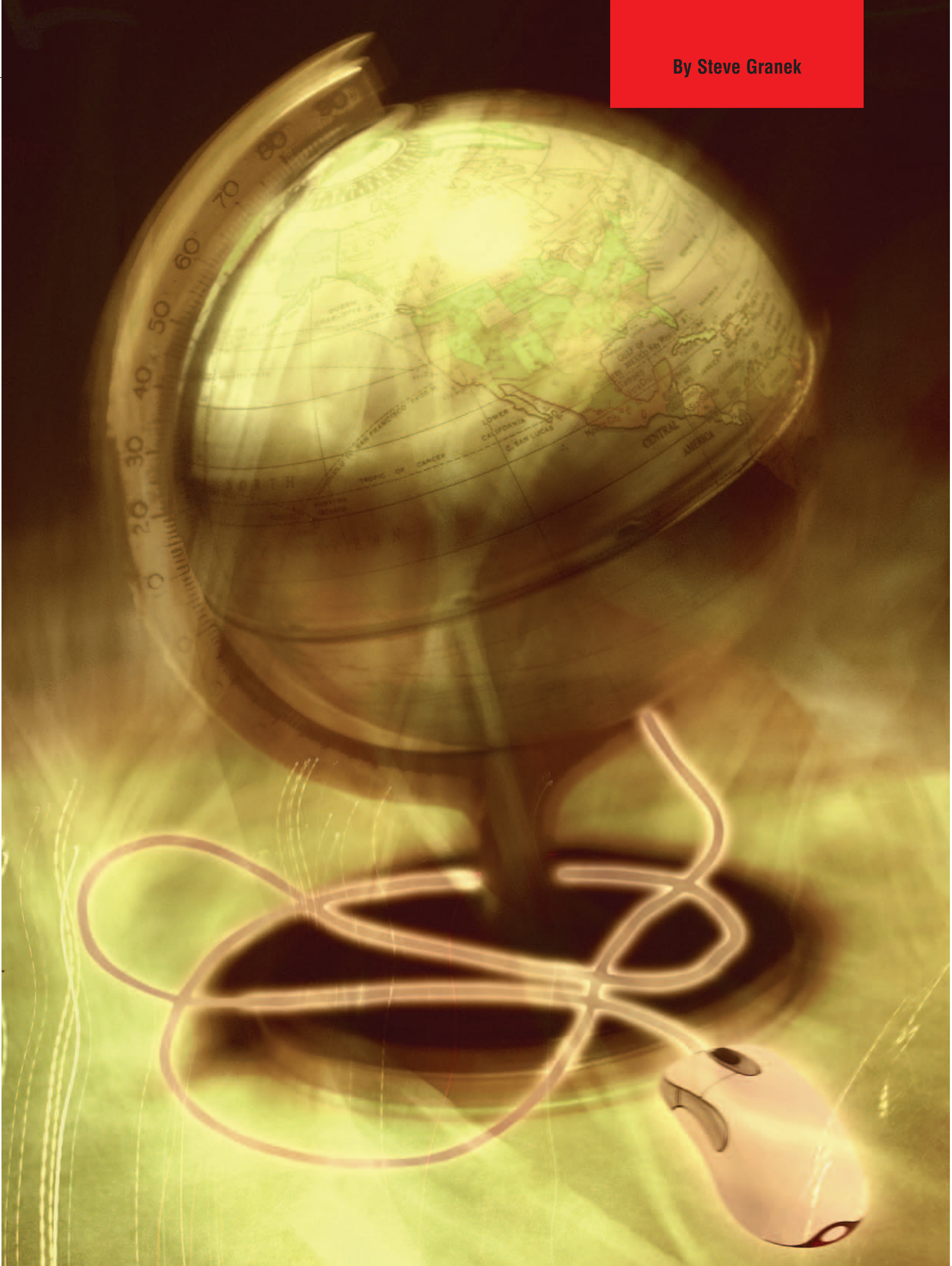
If reliable, affordable, high-quality VoIP services are to emerge — and if service providers are to generate revenues as a result — the industry must address these fundamental issues in a comprehensive, impartial manner.

Many Dimensions Of Interoperability

An important fact of life in the communications world is that communications economics rely on the network effect — that is, communications services increase in value exponentially as the number of parties that can be reached increases. Consider that we once lived in a world full of disconnected, incompatible e-mail systems with names like GroupWise, PROFS, cc:Mail, and MCI mail. E-mail only really took off with the explosion of the Internet and SMTP, which assured a sender that an e-mail would reach the intended target.

