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INTERNET TELEPHONY

www.itmag.com VOLUME 8/NUMBER 10 OCTOBER 2005

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OUR BIGGEST ISSUE YET

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- Inter-Tel's CEO Rauchle on the Future of VoIP
- Rich Tehrani's Executive Suite Debuts
- CT Labs Reviews PingTel's SIPxchange

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

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Annual digital subscriptions to **INTERNET TELEPHONY**®: free to qualifying U.S., Canada and foreign subscribers.
Annual print subscriptions to **INTERNET TELEPHONY**®: free, U.S. qualifying readers; $29.00 U.S. nonqualifying, $39.00 Canada, $60.00, foreign qualifying and nonqualifying. All orders are payable in advance in U.S. dollars drawn against a U.S. bank. Connecticut residents add applicable sales tax. For more information, contact our Web site at www.itmag.com or call 203-852-6800.

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IDENTIFICATION STATEMENT
INTERNET TELEPHONY® magazine (ISSN: 1098-0008) is published monthly by Technology Marketing Corporation, One Technology Plaza, Norwalk, CT 06854 U.S.A. This issue, Volume 8, Number 10 is dated October 2005. Annual print subscriptions: free, U.S. qualifying readers; $29.00 U.S. nonqualifying, $39.00 Canada, $60.00, foreign qualifying and nonqualifying. Periodical postage paid at Norwalk, CT and at additional mailing offices. Postmaster: Send address changes to: **INTERNET TELEPHONY**®, Technology Marketing Corporation, PO Box 21642, St. Paul MN 55121 U.S.A.

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The VoIP Authority

By Greg Galitzine

VoIP 2005:
The Right Time, The Right Place

Life is good. Especially if your name is Niklas Zennström. eBay's Meg Whitman and her team of advisors decided it was time to buy their way into the hottest game in town (VoIP) and they accomplished just that by shelling out up to $4.1 Billion for Skype. Time will tell if they made the right choice, and whether or not they were able to haggle for the best possible price. I've written extensively in my blog (http://voip-blog.tmcnet.com/blog/greg-galitzine) about my less than positive view of the transaction so I won't go into that here, but I'm looking forward to hearing Mr. Zennström speak at the Internet Telephony Conference & EXPO at the end of this month. Perhaps he can share some useful insight regarding the acquisition.

The eBay/Skype merger is just one of the high points of this new season of VoIP (define - news - alert). Persistent rumors of a Vonage IPO, the launch of GoogleTalk, Microsoft's acquisition of Teleo, the list of major players getting into the VoIP game… all says something about the staying power of a technology not too long ago derided as a hobbyist's plaything, useful only for saving a penny or two while giving up way too much in the quality department.

Believe me I've heard it all. We've been publishing this magazine since 1998 and I was writing about Internet telephony back in CTI magazine for several years before that. Our industry has weathered a number of storms and we've been here all along watching, listening, reporting. Take it from me; October 2005 is a great time to be here and to see all of this buzz and activity in our industry. It is certainly the right time to be involved with VoIP.

And amid all this awesome activity in our space stands the aforementioned Internet Telephony Conference & EXPO. This year's event is gearing up to be the largest ever in the VoIP space, with attendance projected at over 7,500. There's simply no better place to go and get educated on what you need to know if you want to build or deploy or resell VoIP. The conference program is truly unparalleled. And the Exhibit Hall at the Los Angeles Convention Center will be full of the industry's leading vendors promoting their latest VoIP products and services.

Back when it all began, who would have thought that the world's largest ISP, the world's largest software company, the world's leading Web site, and the world's largest networking company would all be in one place addressing the audience on the subject VoIP?

This brings us to the "show issue," which you hold in your hands. It's our biggest issue yet! And while size is relative, rest assured that this magazine has never been so full of pertinent news and information that you — as a reader and a consumer of knowledge — absolutely need to help you make the important decisions regarding this space.

For our loyal subscribers, it's so nice to welcome you each and every month and let me say how thrilled I am to greet you again. Hopefully, I'll see you at the event.

And, for those of you who may have just recently picked up a copy at the show, I urge you to look through this issue and then go online and subscribe. Make sure you don't miss this wonderful opportunity to receive the industry's leading magazine. And if I haven't yet done so, may I welcome you to Los Angeles. This week at least, it's the right place to be for VoIP.

-Greg Galitzine, ggalitzine@tmcnet.com

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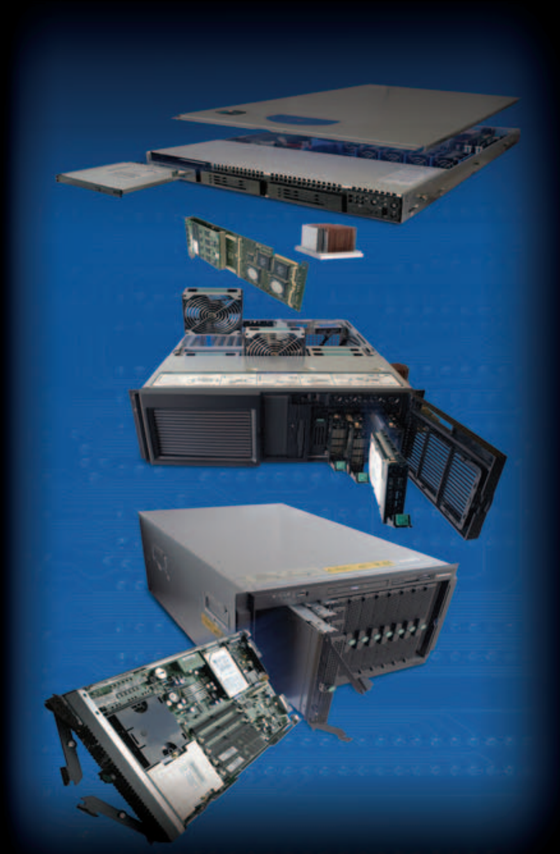

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10. Colorado

QUOTE OF THE MONTH:

"Voice is the next phase of the Internet. We will voice-enable anything and everything. We will do this because voice is the most natural interface we have and we will do this because there is money to be made by doing it. Most importantly, after eBay voice-enables every product and service they have, we will do it to be competitive and to meet customer demand."

– Rich Tehrani (page 8)

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 5.9 million unique page views per month, translating into over 617,000 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.

VoIP, Cost Reduction Driving Fixed-Mobile Convergence
The VoIP industry and cost reduction are the two major drivers behind the fixed- mobile convergence (FMC) market, where a consumer has dual-use, home/mobile phone, according to a new ABI Research study. http://tmcnet.com/175.1

IT and VoIP Relieving Katrina's Aftermath
Front Range Internet Inc. of Colorado joined a growing group of technology companies trying to help hurricane Katrina's evacuees and rescue personnel by providing means of communication. The Internet service provider will offer Internet access and phone services to allow evacuees to get in touch with family members and rescue personnel to quickly access urgently needed information. http://tmcnet.com/176.1

Hopelessly Devoted: A Customer Communications Renaissance
Communications between customer-facing companies and their customers in the best possible manner — in efficiency, accuracy, speed, flexibility and response opportunity — is at the very core of what technology providers in the customer service and communications industry must aim for. Companies often forget or completely disregard this fact. Why? Let's call it corporate ADD (attention-deficit disorder. http://tmcnet.com/177.1

Type Quietly. Someone May Be Listening.
Need more news to make you paranoid about personal data security? Someone may be listening to you type. Say, what? According to researchers at UC Berkeley, the distinctive sounds made by typing on computer keyboards can be decoded, allowing security breaches caused by what the researchers call "acoustic snooping." Scientists said they have been able to take sound recordings of typing on keyboards and run them through a computer, using a special algorithm to retrieve as much as 96 percent of the typed characters entered. http://tmcnet.com/178.1

Datamonitor Predicts U.S. Outbound Market Will Shrink
U.S. contact center agent positions are in trouble, according to new research by independent market analysis firm Datamonitor. This conclusion was drawn from the analyst group's report entitled, "Contact Center Outsourcing in the United States." http://tmcnet.com/179.1

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By Rich Tehrani

Our Biggest Issue Yet!

(There's just so much to cover...)

Having written about VoIP since the mid-late nineties and in this publication since 1998, I can tell you something amazing is happening in the market. The energy level keeps rising. This is our biggest issue ever and for that I thank you the readers for loyally reading this magazine through good times and bad and responding to the advertisements you have seen in these pages over the years. Whenever I come into contact with readers that have been with us since 1998 they always tell me they knew that VoIP would one day be here. They thank us for being the only publication consistently covering the space over the years. To all of you I say you're welcome. I am of course also speaking on behalf of Editorial Director Greg Galitzine, and TMC CTO and head of TMC Labs, Tom Keating.

To continue, I have some great news to share with you. Today is the big day. VoIP is here. There are a number of reasons that lead me to believe we have reached an amazing threshold and are beginning to pass it. Perhaps the most shining example is the fact, that as I write this article, eBay announced they are purchasing Skype Technologies. Couple this with the recent news that Vonage is filing for an IPO.

News of acquisitions and IPOs were commonplace in the technology market in the 1990s but recently we haven't seen so many. This is why it is even more amazing to me that we are once again seeing multi-billion dollar acquisitions for companies that don't make much money.

Which brings up a valid point: the industry's biggest challenge is to determine how to make money. Is Vonage making money? We will find out soon. What is the company's plan for generating revenue tomorrow? This is a more pertinent question. There seems no more appropriate time to discuss what is next for VoIP (define - news - alert) than today, perhaps the VoIP market's biggest day. In just a few short years, VoIP has gone from a technology that was not too well-known to a household word. VoIP today reminds me of ATM machines. A few years after they launched them, even your grandmother was using one.