Today's PSTN is similar to the current state of e-mail. Dial a number, and you can assume the intended target will be reached, whether a customer of Verizon, Sprint, Cingular, or British Telecom. Now contrast this with VoIP. Each VoIP network is an island, glued together by the PSTN. One reason for this lies with naming and addressing, the often-overlooked linchpin of any network. Names are memorable strings of numbers or words, which consumers use to make phone calls, send e-mail or access Web pages. The PSTN uses telephone numbers for names and the Internet uses domain names. Domain

By Steve Granek



names are used to create Uniform Resource Identifiers (URI) like an e-mail address. Addresses are often lengthy strings of numbers that are used by networks to identify other network elements. Addresses used in the PSTN include SS7 addresses and addresses on the Internet are IP addresses. Networks use names to derive addresses.

The industry has rallied around the IETF ENUM (define - news - alert) standard, based on DNS, to derive an Internet name from a telephone number. ENUM will help unify the naming and addressing between the PSTN and VoIP networks, and between different VoIP networks.

One challenge around the practical deployment of ENUM for telephony is the notion that it should be implemented as a publicly accessible DNS registry like .biz or .com. First, ENUM duplicated an existing "name space" — the e164 numbering plan — with matching domain names. While seemingly straightforward, this process is actually very complex due to the imperative that there should be a unified authority for both the telephone number and the related ENUM domain. The alternative would be routing chaos in which, depending on what network you are on, you could reach many different people with the same address. This problem is especially tricky in places where number portability exists, where tens of thousands of telephone numbers change service providers everyday. How could a publicly accessible ENUM registry keep track of who has authority over the telephone number?

Second, putting carrier information in a public DNS database creates privacy and security risks. While most groups that are planning the deployment of ENUM are trying to avoid these problems, there are some difficult technical issues to resolve as well as complex policy issues from in country and international regulators. To work around these issues, carriers and enterprises are developing "private" databases using ENUM technology for routing between IP endpoints. Methods of interoperability between these private ENUM environments and public ENUM, when it comes to pass, will be important.

It is also worth pointing out that the PSTN and Internet have evolved using different models of inter-party compensation. Business model interoperability becomes even more complex when one considers the economic dependence of many rural carriers on compensation models that include payment for termination. These issues have yet to be adequately discussed, and the dialogue will need leadership with an eye to the greater good from all segments of the industry.

The Provisioning Process

As previously mentioned, next-generation VoIP service providers are challenging the existing provisioning processes associated with number acquisition, management and portability. To date, these newer players have managed these traditional PSTN processes through wholesale relationships with CLECs, who in turn typically rely on manual and, in some cases, semi-automated systems to interact with ILECs.

Presently, customer order requests — for initial service activation, porting in, and transferring in phone numbers — require providers to submit orders manually, via e-mail or fax, to CLECs. If the request is for a ported-in telephone number, the CLECs must interface with other carriers to complete the process, which can be cumbersome and time-consuming.

For VoIP service providers, the resulting delay in service activation and lack of visibility into the process can result in errors, dissatisfied customers, missed revenue opportunities, and an increased risk for customer churn or deactivation. To enable these new business models to scale, this manual process needs to change.

Fortunately, the industry has recently made progress towards enabling full flow-through provisioning; all the way out to the VoIP service providers' customer care representatives. Improvements such as these will help

Communications services increase in value exponentially as the number of parties that can be reached increases.

streamline customer acquisition; increase customer satisfaction, and accelerate revenue realization.

Service providers invest precious capital in software, hardware, and personnel for new VoIP equipment and support systems, and the result can be an unsustainable jump in OPEX and CAPEX in the face of intense price pressure on basic voice services. For many, this happens while they must still support an existing parallel PSTN infrastructure that will not work for VoIP. In addition, these service providers often discover that their initial investments do not match emerging market imperatives. For example, a service profile that is designed for simple consumer voice services will require enormous additional investment to also support business multimedia. ROI can be an unattainable goal — in an unforgiving industry environment. This situation begs for new approaches. One alternative is starting to gain traction with immediate bottom-line results: a clearinghouse service that enables new service creation without requiring expensive infrastructure investments and their related timelines.

VoIP And Security

Security is the soft underbelly of IP networks. While there are many reasons for this, one prominent security threat is the ease by which people can assume false personas and impersonate others in IP environments. VoIP services are not immune, and several scams have been reported recently in which "bad guys" fraudulently "spoof" others' identities in SIP to gain access to networks, send SPAM, commit identity theft, and distribute malware.