But is usage by the elderly a barometer of success? In and of itself, perhaps not. But there is something exciting happening. Once people get comfortable with VoIP and start using it they realize they can do things they couldn't do before. People are listening to voice mail on their computers, they are using a GUI to manage their call forwarding, and they are using their telephones anywhere there is Internet access. After a few decades where the biggest advancement was Caller-ID and touch-tone dialing, people are noticing a change for the better.

WiFi Telephony

The population is slowly discovering the fact that they can use VoIP on a wireless network and use WiFi telephony to do all the amazing things they can do with landline VoIP and more. Service providers are seeing that WiFi telephony and dual mode phones will allow them to make more money. Of course we are early in the technology's infancy but I predict soon WiFi telephony will be advocated by cell phone providers.

Pay Per Call

There has been a tremendous amount of speculation documented on places like my blog at Tehrani.com that VoIP is going to be used by companies such as Google to allow pay-per-call advertising. Google makes a nice living selling ads on a pay per click revenue model today. Using VoIP, they can charge advertisers to make their phone ring with a customer call. Google has even started to experiment with print ads where they sell advertisers business card size ads that can have Google managed phone numbers. Google acts as a middleman and is able to measure advertising effectiveness in this manner.

Triple Play

Verizon (quote - news - alert) is rolling out FIOS, its much-anticipated fiber-to-the-home solution allowing customers to take advantage of voice, video, and data from a non-cable company. Bundling services is something we will see more and more of. You don't need to own a data pipe to be able to stream video and radio to customers. I expect to see Vonage, Skype, and others soon doing exactly this.

WiMAX

Vonage (news - alert) has also been experimenting with WiMAX as a way to bypass the pipes owned by the cable/ILEC duopoly. In order for WiMAX to work, there needs to be affordable licensed spectrum and the unlicensed spectrum must not get too clogged with towers. It is early in this technology's lifecycle but for now we can say WiMAX will do exceedingly well in rural areas; metropolitan areas are a bit of a question mark.

BPL

Broadband over power line can revolutionize how we can communicate as the FTC has hinted or it may just be a dud. It is perhaps too soon to say, but if it does work, imagine being able to plug a WiFi access point into a wall and immediately having a hosted, wireless PBX at your disposal. An MIS department might be able to just plug access points around the corners of an office and have full data and voice connectivity. Wiring closets may soon go the way of the mainframe. In fact you'll be able to put your old mainframe in the wiring closet if you still have a sentimental attachment.

Free WiFi

Google (quote - news - alert) has been dabbling in providing free Internet access via WiFi hotspots. They are experimenting with the ability to show ads to WiFi users based on their physical location. Imagine a movie theater advertising its movies on Saturday night to WiFi users within 25 miles of the theatre. How about a restaurant that has a lighter than usual lunch crowd quickly getting ads out to everyone within walking distance in Manhattan?

If the concept works, we may expect Google to be responsible for widespread adoption of free WiFi in the country and beyond. The company is flush with cash, they can do anything they want right now.

Monitoring To The Max

VoIP makes it exceedingly easy to monitor and record phone calls. One of the problems with Skype is that corporations can't monitor its use. Expect as part of eBay's strategic plan to roll out an enterprise-friendly VoIP service that allows monitoring. Eventually Skype can even offer a service that listens into calls and scans for keywords. Most corporations would like to ensure there is no profanity spoken over the phones at work. Others need to record and monitor calls for Sarbanes Oxley or HIPAA compliance. Call centers will need to monitor their agent's conversations for profanity and the mention of a competitor's name. Google, eBay, and other VoIP providers are in a perfect position to do this for a fee.

Just In Time Communications

This market is really on fire — I describe JiTC as the ultimate productivity booster… a set of technologies squeezing inefficiency out of communications. I am seeing collaboration tools coming online from many companies. Some are PBX providers such as Avaya and Inter-Tel, and others are software companies like Oracle and Microsoft. There is a bright future in helping people communicate more effectively.

Development Tools

It seems that ATCA is the new development platform for VoIP applications and Intel and others are pushing this platform as the perfect place to develop your products, whether they are softswitches or session border controllers. By going to ATCA, companies can save money on proprietary buses and allow Intel to share in their development efforts.

HMP is gaining steam but is still not the best way to go for large-scale service provider applications. In time HMP will challenge DSP resource boards but many in the industry representing the board makers are eager to point out that the more sophisticated host processors get, the more we need them to do. This means that the high-end market will always need DSP resource boards.

Skype Voice Service

This could have fit in someplace above, but it is such a big idea that I decided to put it in its own category. By teaming up with VoiceXML hosters and platform makers, Skype allows any voice developer to make applications that can be accessed by Skype users worldwide. If you have a horoscope application that works in ten languages, today, you need banks of 800 numbers in different countries. Not anymore. Now you use Skype and everyone has access immediately. Skype (news - alert) takes a percentage as any middleman would. In speaking with people in the industry, they tell me off the record that porn will be the biggest application of this service as phone numbers will not show up on a credit card or phone bill.

Just like the Internet itself, the porn industry will be a huge part of the business but its use will finance better infrastructure and other large applications — which I predict will be gambling, horoscopes, games, and dating lines. Hey, I didn't say the future will be pretty.

The Presence Of Presence

Within a few years, expect your PC or laptop to have built-in Bluetooth. We will all have Bluetooth phones. Software will know where we are within the enterprise at all times. Again, I am just the messenger but I would love a day when I know where coworkers are at a moment's notice. If the marketing team is in the conference room, I may decide I need to connect with all of them. Many employers will want to know if workers aren't in the office and this is a simple way to make it happen.

Brutal VoIP Wars

Every person on the planet who talks on the phone will use VoIP. This is inevitable. The question is will there be enough business for all? Microsoft, Vonage, eBay, Skype, AOL, Yahoo — and these are

Voice Enabled Communities: The Future Is Here

By Rich Tehrani

People from outside the United States are registering for Internet Telephony Conference & EXPO at a breakneck pace. We had more international attendance at our Miami show (February 2005) than any other domestic VoIP event, but I wasn't sure if we would be able to do the same in California at the end of this month. It seems the interest level is very high overseas and many of our conferees — service providers, enterprises, and resellers are registering from around the world.

I suppose the Skype/eBay acquisition is part of the reason. This one acquisition has really raised the prominence of VoIP to such an extent that people are beginning to understand that VoIP is not just about cheap minutes.

Many Skype users live overseas and Skype itself is in Europe so this acquisition has really woken Europe and the rest of the world up to the possibilities of VoIP.

What really excites me is that at first many analysts didn't understand why eBay would want to buy Skype and then a few days later, they got it. It is all about the services. Well, some of them are still scratching their heads. At least for now.

We are really on the threshold of VoIP 2.0. eBay's purchase of Skype proves this point. Sure, eBay will continue to make money from SkypeOut and SkypeIn, but the serious revenue is going to come from applications such as pay per call, allowing conferencing between buyers and sellers and a host of voice-enabled e-commerce applications we can't imagine today.

The whole concept of voice-enabling communities is in its infancy. There are hundreds of thousands of communities on the Web from information sites to dating services to blog sites. The next phase of growth in these communities will be voice enablement. We will see the ability to click to call other people in a community as a service that many communities allow.

In the online dating world we are already starting to see this happen, as in this application people are usually not willing to give a phone number to someone they don't know but are willing to engage them in a conversation. I am told these conversations will last for a while — at least until one of the people says something really stupid. Of course, this is what my single friends tell me. The obvious next step for dating services is to allow video, as all my friends also tell me dating sites always seem to feature pictures of people that don't actually look like the people when you meet them. Video should solve this problem and I expect many people to ask to have a video conversation before they get together. Note to self, now is the time to buy stock in video camera and cosmetics companies.

There is revenue to be generated in this model beyond dating as a site that allows you to sell cars can now allow links to have potential buyers call a seller with a single click. These sites can now charge to add voice capability. They can charge a voice fee of $5 a month or a dollar per call. Of course these numbers are just ideas, please don't price your service based on my off-the-cuff numbers.

This week I wrote about how Amazon.com is adding a new call center, and I went on to mention that Jeff Bezos is credited with saying that the web would kill the call center. It turns out that e-commerce actually fuels the need to speak with people. In the 1999-2000 timeframe there was a surge in call center equipment sales due to the growth of e-commerce.

I predict we will soon see the same thing happen with VoIP equipment. There will be a mad dash to voice enable communities. Newspapers, Web sites, blogs, classified sites, sports sites, every site. If you have a decent-sized community of interest expect that you will have to figure out how to voice-enable it.

Voice is the next phase of the Internet. We will voice-enable anything and everything. We will do this because voice is the most natural interface we have and we will do this because there is money to be made by doing it. Most importantly, after eBay voice-enables every product and service they have, we will do it to be competitive and to meet customer demand.

The voice-enabled Internet is here. I hope to see you at the Los Angeles Convention Center on October 24–27, where we are proud to offer the best education on how to voice enable your company, your Web site and your life. The future is VoIP 2.0 and you can see it live at Internet Telephony Conference & EXPO. IT

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just a handful of companies in the space. I didn't mention AT&T, Vonage and others. Then there are hardware companies. US Robotics, for example, picked today to get into the VoIP hardware business. Will there be enough business to go around? This is an important question as VoIP grew and died from 1999–2001 because of over-investment by CLECs and not enough customers.

This time it is different as today's VoIP companies spend a fraction of what yesterday's CLECs spent. Few companies are laying down miles of fiber to homes and offices. Moreover there are customers this time. Vonage has about a million and Skype has over 50 million.

I predict there will be a shakeout at some point but there are billions of dollars out there for the smart companies to pocket if they move quickly enough. After all, look at Skype. The company released a bit of software and became a phone company supporting 50 million users in just a few years and without too much VC money. Now that's a disruptive market.

China

I couldn't think of a better heading for this topic as China in just a few days announced it will fine users of Skype Out and then China Telecom decided to block the Internet ports that carry Skype traffic. In China and in other countries you need a license to carry VoIP minutes. If the world copies China, we are in a tough pickle as an industry. Sure, VoIP will still be used in these countries but markets will not be truly open — meaning little consumer choice. In many countries this isn't news. Hopefully the WTO will help stop port blocking.

VoIP Peering

More service providers, Fortune-class corporations, and contact centers peer every day, building a new voice Internet. Soon, most VoIP users will bypass the PSTN altogether. It is at this point the Universal Service Fund will crumble and the U.S. government will look to tax broadband to make up for the deficit — perhaps sooner.

In the meantime, VoIP users who peer will enjoy better quality VoIP calls and eventually my personal dream of stereophonic surround-sound conversations will become a reality.

Reflect & Learn

Twice a year, the industry gets to reflect on all of the above issues and moreover learn about them in more depth and detail. Of course I am talking about Internet Telephony Conference & EXPO (http://www.itexpo.com) — the preeminent VoIP conference. This year we are so excited to announce that we have the best educational program we have ever developed and we will go into depth on all of the above. There is nothing left out. Skype founder Niklas Zennstrom is back by popular demand, returning to speak at the show where he made his U.S. debut. This will mark the one-year anniversary of Mr. Zennstrom speaking at ITEXPO and I am more excited than ever to speak with him so I may learn more about how the eBay acquisition will change the landscape of communications. I hope to see you there.

Oh, and don't forget! We will have Carly Fiorina and Former FCC Chief Michael Powell speaking as well. You won't want to miss this show. I am just blown away by the level of companies — from service providers to resellers and Fortune-class enterprises that are registering. We expect this to be the largest Internet Telephony Conference & EXPO ever and are very excited to be bringing the world's largest VoIP event to the Los Angeles Convention center October 24–27, 2005. IT

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Arizona Cardinals Team Up With Cisco

By David Sims

The Arizona Cardinals and Insight are teaming up to run a Cisco (quote - news - alert) IP Communications system throughout their training facility in Tempe and their new football stadium, scheduled to open in 2006.

The system, implemented and managed by Insight, will provide approximately 750 Cisco IP phones at the new Cardinals stadium, which will open next August of next year.

In July, Insight hooked up the Cardinals' Tempe, Arizona training center with voice, data, and video communications. Digital video technology operating from the converged network will, in the words of Insight officials, "allow team members to participate, via video, in real-time training sessions from both the stadium and their Tempe training facility simultaneously, which brings a new dimension to the process of game preparation."

Video-enabled Cisco IP phones for executives will "facilitate video conferencing and staff," allowing them to "send data, game film and marketing materials from one facility to the other, saving time and operational expenses. The systems at both facilities will be fully linked, fully integrated and fully redundant."

In the new stadium, Cisco IP phones with touch-screen LCDs in the stadium suites will allow fans to pick players off of a virtual roster during a game to engage in a fantasy football game. Using the phone's touch-tone color screen, fans will also be able to buy tickets for future games, shop at the Cardinals' online store, order food and beverages and pull up football statistics.

http://www.cisco.com

New World Symphony, Northwestern Tap LifeSize

By Johanne Torres

Thanks to a new partnership between New World Symphony (news - alert), Northwestern University, and LifeSize, conference attendees from 207 universities will be able to experience live music instruction on high definition video and audio.

LifeSize, a high-definition video communications company, announced that it is currently working with Chicago's Northwestern University and New World Symphony in Miami to demo a live music instruction session using LifeSize's high-definition video and audio products.

The live, interactive demo will be conducted on September 20th during the Fall 2005 Internet2 Member Meeting featuring a high-definition video communications link between the conference site in Philadelphia, the instructor located in Chicago, and the music students in Miami.

"We are tremendously excited to collaborate with LifeSize and Northwestern University in this demonstration of next-generation video and audio communications technology," said Tom Snook, New World Symphony CTO. "New World Symphony is a virtual institution relying extensively on distance learning to provide our students with the highest levels of musical instruction. LifeSize's high-definition products enable us to enhance the remote communications experience for instructors and students, creating a learning environment very much akin to that of a traditional setting where all parties are in the same room."

According to the companies' news announcement, "Internet2 conference attendees will be able to view the three-way video multipoint live music instruction at up to 1Mbps with LifeSize's flagship high-definition video communications solution, LifeSize Room."

"Even at a bandwidth-rich institution such as Northwestern, there is an appreciation for how lower bandwidth video technology can help us to reach new audiences for our teachers and researchers," reports Larry Amiot, senior digital video engineer at Northwestern University. "By offering better quality conferencing at low bandwidths, LifeSize's high-definition video promises to dramatically improve the user experience for any organization."

http://www.lifesize.com
http://www.northwestern.edu
http://www.nws.edu

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Toshiba Announces Video Communications Solution

Toshiba America Information Systems Inc., (news - alert)Digital Solutions Division (TAIS DSD), announced its new Video Communications Solution (VCS) for its Strata CIX family of IP business communication systems.

Toshiba's VCS will initially provide affordable point-to-point (P2P) video communications and collaboration (desktop and applications sharing) capabilities. Using voice extensions, video can seamlessly be added to telephone conversations. A number of default settings (on, off, non pop-up) can be configured to meet the individual user's needs.

Features include:

• Integration — Fully integrated with Strata CIX systems and pre-installed when shipped with the Media Application Server (MAS) and managed by eManager, VCS easily extends the user's Multi-media experience, including video and data sharing with the VCS client running on Windows XP.

• Video — Friendly and easy-to-use interface allows participants to set up video communications and conferencing to see, hear and interact with each other.

• Collaboration — Participants can use various communications tools such as voice or Instant Messaging, share and collaborate with their desktop applications or documents, and exchange files anytime and anywhere.

Functionality is integrated into the telephony capabilities with features specifically tailored to handle video telephony, including features such as hold, transfer, and forward.

Future releases of VCS will include features such as Presence (Ability to see where Strata users are in real time, what they are doing, and their individual preferred method to be reached), Instant Messaging, three-party video conferencing, and a Toshiba's Multipoint Contact Unit (MCU) to support up to eight parties with full video conferencing and enhancements to collaboration capabilities.

Video communications takes productivity to a new level by allowing remote workers to see, hear, interact and collaborate on projects together, no matter where they are physically located. Whether it's used for daily organizational calls or VIP communication, VCS provides the remote way to conduct virtual in-person meetings for staff education, field engineer support, sales staff support, and much more. Because it takes the place of face-to-face meetings, video communications can dramatically reduce the cost of business travel.

Toshiba's new VCS works with not only Toshiba's newer Strata CIX IP platform but also with older Strata CTX TDM business communication systems that have been upgraded with Release 3.1 to add IP capabilities. Meyer said, "With VCS, our users can improve their communications and productivity with an affordable entry cost to video communications. Toshiba is once again delivering 'big company' features and functionality to its SMB customer base."

http://www.telecom.toshiba.com

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VoIP Driving Security Appliance Market

By Johanne Torres

A recent study conducted by research firm In-Stat found that more than seventy five percent of businesses that adopted VoIP-calling services have plans to replace their security appliances within the next year.

The firm's analysts believe that while "VoIP can add many desirable new features to business telecom systems, vendors and customers face difficult security challenges to realize these benefits." The study forecasts that the security appliance market is ready to grow over the next few years, possibly reaching gains of at least $7 billion by 2009.

"Traditional firewall technologies can complicate several aspects of VoIP, most notably dynamic port trafficking and Network Address Translation (NAT) transversal," says Victoria Fodale, In-Stat analyst. "Security product vendors are adding functions that address voice applications in their products, but, as history has shown, security typically lags behind advances in technology."

Recent research also found that companies with 500-999 employees increasingly show more concerns about VoIP security than companies of other sizes. Analysts also found that companies have begun to assign larger budgets for security appliance purchase.

In-Stat's study titled *Trends and Spending Plans for Security Appliances: Are We Ready for VoIP?* also revealed that "reliability is by far the most important criteria for the purchase of new security appliance products by businesses."

http://www.in-stat.com

HP, Macromedia To Help Carriers Deliver Triple Play

By Robert Liu

Macromedia's (quote - news - alert) recently introduced, solutions-centric Flash Platform received a favorable endorsement as key programming model and developer toolset for the telecommunications field when Hewlett-Packard (quote - news - alert) announced it will jointly integrate and market the software package as part of its HP Services Delivery Platform (SDP).

In its press release, Macromedia said the partnership will enable carriers, network equipment providers, and telecommunication ISVs to streamline the creation of new communications, messaging, and collaboration solutions across fixed, mobile, and broadband networks.

Flash Platform is a package that Macromedia introduced in June that is designed to leverage the massive Flash player installed base by tying in popular software titles like the Flash MX design tool, Macromedia Flex for applications development, Macromedia Breeze for delivering online communications, and Macromedia Communications Server for two-way audio/video streaming. Using those software tools, Macromedia engineered industry-centric vertical solutions in a service-oriented architecture (SOA) approach, making the partnership with HP SDP a natural fit.

Macromedia and HP are clearly hoping to capitalize on the urgency of telecom service providers to roll out next-generation communications that unify voice, video, and data. For example, instead of working with say IBM and its IMS solutions partner Ubiquity to create and deploy multimedia IP services, carriers can tap into the lightweight and widely distributed Flash technology to speed deployment. In fact, the solution has already been deployed by Telecom Italia, which has deployed a video chat application called "Rosso Alice" and Telecom Italia's CTO Stefano Smareglia has previously stated that Flash has enabled its "Rosso Alice" to compete "with other entertainment channels and platforms."

"Carriers investing in multi-service IP networks will accelerate time to market for the broad adoption of services by leveraging the instant 'network effect' of Macromedia Flash and the reach of the HP SDP, while still maintaining strong brand control and differentiation," said William Stofega, senior research analyst, VoIP Services, IDC. "The Flash Platform combined with the HP SDP creates a robust, multiplatform solution for the enterprise that is suitable for stringent, mission-critical environments across the telecommunications network."