SIP (define - news - alert) identity standards have been released to deal with these transgressions. It is imperative that software and hardware vendors implement support for SIP Identity sooner rather than later. VoIP can ill afford to develop a reputation in the market as the "voice cousin" of e-mail when it comes to security and privacy. It will require a concerted industry effort toward broad adoption of SIP Identity standards to avoid this fate, and solutions will likely require new services for independent "real-time" identity verification.

Also, it is important to realize the trust models for VoIP, even in a carrier context, are fundamentally different than those of the PSTN, because in VoIP, many more parties are true peers. The most obvious example of this is an enterprise with an IP-PBX. At first blush, it appears that a specialty firewall/network address translation device, like a session border controller, is adequate for securing the carrier's network (and for that matter, the enterprise's network). But that assumes the problem lies in trusting the network directly adjacent to yours. Closer examination shows this to be a false sense of security, since there is really no way to know the true source origins of traffic entering a carrier's network. So, while border elements form part of the solution, they are by themselves, inadequate. SIP Identity remains critical. This is the reality of VoIP.

Next Steps

VoIP is clearly coming to the mainstream. It promises creative ways to bring new personalized services to customers both businesses and consumers. While slick demonstrations of new technology are always exciting, mainstream adoption will require complex infrastructure processes to be efficient, both inside of and between service providers. Resolving the problems of inter-party interaction in the PSTN has proven a challenging and long process, with much inefficiency still rampant. It is in the best interest of all parties to avoid using that process as a roadmap for resolving VoIP interoperability, security, and provisioning efficiencies. This challenge that will take time, but it can be addressed and solved through the collective work of the industry at large including service providers, vendors, standards bodies, and the regulatory community. **IT**

Steve Granek is vice president, Advance Services at NeuStar. For more information, please visit the company online at http://www.neustar.com.

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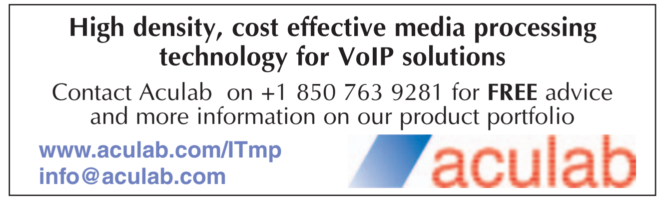
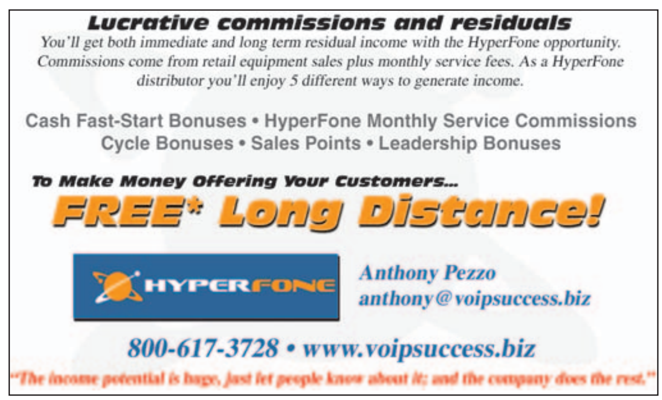
VOIP MARKETPLACE





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VoIP MARKETPLACE

CEO Spotlight

Zee Hakimoglu
President and CEO
ClearOne Communcations

In the CEO Spotlight section in *Internet Telephony*® magazine, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Zee Hakimoglu, president and CEO of ClearOne Communcations.

GG: What is ClearOne's Mission?

ZH: ClearOne's (news - alert) mission is to deliver premium audio conferencing products to the world. In today's competitive global environment, group meetings between geographically separated participants are an essential part of doing business. Audio is mission critical. Regardless of whether the meeting is conducted through video conference, telephone, or the Web, audio is integral to effective conferencing. After all, even if you can see who you are talking to, you won't be able to communicate if you can't hear what they are saying.

Companies are looking for ways to be more productive, and group meetings are an area that can be easily improved. Our products are developed to enable meetings between distant participants that are as effective and productive as if everyone were in the same conference room.

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

ZH: My vision is that the name ClearOne becomes synonymous with audio conferencing and is the first company that comes to mind when deciding on products for a conferencing space. We have supplied audio equipment to Fortune 500 companies for more than 20 years and are fortunate to have a very strong foundation in developing high-end, installed audio conferencing products. In fact, we are the worldwide leader in this space. Building upon this position should bode well for us in the next-generation telecom market. As enterprises make the switch to VoIP (define - news - alert), they are forced to evaluate all of their telecom needs, including products for group communication from the executive office to the boardroom to the large corporate training room or auditorium. By leveraging our technology expertise and market leadership, we can tap into new opportunities being created by the increased adoption of VoIP.