As part of the joint initiative, HP will offer turnkey systems integration services to the growing number of telecommunications service providers deploying Flash Platform products in their networks. Macromedia and HP will initially target their joint sales efforts to major carriers around the world that have deployed key elements of the SDP. As part of their professional services engagements, HP will resell elements of the Flash Platform, including Macromedia Breeze for delivering online communications and Macromedia Flex for developing rich Internet applications.

Most logically, the first application based on Flash to be integrated with the SDP is Macromedia Breeze, which has already been tightly integrated with popular platforms such as Salesforce.com's customer relationship management (CRM) solutions. Cable & Wireless, KDDI, and Verizon have also been working to deploy the Macromedia Web conferencing solution within large enterprises for internal training and communications.

http://www.hp.com
http://www.macromedia.com

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Pannaway Releases Broadband Access Manager

By Ted Glanzer

Continuing to make waves in the rural telco next-generation broadband service convergence market, Portsmouth, N.H.-based Pannaway Technologies, Inc., (news - alert) announced the release of its Broadband Access Manager (BAM), which is used to manage all components of its Service Convergence Network (SCN).

According to a company statement, "Pannaway's RUS-listed SCN is the industry's first end-to-end IP solution for secure, converged broadband transport services, allowing companies of all sizes to optimize the existing 'pipeline to the premise' to build out a network for Triple Play services."

As part of the SCN, the BAM server is a "sophisticated but extremely affordable" broadband element management system that provides telcos the flexibility to leverage the benefits of IP while simultaneously reducing installation complexity and lowering operational expenses.

Priced at $17,200, the BAM server is delivered on a 1RU appliance that "runs 24/7," and redefines the term truck roll for telecom service providers.

Indeed, the integrated capabilities of the BAM Server include the following:

- Automated Line Qualification;
- Auto Network Discovery;
- Dynamic Service Activation;
- Rapid Device Identification;
- Enhanced Network Trouble Shooting;
- Remote Device Analysis;
- Group Device Management;
- Automated Subscriber Management; and
- SIP Call Control Manager Configuration.

"Pannaway placed a big bet on an IP/Ethernet infrastructure early on and are now poised to reap the rewards," said Matt Davis, Director of Broadband Access Technologies at Yankee Group. "Well-integrated IP/Ethernet networks will reduce [operational expense] costs through simplification, automation and by streamlining a variety of network management tasks."

Proof, as they say, is in the pudding.

The BAM Server was a primary reason why Gilmer, Texas-based ETEX Telephone Cooperative announced that it selected Pannaway for the delivery of next-generation broadband services.

ETEX Telephone, which supports 17,000 access lines in northeast Texas, said that it plans to transition their existing ATM network to one that leverages pure IP from end-to-end, according to a news release.

"We knew early on in our evaluation process that a pure IP network would be a better investment for us based on the types of applications that we plan to deliver," said Charlie Cano, engineering manager at ETEX. "Two big factors that led us to a Pannaway selection included their broadband access management system for end-to-end service provisioning, line testing and network troubleshooting, and their commitment to becoming more than just a technology provider, but a true partner in every sense of the word."

Another Pannaway customer echoed those sentiments.

"Using BAM, we can leverage an automated management system to remotely qualify lines, deploy CPE and push out regular maintenance releases," said Carl Burgess, CEO of Bartow, Fla.-based CLEC REI Communications. "This will minimize truck rolls, keeping our expenses down, ultimately translating to very competitive prices for our subscribers."

http://www.pannaway.com


Looking
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VoIP, Inc. Offers Private Network 911 Service

By Ted Glanzer

One thing has not been lost in the recent news regarding the Federal Communication Commission providing 30-day extension of time to PSTN-connected VoIP service providers to comply with the notice provisions of its E911 order.

The service providers are still required to offer E911 to their subscribers by Nov. 28, a deadline viewed by many as an unrealistic one to accomplish such a monumental task.

Thus far, several companies, such as Intrado and TCS, have stepped forward to offer solutions that route VoIP subscriber 9-1-1 calls into the dedicated wireline E911 network.

However, VoIP, Inc., (news - alert) CEO Steven Ivester said that in their haste to comply with the FCC's mandate, service providers are risking emergency call quality and security.

"The industry has been focused on creating quick solutions to meet FCC deadlines related to recent regulatory actions that require VoIP service providers to offer 911 services to customers," he said. "Some of these solutions continue to use unsecured, and best-effort transport of the Internet for critical emergency calls."

Accordingly, VoIP, Inc. has entered the FCC E911 fray, announcing the launch of "the industry's first private network 911 service for broadband and packet communications," promising "unprecedented" quality and reliability.

"We are changing the dynamics at VoIP, Inc., with this unprecedented move to go beyond the requirements of the FCC order to help ensure reliability and quality of 911 calls," Ivester said.

The service utilizes VoIP, Inc.'s proprietary VoiceOne network; once a call is received on the network, it is analyzed for the proper delivery to a Public Safety Answering Point (PSAP), call center, or other emergency location.

"Based on the destination of the location required to deliver the call, it is maintained on the private VoiceOne MPLS backbone, ensuring the best-possible quality and redundancy for transport, compared to other solutions utilizing the Internet for call delivery," VoIP, Inc. CTO Shawn Lewis said.

According to the company, the 911 service feature can be managed by carriers, service providers, cable operators, and others through the VoiceOne Web portal or XML provisioning system "in near real-time fashion" — a feature that is "not offered by solution providers in the 911 industry today."

"Offering IP related services, with the quality of service and reliability levels carriers and consumers demand and deserve, was our primary focus in designing this product offering," VoIP, Inc. CTO Shawn Lewis said. "Secondly, making the service affordable and available to all with a simplified interface was critical."

http://www.voipinc.com

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StarVox Partners With Veraz To Upgrade Domestic VoIP Network

By Ted Glanzer

San Jose, Calif.-based next-generation service provider StarVox Communications, Inc., (news - alert) has upgraded its domestic VoIP network using Veraz Networks' lastest softswitch release and media gateways. The upgrade, according to a company release, "enables StarVox to offer business the premium voice quality that they are used to, using [service level agreements] that guarantee the Quality of Service of VoIP."

StarVox' Vice President of Marketing Rich Barry explained that utilizing Veraz Networks' equipment is key because businesses are "very hesitant to switch to VoIP networks" if they lose any voice quality.

"Having Veraz Networks as our softswitch and media gateway supplier gives StarVox the most advanced and flexible next-generation network in the market," said StarVox President and CEO Doug Zorn in a prepared statement. "This means that we can meet our goal of offering businesses the most advanced next-generation network solutions."

StarVox' domestic VoIP network, which was launched in mid-July, has expanded to cover most major metropolitan areas in the United States, with long distance and toll-free calling within reach of over 80 percent of business sites, and local phone service to over 50 percent of business sites.

Barry said that StarVox has brought on a large number of wholesalers who are using the network and that the company is starting to gain retail customers through agent channels.

The next "big thing," according to Barry, is the switch from the PSTN last mile to the VoIP last mile, in which a gateway is installed at a customer's site. Such a switch would enable a customer to handle about three times the number of calls as a regular PSTN (define - news - alert) voice network.

"There is a strong economic advantage to go to VoIP last mile," Barry said

Veraz' Senior Director of Product Marketing Ed Camarena said that, "the StarVox announcement represents an ongoing trend we're seeing with service providers taking advantage of services oriented solutions for the customers."

"StarVox's roll-out of converged IP-based applications on a nationwide basis supports our vision of using the Veraz softswitch and media gateways as the core components of next-generation service provider networks," said Doug Sabella, president and CEO of Veraz Networks, Inc.

Recently, Veraz Networks, (news - alert) a global provider of softswitch-based packet telephony solutions, announced that it was the lead vendor for a deployed solution at Cable & Wireless for 600,000 prepaid residential and calling card subscribers in Panama.

Veraz also announced a recent win in June with one of the largest service providers in India.

"We've got a lot of traction in the U.S. and outside the U.S. We're very well-positioned on a global basis," said Camanera, noting that Veraz has delivered VoIP products to service providers in about 50 countries. "[VoIP] is becoming the technology… for the world."

http://www.starvox.com
http://www.veraznetworks.com

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SunRocket Picks XO For VoIP Termination

By Johanne Torres

Recently, XO Communications, Inc., (news - alert) announced that the company has been selected by SunRocket for VoIP termination services.

The new agreement calls for SunRocket (news - alert) to use the XO VoIP Termination to support its Internet home phone service. According to the company's news release, "XO VoIP Termination enables SunRocket to hand off voice traffic to XO, allowing SunRocket's customers' IP-based calls to be routed across the XO IP network at the highest priority and terminated to the public switched telephone network at destinations across the United States."

"SunRocket delivers reliable, best-of-breed services to our customers, and choosing the right network partners ensures that we continue to provide that outstanding quality of service, including enhanced 911, while growing rapidly," said Rob Mainor, chief operations officer at Vienna, Virginia-based SunRocket. "XO is a leading enabler of VoIP service providers and will enable us to broaden our network coverage and performance as we continue to rollout SunRocket service nationwide."

By selecting XO to terminate VoIP (define - news - alert) traffic, companies virtually hand over their VoIP calls directly to XO for termination to the public switched telephone network (PSTN). The agreement will allow SunRocket's voice traffic to be routed across the XO OC-192 class of service-enabled network at a higher priority level. The traffic will pass through an XO softswitch gateway and will be carried across the XO IP network to an XO softswitch gateway nearest the final destination of the call. The process will convert the traffic by a XO softswitch to a format accepted by the terminating local service provider.

"SunRocket's selection of XO is another demonstration of our success in delivering advanced wholesale VoIP solutions for today's new breed of phone companies," said Ernie Ortega, president of carrier sales at XO Communications. "Providers like SunRocket are driving the adoption of innovative voice services and we are proud to be a partner for them."