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

ZH: Quality and innovation are two areas that I believe truly separate ClearOne from the competition. Throughout our 20-year history, we have focused on developing the highest quality products possible, and our market leadership in the high-end installed space validates that. ClearOne has also earned a reputation for unrivaled customer service and support, which I believe is every bit as important as the quality of the product itself. This dedication to customers extends to not just the end users of our products, but to our channel partners as well, who we view as the lifeblood of our company.

At the same time, we have not let product innovation take a back seat to quality. We developed many truly innovative products, including the industry's first audio conferencing system to use Distributed Echo Cancellation, the industry's first wireless conference phone, the industry's first fully expandable conference phone that daisy-chains multiple phone units, and the industry's first product to fill the wide price/performance gap that existed between plug and play tabletop conference phones and professionally installed audio conferencing systems.

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?

ZH: The biggest hurdle in my mind is that standards are not yet solidified. With the tremendous amount of competition and growth in the VoIP industry, many new companies are developing unique solutions that create interoperability problems and ultimately slow down adoption.

> The biggest hurdle is that standards are not yet solidified.

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GG: What are some of the technology areas where ClearOne is increasingly focusing, and why are these areas so important to the future of your company?

> I believe the industry will continue to see increased competition from outside of the traditional telecom markets.

ZH: We are focused on developing technologies that drive adoption of conferencing products across a diverse customer base. Improving ease of use across all of our product lines is a significant area of focus, as is implementing and improving the ability to control conferencing products on the network, and development of VoIP-based products. In addition, we continue to focus on advanced technologies that will further enrich the audio experience for users of conferencing products.

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

ZH: I expect to continue to see a tighter integration between computers, smart devices and telephony, as well as increasing connectivity and access through a much wider range of devices. Because of this, I believe the IP telephony industry will continue to see increased competition from outside of the traditional telecom markets. For example, network and computer companies are becoming heavily involved in telecom, cable companies are providing voice services, and traditional voice companies are providing data services. From a competitive standpoint, there is no single area for companies to monitor. Rather, they need to be aware of the total technology landscape.

I also expect to see continued convergence of voice, data, and video. IP is the key enabler for this convergence, as it not only creates cost savings, but more importantly offers benefits of increased functionality and productivity. This convergence is inevitably headed toward the desktop, which for the conferencing industry means that the complex room video conferencing systems that are often just gathering dust will be replaced by desktop systems that are as easy to use as your PC or telephone. IT

CEO Spotlight

Eli Borodow
CEO and co-founder
Telephony@Work

In the CEO Spotlight section in *Internet Telephony*® magazine, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Eli Borodow, CEO and co-founder of Telephony@Work.

GG: What is your vision for Telephony@Work and how is the company positioned in the IP Contact Center market?

EB: Telephony@Work (news - alert) was founded in 1998 with the mission of empowering businesses to deliver world-class service on the phone, fax, and Internet without the costs, risks, and delays of traditional technology deployment paradigms.

We originally developed our multimedia IP Contact Center technology to empower large service providers and multi-site enterprises to centralize contact center infrastructure to support diverse business units across locations on common infrastructure — without sacrificing privacy or local autonomy. This has become known as "multi-tenant" technology and we're the dominant player in that space. That's probably why we're so closely associated with large-scale deployments at such companies as MCI (quote - news - alert), TELUS (quote - news - alert), Siebel (quote - news - alert), ABN AMRO (quote - news - alert), and so many others. While we dominate in the multi-tenant arena, we also fit very well in single-site contact centers. Our technology can scale down very cost-effectively to address contact center needs across the spectrum of opportunities.

Over time we've also been widely recognized as the leader in "adaptive" IP Contact Center technology, which enables companies to easily modify any technology-driven business process or routing rule in real-time, at no cost. This lets companies 'fix' strained technology-driven business processes in real-time to maximize both efficiency and revenues. Companies that have implemented our technology and embraced a philosophy of change have seen tremendous productivity gains. Larger companies have seen productivity gains of as much as 30 percent while smaller companies, typically lacking in-house expertise and consulting budgets, have seen even greater gains.

This philosophy of business empowerment is core to everything we do.

GG: What are the problems that Telephony@Work's IP Contact Center technology was designed to solve?

EB: Our technology was designed from the ground up to empower companies to fully leverage the benefits of IP-based contact center technology and eliminate the core challenges associated with traditional technology deployment paradigms.