Paul Erickson and Joyce Dorris, former MCI executives who were responsible for MCI's "1-800-COLLECT" and "The Neighborhood Built by MCI," announced they set up VoIP-enabled call service company SunRocket back in November last year with the aid of Nokia Venture Partners, which provided the startup's first round of financing.

SunRocket launched the SunRocket Signature Services, a new residential primary-line phone service priced at $24.95 monthly, with no activation charges, no contracts, term-commitments, activation fees, shipping charges, equipment charges, or cancellation fees. The service is currently available in Washington, D.C., Baltimore, and Boston.

http://www.sunrocket.com
http://www.xo.com

Telstra Completes Vistula's VoIP Platform Implementation

By Johanne Torres

Telstra Europe, Ltd., (news - alert) has signed off on the technical implementation of Vistula Communications Services, Inc.'s (news - alert) V-Cube platform.

Technical implementation of the VoIP platform represented the final clearance stage before full product launch by Telstra Europe, a subsidiary of Australia communications giant Telstra Corp., Ltd., of a number of hosted VoIP PBX services to client groups in the United Kingdom and Ireland.

According to Vistula's news announcement, "Telstra is on target for its VoIP products going live in October of this year, although some customers have been using the services successfully since June."

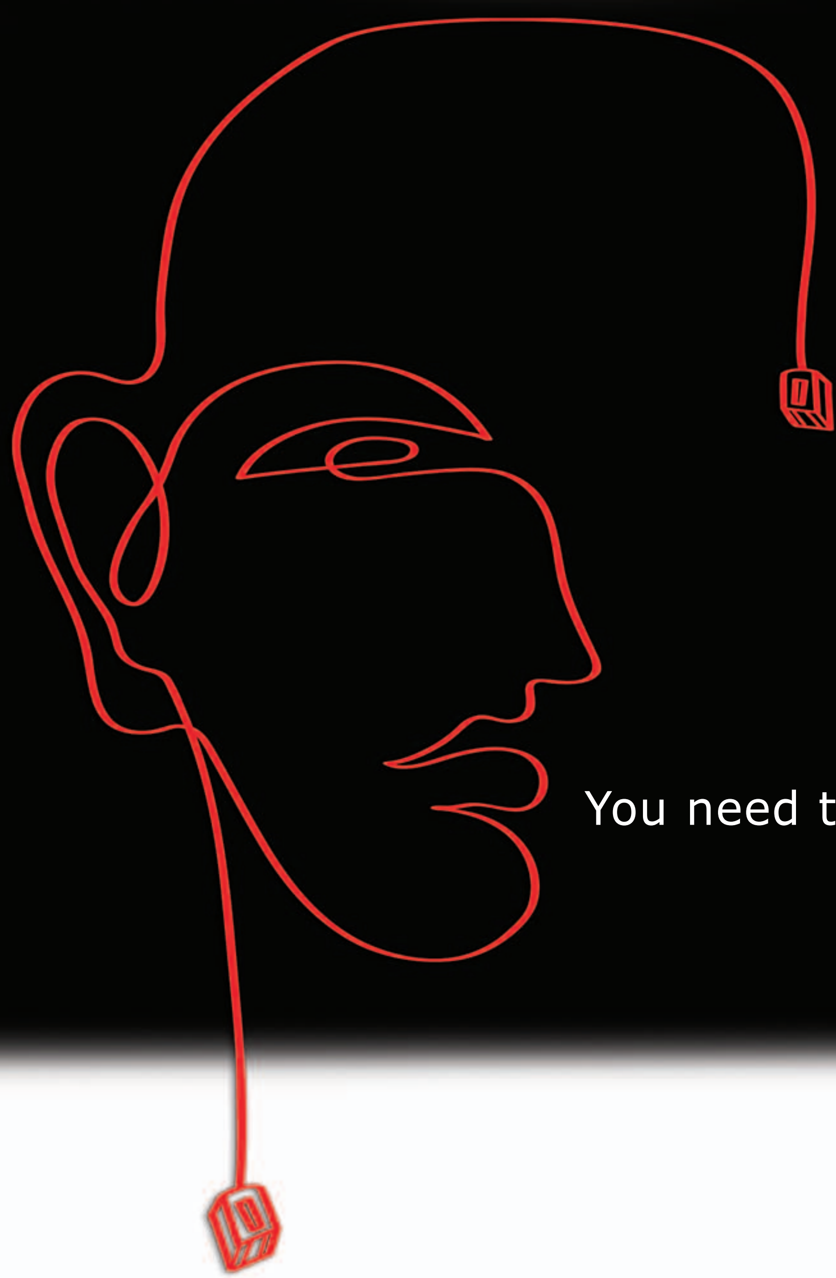
"This is a significant event for Vistula with V-Cube reaching its go-live status with a major Tier 1 carrier," said Rupert Galliers-Pratt, Vistula's chairman and CEO, "we are obviously delighted with the progress to date and look forward to Telstra's success in the UK and Irish markets."

"We set ourselves an extremely aggressive launch target and are delighted with the progress we have made in creating our product set in line with the requirements of our Sales and Marketing divisions," noted Telstra Europe CEO Dave Thorn. "We have been working hand in hand with Vistula's engineers for some months and feel that we have achieved everything that the company promised. We look forward to the next phase when we bring our paying customers on-line. We already have a healthy pipeline of enquiries for our service due to its feature rich applications and market leading price point and expect to make a series of customer announcements shortly."

Vistula announced back in April that Telstra had signed an exclusive, five-year contract to distribute Vistula's V-Cube VoIP platform in the United Kingdom and Ireland.

In addition to a set of telephony features, the agreement called for Telstra Europe and Vistula to create a client Web-based portal that takes customers through sign-up, giving them control of their own VoIP domain, and access to online features such as Web-based voice mail and statistical information about their usage.

http://www.telstra-europe.com
http://www.vistula.com



WildPackets Launches Portable VoWiFi Analyzer

By Johanne Torres

WildPackets, Inc., (news - alert) recently introduced the AiroPeek VX v1.0 Expert VoWiFi (Voice over Wireless) analyzer. The network analysis provider also upgraded its line of AiroPeek SE and NX WLAN analyzer products.

The three products now include simpler configuration settings that provide ready enforcement of wireless network security policies, GPS location-tagging to enhance site survey exercises, and real-time application analysis to better identify performance issues, including those masked by WEP and WPA encryption.

"The AiroPeek product line has had a very large and loyal following worldwide since the SE version first shipped in 2001," noted Mahboud Zabetian, WildPackets' CEO. "Over 6,000 organizations own AiroPeek, and consistently invest in the product's ongoing development through our maintenance programs. A wireless voice product is a natural progression for us and the next step for many of them. Our 15 years of self-funded growth and profitability have been made possible through focused delivery of relevant, affordable solutions for our customers for each emerging technology: in the wired world, in wireless, and now in voice."

The AiroPeek product line includes site surveys, security assessments, client troubleshooting, WLAN monitoring, remote WLAN analysis, complete VoIP analysis, and application layer protocol analysis.

All three products are currently available and have SRPs starting under $1,000.00 USD. Current maintenance customers receive v3.0 of their registered AiroPeek NX or SE product for free, along with a lower-priced upgrade offer for AiroPeek VX.

This news follows the company's recent announcement about it releasing version 3.0 of the Omni enterprise network analysis platform in June. The new version adds in-depth VoIP call analysis, support for PCI-based T1/E1 WAN analysis, and enhanced support for Datacom Systems and Net Optics matrix switches.

The company explained that the Omni architecture is comprised of OmniPeek Consoles and Omni DNX Engines that deliver coverage and analysis of local and remote networks. The service and product suite can analyze VoIP (define - news - alert), Gigabit, WAN, Wireless, and Ethernet traffic from live data captured by Omni DNX Engines deployed in any local or remote location. WildPackets says that OmniPeek Voice adds real-time VoIP diagnoses to the statistical analysis and distributed, real-time network troubleshooting provided by the Omni solution.

http://www.wildpackets.com

Tripp Lite Introduces Digital UPS System With LCD

Tripp Lite (news - alert) has introduced a new UPS System that features a digital LCD display. The SMART1000LCD Digital UPS System features a large, backlit display designed to clearly and dynamically indicate input voltage, UPS battery capacity, and a variety of operational conditions.

"The liquid crystal display featured in Tripp Lite's new Digital UPS System is a radical departure from standard LED sets featured on traditional UPS systems in this class," said David Slotten, Tripp Lite Director of Product Management. "The LCD display makes the UPS a more dynamic diagnostic tool that not only allows end-users to better understand their power conditions but also reassures them that their equipment is completely protected against downtime and damage."

The SMART1000LCD Digital UPS System provides 1000VA of conditioned power, long-lasting battery backup and eight outlets to accommodate a variety of computer and home entertainment components. Line-interactive operation including Automatic Voltage Regulation (AVR) corrects brownouts without using battery power. A versatile cabinet is adaptable for both tower and rackmount (2U) applications. The LCD display is rotatable for easy viewing in any position. The SMART1000LCD also features single-line phone/DSL surge protection (for phone, fax, modem or DSL connections) and single-line coaxial surge protection (for cable modem, DSS satellite, cable TV or antenna connections). The HID-compatible USB port supports the UPS monitoring functions built into Windows and MAC operating systems. PowerAlert Software, available as a free download, provides automatic unattended shutdown.

http://www.tripplite.com

A New 4U 8-slot CompactPCI Subsystem

ADLINK Technology Inc., (news - alert) a global leader in developing CompactPCI and AdvancedTCA platforms, presents a new 4U rackmount CompactPCI enclosure with eight-slot 6U backplane and redundant power supplies. The cPCIS-6418 is designed to deliver optimal performance for telecommunications, CTI, Voice over IP, and network equipment providers.

The cPCIS-6418U is a high-performance CompactPCI platform with a 64-bit, eight-slot 6U backplane in a 4U enclosure. The eight-slot backplane dedicates one slot to the system CPU board and seven slots to peripheral boards. It supports a PICMG 2.5 compliant H.110 CT Bus. Featured with high availability and room for expansion, the cPCIS-6418U is compliant with the PICMG 2.5 computer telephony integration standards and is ideal for Voice over IP and network equipment applications.

This subsystem is equipped with a front access, hot-swap cooling system for effective CPU heat ventilation and high-density computing. The cPCIS-6418U uses hot swappable 500W + 200W redundant power supplies with a universal AC input. The cPCIS-6418U uses a 4U enclosure and is categorized as a high-availability model. The cPCIS-6418U was designed not only considering overall system power loading but also overheating issues from multiple CPU boards within the enclosure.

ADLINK also offers a compatible 6U CompactPCI CPU board for use with the cPCIS-6418U: cPCI-6860A and cPCI-6840. These two CompactPCI CPU boards are easily expandable and can be utilized to take full advantage of all the cPCIS-6418U's capabilities. The cPCI-6860A is a 6U CompactPCI dual low voltage Xeon processors Host single board computer and the cPCI-6840 is a 6U CompactPCI highly-integrated Pentium M processor PCI 64-bit/66 MHz Universal Blade.

The cPCIS-6418U is competitively priced at $2695 and is available with OEM discounts.

http://www.adlinktech.com

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Phihong Multiport Midspans Now Compatible With Gigabit Systems

Phihong (news - alert) announced that the company now offers a multiport midspan power injector for Power-over-Ethernet applications that can be used with Gigabit systems. A key differentiator according to company officials, is that the Phihong Midspans are not limited to working only on 10baseT or 100baseT systems.

"Our customers realized that Power-over-Ethernet was a smart solution to their networking needs, but the midspans on the market were not designed for high-speed applications such as broadband wireless," said Keith Hopwood, Vice President of Marketing for Phihong USA. "We designed our low-cost injectors so that they can be applied to any network without compromising the data rate. Now IT professionals can take our Midspan and get PoE on any network without having to check compatibility."

Available in one-, eight-, 16 and 24 port versions, Phihong's high-speed Midspans offer full 15.4W per port at a cost that is 40 percent lower than competitor solutions.

PoE allows both power and data to be carried over an Ethernet cable. A midspan is used to implement PoE into an existing network where power can be inserted on the spare pair.

The enterprise midspan provides one to 24 ports to upgrade an existing architecture with total output power up to 370W. Offering 15.4W per port, the midspan will deliver full power to all ports so there is no need for calculating which devices to connect. Each midspan comes with software to control and monitor each port along with full power management capability for load shedding in back-up power conditions.

Pricing for the 24-port Midspans start at $295 in OEM quantities.

http://www.phihong.com

NETGEAR Chooses Centillium To Speed VoIP Platform Development

By Johanne Torres

NETGEAR (news - alert) has selected Centillium Communications, Inc., (news - alert) in order to speed things up as it develops a new VoIP platform for the consumer and small business markets. Specifically, NETGEAR selected Centillium and its newest Atlanta system-on-chip (SoC) system in order to reduce development time and preserve software investments across multiple platforms.

Centillium's Voice Services Platform (VSP), part of the Atlanta product and service suite, includes a VoIP media terminal adapter (MTA), a VoIP cordless phone, a two-line gateway and four-line SOHO router, and a triple-play set-top box. According to the companies' news release, with the VSP in place, it should take NETGEAR less than three months to have the service up and running.

"NETGEAR is committed to bringing our customers the most advanced solutions, and the software portability of Centillium's Atlanta family ensures that we are making a secure investment in a technology that can be easily upgraded as customer requirements change," said Patrick Lo, chairman and CEO, NETGEAR.

"Atlanta's high-performance routing and throughput delivers superior voice quality, while its versatile, configurable software facilitates cost-efficient, quick time-to-market. With each Atlanta family member optimized to deliver a different level of capabilities, the built-in software portability provides significant benefits for our customers to upgrade their service offerings without having to retool software," he added.

All devices in the Atlanta family are both Linux and VXworks-compatible. Other Atlanta products include the Atlanta A70, an entry-level SoC for clear voice quality; the A80 for an added capability to interface with any WiFi or high-speed adapter; the A90 for SOHO clients with four voice channels and a routing engine available at 100 Mbit/second; and the A100 for enterprise-level security and encryption of all data and voice.

http://www.centillium.com

http://www.netgear.com

Jabber: The Reluctant Voice Player

By Robert Liu

Jabber (news - alert) plans to integrate Session Initiation Protocol (SIP) directly into voice capabilities of its core Extensible Messaging and Presence Protocol (XMPP) — occasionally referred to as the "Jabber" protocol — by next spring.

The news underscores the growing importance of voice signaling in the instant messaging and presence realm. While AOL (quote - news - alert), Yahoo (quote - news - alert)and Microsoft (quote - news - alert) have long dominated the IM space, upstarts like Skype have demonstrated how even industry stalwarts can become marginalized if they don't stay on top of the latest trends.

"IM offerings continue to increase their functionality to include features such as audio/video, conferencing, presence, and Voice-over-IP, so that they are increasingly resembling collaboration suites," according to Matt Anderson, market analyst at Radicati Group.

As such, Jabber, the company that founded and supports the open source instant messaging platform used across tens of thousands of Internet servers by millions of users, plans to introduce new SIP capabilities to its Jabber Extensible Communications Platform (XCP), eliminating the need for a gateway. Jabber will first introduce server-side components in the first quarter of 2006. SIP-based voice signaling will be introduced on the client-side at some point next year, explained Joe Hildebrand, CTO at Jabber.

"From a Jabber device, you'll be able to ring SIP phones," Hildebrand said.

Historically, Jabber's XMPP technology, which has been accepted as a standard by the Internet Engineering Task Force (IETF), has tapped into SIP through an extension known as SIMPLE, which stands for "SIP for Instant Messaging and Presence Leveraging Extensions." But SIMPLE has been slow to gain industry-wide acceptance in part due to political wrangling as well as fragmentation by Microsoft, which uses its own version of SIMPLE in its Live Communicator Server.

Yet while SIMPLE has stumbled out of the blocks, the importance of SIP continued to grow exponentially. "SIP's not going away. We have to interoperate with it," Hildebrand explained.

The confluence of events has led the Jabber Software Foundation (JSF) to accelerate work on its own proposal to conduct multimedia sessions in XMPP over the Internet, dubbed "TINS" or Transport for Initiating and Negotiating Sessions (Jabber Enhancement Proposal #111). Hildebrand explains the goal was to capitalize on all of the work already accomplished by the SIP community.

"At first, we thought there should be some sort of convergence protocol between SIP and XMPP. But that wasn't practical," he said. "There have been people in the JSF community that want us to take a separate path for voice signaling. We might as well re-use all the work done by the SIP community."

Currently in only "experimental" stage, TINS will be fast-tracked by the Jabber Council in the next few weeks so that it can advance quickly to "draft" status. The proposal essentially maps specific syntax of XMPP to the semantics inherent in SIP. "The actual TINS specification is a document that says, 'When you see this in SIP, do this in TINS,'" Hildebrand said.

The move follows Google's (quote - news - alert) recent decision to use XMPP as the basis of the technology for its Google Talk IM platform. Now analysts believe that endorsement along with the latest outline of its technology roadmap could breathe fresh life into the XMPP platform, which has been losing relevance because of the growing importance of SIP.

"I was interested to see what Jabber was going to do eventually. I was surprised when Google chose XMPP. I guess it is kind of inevitable" that Jabber integrates SIP-based functionality, Radicati's Anderson explained during a telephone interview.

The company currently offers a SIP/SIMPLE connector (Version 1.0) and will soon offer a connector to Microsoft LCS's flavor of SIMPLE (Version 1.1). The new SIP-based functionality will be released as Version 2.0. And the irony of inevitability wasn't lost on Hildebrand.

"Voice is something that we're adding because a lot of people are asking for it. But the core value of our system is really about data transfer," Hildebrand admitted.

http://www.jabber.com

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TelTel Launches IP Telephony For Service Providers

By Johanne Torres

SIP-based Internet telephony technology provider TelTel (news - alert) recently launched the TelTel SIP Virtual Network Operators (SVNO) Partnership Program. The program is said to be able to allow ISPs, ITSPs, CLECs, carriers and other Internet service and content providers to offer SIP-based telephony and multimedia services without having to build their own SIP infrastructure.

According to the company's news announcement, "TelTel's SVNO Partnership Program is ideal for service providers that are seeking an immediately available total solution for rapid penetration into the VoIP marketplace." Additionally, partners will be able to either provide VoIP independently or as an added-on bundled service to existing offerings.

Businesses can sign up for a one-time fee, which covers access to the PsipTN platform, SIP termination and origination, a softphone available for partner customization, full integration with the partner's existing operation and customization with legacy networks. Additionally, SVNO partners can deliver conferencing, voicemail, PBX features, and Web-based portals to manage services.

SVNO partners can offer customers co-branded and remote configurable SIP equipment such as IP phones, ATA and videophones. SVNO partners with proprietary networks can also connect the network to the SIP infrastructure by offering their own termination and origination to the PsipTN. They can still use the PsipTN's existing termination and originations for areas outside their coverage area.

"The SVNO partnership program is an innovative business concept designed for rapid deployment of VoIP services within a partner's current marketplace." said Jack Chang, TelTel's COO. "Our partners will have a strong competitive edge over others by leveraging TelTel's proven expertise and know-how in delivering complete end-to-end VoIP solutions to their customers."

TelTel previously announced the launch of a Developer Kit for the TelTel Public SIP Telephone Network (PsipTN). The new kit gives service and platform developers access to the PsipTN, allowing them to develop "TelTel Ready" products and services.