Traditional technology deployment approaches have been tremendously expensive, complex and time-consuming. That's because they require integrators to "recreate the wheel" for every deployment — by integrating as many as 24 separate technology point solutions to deliver a complete solution. The problems we're solving emerged because of the way that contact centers have historically been deployed; with each new emerging technology being 'bolted' onto the legacy infrastructure via custom integration. Not surprisingly, mainstream business publications, analysts, and academic research consistently report that 60–70 percent of contact center deployments never achieve their stated objectives due to complexity and budgetary restrictions.

Systems provisioned in this model are also notoriously rigid; requiring a lot of time and money to implement changes over time — because of the multitude of different integration points, administration consoles, and programming envi-

> The philosophy of business empowerment is core to everything we do.

ronments involved in maintaining traditional solutions and implementing changes. Regression testing following such changes is also expensive and time-consuming. This is why most companies forklift their legacy contact center technology investments every five years — because the increasing delta between what was originally provisioned and current needs becomes unsustainable over time.

For managers who are compensated based on performance, proposing reliance on a remote IT staff is a hard sell.

Disjointed business processes across diverse systems is another consequence of the traditional point solution approach; because the same routing and customer priority business rules can't be easily applied across all communications channels. In addition, because the various components aren't integrated-by-design, there are multiple databases associated with different parts of the provisioned solution, providing only a limited view of data and inhibiting the consolidation of customer data in real-time.

As companies have begun migrating to IP-based solutions, traditional vendors have taken the approach of adding yet another layer of complexity by selling yet another point solution to implement voice-over-IP in the contact center. The downside of this approach is that it neutralizes a number of the core benefits empowered by IP-based transport. For example, they can't centralize technology resources without sacrificing local autonomy.

This is a huge issue. There are tremendous inefficiencies associated with the 'traditional' approach of deploying and maintaining diverse contact center systems at every location. These inefficiencies include duplication of systems and licenses at each site, shortages of software licenses at some locations while the needed resources sit idle at other locations, and the duplication of staff required to maintain each set of systems at every location. IP contact center technology can eliminate these inefficiencies by empowering companies to centralize technology resources and leverage them across a global network, thereby dramatically reducing technology costs across all locations. That said, extending 'traditional' site-specific point solutions to support other locations requires those remote locations to entirely give up their autonomy over their own technology-driven business processes. For managers who are compensated based on performance, proposing reliance on a remote IT staff is a hard sell. That's why most companies that have extended their legacy solutions never realize the benefits of enterprise unification. Vendors that sell technology on this basis typically tout savings resulting from reduced transport costs, but that's really a marginal benefit when compared to the compelling productivity gains that come with cross-location enterprise unification.

GG: How does Telephony@Work IP Contact Center technology address these challenges in ways that are different from traditional approaches?

EB: Telephony@Work's CallCenterAnywhere technology was designed to empower companies to dramatically reduce costs by centralizing technology resources to support diverse locations — while preserving and enhancing local autonomy. Of course, we also had to address all of the manageability, scalability, reliability, and network security issues required for centralized technology to become a viable alternative to system and staff replication at every autonomous site.

Our technology was also designed to enable companies to adapt to changing business conditions in ways that would have seemed impossible just a few years ago. Our award-winning CallCenterAnywhere solution delivers all of the multimedia IP Contact Center technologies needed to run a world-class contact center — while empowering real-time, no-cost business process modification. To achieve this, we looked at all of the traditional needs-analysis questions that integrators use to define 'scope-of work' for their contact center customers. We then spent years translating those needs-analysis questions into Web-based menus, drop-down lists, and radio buttons in a unified, browser-based provisioning interface. The benefits of this approach include dramatically reduced provisioning costs and time-to-market, a huge reduction of cost-of-ownership expenses and the revenue-generating benefits of real-time, no-cost modifications to all technology-driven business processes.

The key, of course, was to pre-program all of the potential outcomes in an architecture that would enable changes to be implemented in real-time — at no cost. This also empowers managers to more meaningfully contribute to performance through knowledge of best practices and the agility to implement routing and business process changes in real-time.

That's not to say that we deliver a 'black box' — in fact, our technology is built on an open architecture on top of Web services. That enables our customers to extend our applications in ways that we might never have imagined.

Once we achieved our original objective of 'productizing' the spectrum of 'traditional' business process configurations across the various media channels, our focus shifted to finding new ways of leveraging our integration-by-design approach. Today we deliver efficiency and value that go far beyond the limits of traditional 'integration-heavy' solutions — value that could never delivered in any reasonable amount of time in the traditional systems-integration paradigm. That ROI-focused, out-of-the box thinking is another reason we've had so much success over the last few years. IT

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