The new PsipTN Developer Kit from TelTel features billing, provisioning, and co-branding APIs, together with a hosted "TelTel Ready" Asterisk-based platform for rapid application development and deployment. It offers the ability to create applications with commerce and communication capabilities such as Interactive Voice Response (IVR), Information and Entertainment Channels, Conferencing, Social Networking, Voice Messaging, Enterprise and Call Center applications, Customer Relationship Management (CRM), and PBX hosting.

http://www.teltel.com

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Nortel Launches SIP-Based VoIP Services In Rural Markets

By Johanne Torres

According to recent news, Nortel just finished its deployment of the DMS-10 platform in order to bring residential VoIP-based calling services to the rural U.S. market. With an expansion of the DMS-10 voice switch in place rural service providers will be able to offer Session Initiation Protocol (SIP)-enabled VoIP (define - news - alert) calling services.

According to the company's news announcement, "DMS-10 subscribers [have] the option to use traditional VoIP primary phone service and allows users to add cost-effective secondary VoIP lines." Users will also be able to modify call features and call routing through the DMS-10 Web-based portal. The company claims to already have more than 75 service providers scheduled to deploy the rural VoIP-enabled system this year.

"We are happy to see the DMS-10 evolve to a packet-based solution," said Norman T. Welker, president of Colchester, Ill.-based McDonough Telephone, which has approximately 4,200 lines. "With the addition of SIP-based services, we will be able to offer VoIP and phone service beyond our local territory to expand our subscriber base."

"Nortel (quote - news - alert) understands the challenges we face in evolving our infrastructure to a next generation, packet-based network and continues to demonstrate its commitment to rural market providers," said Roy Cranford, translations engineer for Randolph Telephone, which is based in Asheboro, N.C. and services 15,000 lines. "By delivering the SIP-enabled DMS-10, Nortel has given us the ability to evolve our network at our own pace to meet our subscribers' growing telephony needs. Our deployment went exceptionally well, including calls placed from a user 200 miles away while on vacation!"

Nortel will be taking the system on the road as part of its "Next Generation Networks Rural Market Tour 2005." More information on locations and dates of the tour are available at the company's Web site.

http://www.nortel.com

Sendia Announces WorkSpace CRM 3.0

By David Sims

Sendia Corporation, (news - alert) a wireless business application platform vendor, is announcing the release of WorkSpace CRM 3.0 for salesforce.com (news - alert), which features a host of powerful new features and upgrades, including support for salesforce.com's Customforce and other advanced capabilities such as filtered views, increased data capacity, and customizable lists.

Designed for salesforce.com mobile users, WorkSpace CRM 3.0 replaces Sendia's Wireless Salesforce Automation 2.3 and is compatible with popular wireless handheld devices such as BlackBerry Wireless Handhelds and Palm's Treo series.

Last December Sendia's Wireless Salesforce Automation product for palmOne's Treo 600 and Treo 650 smartphones was certified by salesforce.com for the ASP's sforce on-demand platform, according to SmartPhoneToday.

The sforce on-demand platform lets third-party developers customize, integrate, and extend salesforce.com with custom CRM products. Other sforce members include IBM (quote - news - alert), BEA, Borland, and Microsoft (quote - news - alert).

"Wireless SFA from Sendia extends salesforce.com's customer relationship management application to wireless-enabled handheld devices (such as Treo & RIM BlackBerry), making the information accessible and editable from the field," SmartPhoneToday reported.

With WorkSpace CRM 3.0, custom objects such as invoices, purchase orders, HR tracking, expenses, and project management originally created for the desktop version of Salesforce can be accessed on BlackBerry or Treo handheld devices. This allows administrators to decide the exact custom application functionality they want mobilized on their field teams' mobile devices.

With WorkSpace CRM 3.0, custom objects created in the desktop Salesforce appear as individual tabs in the device's user interface.

Alex Klyce, president of Sendia said that WorkSpace is designed bearing in mind that "most salesforce.com customers customize their applications."

For BlackBerry (quote - news - alert)users, WorkSpace CRM 3.0 enables up to 60,000 records to reside on supported RIM BlackBerry devices, five times that of the previous version and the equivalent of 7.5MB of application data storage. For the typical user, this means that all of their existing Salesforce records will be available on the device's local database from installation.

As with version 2.3, data can be accessed through WorkSpace CRM even when a wireless connection is not available.

WorkSpace CRM 3.0 supports custom List Views, which offer customized ways of viewing the information on the handheld device. Configurable by the administrator or mobile user, List Views allows data to be quickly grouped or viewed in specific formats that are unique to the preferences of a company or mobile user.

According to Sendia officials this allows an administrator to create a custom object in Salesforce, selecting only the data fields that would be relevant to a mobile user to be featured on the device and configure the layout depending on preference and frequency of use.

http://www.sendia.com


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PacificNet Brings CRM To China Telecom

By David Sims

PacificNet Inc., (news - alert) which sells CRM, call center, interactive voice response and other services in China, has announced that its PacificNet Epro subsidiary has been selected by China Telecom's Hainan Branch to provide CRM consulting and call center training services.

PacificNet Epro had previously served China Telecom's main customer service centers located in Jiangsu, Hubei, and Guangdong.

According to a company statement, China Telecom's 10,000 information hotline was, "the first customer service center in China's telecom industry," and was designed to provide basic customer service such as answering customer questions, processing orders, and handling complaints.

Beginning in 2005, China Telecom has been trying to upgrade customer service as a competitive advantage in the increasingly competitive Chinese economy. Face it, a central command economy is great at producing Five-Year Plans but isn't the best incubator for responsive customer service, so as the Chinese consumer economy morphs into a competitive one, where customers actually have options, customer service is seen as a key differentiator.

PacificNet Epro is expected by China Telecom to help Hainan Telecom improve its CRM services with the specific goal of generating profit through CRM services, telemarketing and outbound telesales.

Tony Tong, Chairman and CEO of PacificNet, said that price is no longer the deciding factor for Chinese consumers, who are getting more sophisticated about quality customer service: "Today, customers choose a provider in China not solely based upon price, but upon a number of factors including CRM service, loyalty and retention programs. This trend has created the demand for and deployment of large scale customer contact centers."

Tong called the CRM contact center "the new competitive advantage for the market leaders in China," declaring that "to become a market leader in China, whether as a product or service provider, a company needs to devote resources to CRM and customer service."

http://www.pacificnet.com

WCGS And AmeriVon LLC Team Up

World Communication Group Services, LTD, ("WCGS") (news - alert) a full service Switch Partitioning and Co-Location Provider, in conjunction with AmeriVon LLC, (news - alert) a provider of consumer long distance products and services, have announced the product launch of the "SmartCall" long-distance dialer. This new product targets the prepaid calling card and debit card industries.

The SmartCall dialer is designed to allow dial up phone line users to have competitive long-distance pricing similar to that offered by VoIP providers without the added expense of a broadband connection or equipment.

The SmartCall dialer installs on the users' phone line, requiring no external power source. Once installed, the device will direct all toll and long-distance calls directly to the WCGS network utilizing their custom network software. The customer dials long-distance and toll calls directly without the need to enter special pin codes or access numbers. Customers may have either a prepaid or postpaid account on the network depending upon the rate plan purchased.

"We are very excited about the consumer acceptance of this product during our initial market testing," said Arnie Goodstein, CEO of WCGS. "This product will open new sales channels for distributors of prepaid calling card and debit card products. The innovative software we have designed will produce significantly higher profit margins for the vendor and provide better prices for the end user than traditional calling card products. We are pleased to be associated with AmeriVon LLC and to be the provider of the switching platform for their SmartCall product."

"We are delighted to offer SmartCall to the market place," said Bob Segal, President and CEO of AmeriVon LLC. "We feel that our product offering paired with the switching and network platform from WCGS will provide new opportunities for the prepaid calling card and debit card industries."

http://www.wcgltd.com
http://www.amerivon.com

Concerto Extends VoIP Functionality

By Tracey E. Schelmetic

Contact center solutions provider Concerto Software (news - alert) may have an appellation that is not long for this world, but that apparently hasn't halted innovation on the company's part.

The company, which is in the process of merging with the former Aspect Communications Corp. into a company to be called Aspect Software (news - alert), recently announced that it is extending voice over Internet Protocol (VoIP) functionality in several of its principal products with new releases scheduled to be launched in the next two quarters.

"The flexibility of IP allows contact centers to distribute the work to where it makes the most sense, regardless of geographic region, and allows them to quickly react and be proactive regarding changing business conditions, as well as meet regulatory and compliance requirements without compromising customer satisfaction or cost," said Ralph Breslauer, executive vice president of global sales and marketing at Concerto. "Our focus is to continue providing solutions that enable contact centers to take advantage of those IP capabilities, in an evolutionary fashion, thereby still allowing companies to achieve their strategic business objectives in collections, customer service or sales and telemarketing."

That evolution includes support for the session initiation protocol (SIP) to connect the agent to the contact center and the contact center to the network. The products offering new IP capabilities that will enable customers to increase agent productivity and reduce costs without compromising existing technology investments, customer satisfaction, and agent satisfaction, include:

EnsemblePro, the platform for Concerto Unified Edition and a complete solution offering unified inbound, outbound and blended multichannel contact (voice, e-mail, Web and fax;

Spectrum ACD, which supports high-volume customer interaction by integrating ACD functionality with computer-telephony integration (CTI) applications;

Unison Predictive Dialer, an outbound customer contact product that integrates with existing voice and data systems and offers campaign development tools, predictive dialing, call blending, a browser-based agent desktop, real-time statistics and historical reporting; and

Conversations Predictive Dialer, which provides outbound call management capabilities to improve agent productivity, allows flexibility in managing operational costs and leverages existing technology investments. The upcoming release will offer agent VoIP connectivity via SIP.

"When companies move to an IP infrastructure, it doesn't mean they have to sacrifice features and functionality to gain the cost benefits associated with VoIP. This is one of the reasons we've chosen to integrate SIP, which has matured into the technology that we feel confident offering to our customers, while providing the same comprehensive routing, reporting and monitoring features that are synonymous with Concerto," said Roger Sumner, Concerto chief technology officer.

http://www.concerto.com

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Microsoft In Hosted Model Delivery

By David Sims

In what has to be a fairly closely-watched experiment "Down Under," Australian-based AIE Technologies is announcing the launch of http://www.Software4Rent.Biz, what they describe as "a fully automated, Web-based service where users can rent popular Microsoft programs such as PowerPoint, Excel, Project, etc., on an ad-hoc basis such as by the hour, day, week, etc."

The Web site allows users to use a credit card to rent software for as little as $2 a day and can be accessed at http://www.software4rent.biz.

Garry Ohlson, AIE's CEO said that the service "had taken three years to develop as a response to Microsoft's Service Provider Licensing Agreements of 2002."

The system is built on version two of Microsoft's .Net platform. "It works similar to the way that Citrix works, like a thin-client. Everything runs on a host machine." It's so cheap to run, Ohlson says, his company only needs 20 customers to cover their expenses.

AIE calls their offering Software4Rent, and they're billing it as "a world first — an online, Internet accessed system to rent various Microsoft products and soon other software such as accounting, security, design, CRM etc., on an as-needed basis, on demand, online and always available."

Microsoft Australia licensing product manager Thomas Kablau told ComputerWorld that Microsoft (quote - news - alert) has around 62 direct SPLA partners in Australia that offer Microsoft products on a rental basis.

Based on AIE's CALPIM software asset management engine, Software4Rent allows SME, SoHo and private users to be one-hundred proof license compliant with their software at hosted prices.

"Many people cannot justify what they perceive as a high license cost for their perceived low or private usage," Ohlson said, "so rather than buy it, many individuals borrow or download a copy illegally. Now we can give them the option of renting it online very easily and cheaply, (making software) a utility like electricity, water or gas. You should only have to pay for software when you use it."

Jeremy Horton, AIE's Chief Architect said Microsoft is offering "more and more of its products in this manner, as well as the other major software vendors such as SAP, Oracle and Salesforce.com." AIE itself is considering hosting Adobe products as well.

Ohlson added that this model "has been shown to have little impact on license purchases… A license purchase is not cost justifiable for infrequent users, hence the world's high level of software piracy in the public sector. In fact it is now quicker, simpler and easier to rent the latest and current versions of software legally on demand than go to the trouble of locating software that can be copied."

He credited the adoption of broadband worldwide and specifically in Australia recently for being "the big driver behind this change in consumer patterns."

http://www.microsoft.com

Avaya Inks Alliance With Equant

By Johanne Torres

Avaya Inc. (quote - news - alert) and Equant (news - alert)have announced that they have inked a partnership to jointly deliver IP-based communications. The new agreement named Equant an Avaya Strategic Alliance Partner, enabling Equant to deliver an integrated Avaya-based system to its enterprise customers.

The companies have been working together since March of last year. They will first make available a managed IP telephony communications system designed specifically for multinational companies. The system integrates a layer of Avaya applications with a VoIP VPN service from Equant. The new joint service and product suite offer includes end-to-end support from consulting to implementation and ongoing managed services support.

"The combination of Avaya's IP telephony and contact center applications with the Equant global network makes for a very powerful proposition to the global market place," said Denzil Samuels, vice president and general manager, Avaya Global Managed Services. "The global services capabilities each company brings to the relationship further enhances our ability to respond to needs of our customers wherever they may be in the world."

According to the companies' news release, "Equant is licensed to resell Avaya's industry leading portfolio of products as part of their Enterprise Telephony service in 95-plus countries. In addition, Equant will provide hosted IP communications services for selected customers based on Avaya technology."

"This strategic partnership clearly highlights both companies' commitment to providing first-class communications solutions that deliver business value to enterprises around the globe. As IT executives continue to move their organizations to converged IP communications, they will be looking for business technology partnerships that can deliver the right solutions on a global scale. Equant and Avaya will be at the forefront of this," noted Pierre-Louis Biaggi, head of Equant's Integration Services Business Unit.

http://www.avaya.com

http://www.equant.com

Vonage Activates One Millionth Line

By Johanne Torres

"In a very short time, Vonage (news - alert) has woken up a dormant telecommunications industry and sparked a tidal wave of change," said Jeffrey A. Citron, chairman and CEO of Vonage Holdings Corp., as the company announced that it had activated over one million lines.

News of the milestone follows persistent recent rumors about the possibility of the company filing for an IPO. According to TheDeal.com, the VoIP service provider is planning to file for an initial public offering (IPO), in a move that would help the company raise as much as $600 million.

The company had already managed to raise a whopping $408 million in venture capital rounds led by Bain Capital, with strong participation from existing investors including New Enterprise Associates (NEA), 3i, Meritech Capital Partners, Institutional Venture Partners, in addition to other investors.

Last May, the company announced it had completed a successful $200-million private financing round, and then told a Red Herring magazine reporter that it had no immediate plans to file for an IPO.

"A rumored Vonage IPO would be a huge deal in the VoIP market," wrote TMC president Rich Tehrani. "It is huge because Vonage will be a barometer for the entire VoIP industry and more importantly its success or failure will be amplified a thousands times due to its influence on the companies in this space and the capital markets reaction (or over-reaction) to whatever develops."

Vonage has been busy announcing key partnerships in the VoIP industry. It recently teamed up with cordless phone giant VTech, along with Texas Instruments (TI) to bring the Vonage-ready IP-8100-2 cordless handset to over 8,000 retail locations around the U.S. The devices, designed by VTech and bundled with a VoIP chipset by TI, are configured with Vonage's VoIP phone-based service and retail for $149.99 with a $50 mail-in rebate after 60 days of service. The ip8100-2 is currently being sold at retail locations nationwide, including SAM's Club, Radio Shack, Best Buy, Circuit City, Staples, Fry's Electronics, Office Depot, and CompUSA.

The VoIP (define - news - alert)service provider has also been working with E911 technology provider Intrado to comply with the Federal Communications Commission's (FCC) recent VoIP E911 ruling. The agency ordered that all providers have VoIP E911 access up and running before the end of November. Vonage, which reportedly set $10 million aside to be used solely for its E911 efforts, has been working with Intrado since November last year.

http://www.vonage.com

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By Jim Machi

Video: This Time It's Real. Really.

When I joined the telephony industry, the company I was at (Dialogic) specialized in media processing. Media processing for telephony in those days meant functionality like voice mail, conferencing, fax, and DTMF tones. Even though video was possible at this time over traditional telephony networks, it never really caught on. One problem was the not-quite-real-time video and its jittery, jerky motion. Another huge go-to-market issue was that there weren't many phones able to use this capability, even if it had worked to perfection. So video never really entered the telephony mainstream.

Fast forward a few years to the mid-to-late 1990's as the nascent IP telephony industry was just starting. The protocol of choice at that time was H.323, the protocol used for video conferencing. The IP telephony industry used an existing spec and took only the part of the protocol that was useful for voice. But I, like others, took note of the video part of the spec and wondered if and when video would be combined with traditional telephony media so that the definition of telephony media by default would include video, as it includes voice and fax today.

Fast forward to today. I'm not sure the definition of telephony media will ever include video by default because we now use the term "multimedia" to denote video. The marriage of traditional telephony with data and the coming-of-age of convergence has thrust video to the forefront, probably justifying this new term. Multimedia is alive and well thanks to the use of 2.5G and 3G cellular networks.

The basic use case for video revolves around the cell phone and the presence of cameras on cell phones. I, for one, initially didn't see the point of viewing a picture on a tiny one-inch-by-one-inch screen. I mean, what could you actually see? I guess if you put it really, really close to your eye and squinted at it, it would look like a huge flatscreen. But I never really got it. At any rate, when I needed to upgrade my cell phone to GSM so I could use it overseas when I travel, I got a camera phone since it wasn't that much more expensive.

The marriage of traditional telephony with data and the coming-of-age of convergence has thrust video to the forefront.

Guess what? Yep, I use it. And when people send me pictures, I can actually see them. But better yet, when I send pictures I take with the camera phone via email, they come out pretty good. So we have multimedia communications using the cell network and the data communications network.

This use case is similar to a voice mail application. But instead of voice mail, you have video mail via video play/record. I can also envision the color ring back tones (CRBT) that are so popular today overseas, and are just beginning to enter the US market, morphing into your favorite video playing while the phone is ringing (instead of your favorite song as in CRBT). I can also envision a video portal where you call in to choose content by browsing a menu. So instead of simply reading or hearing the latest score for the New York Mets game, you can actually see the game-winning hit by Mike Piazza.

From a network architecture perspective, two new elements need to be added: the video messaging server and the multimedia gateway. If the video message is staying within the cellular network because you are just sending a video to your friend's cell phone, the gateway wouldn't necessarily be required and you would only need the video messaging server. This would be similar to the network voice mail system that you currently use when you access your cell's voice mail. If someone left you a video message while your phone was off, you could access the video message when it was convenient for you.

From a technology standpoint, if you could treat video as another type of communications data, you wouldn't need to add a unique video messaging server, but could instead add the video capability to the existing unified messaging server. In fact, as Intel develops building block ingredients at the board level and host media processing at the software level for these messaging platforms, that is the approach we are taking.

A multimedia gateway would be required to deal with different types of endpoints that would connect to the mobile network: laptops, PDAs, IP phones, or even standard PSTN (define - news - alert) phones. The gateway would take the video format of the cell network (3G-324M) and convert it to voice with video coders that the IP and PSTN network could then use. For instance, in the case where the video message is coming to a phone on the PSTN network that doesn't have video capability, the video part needs to be stripped out and the voice converted to the standard 64 kbps. In the case where the video message is coming to a client on an IP network (such as your laptop or IP phone), the voice would need to be converted to G.711 (or possibly a compressed voice format such as G.723.1) and the video may need to be converted to a client-based video format such as RealVideo or Microsoft Windows Media if it doesn't support the type used in the cellular network.

Some IP clients might require additional multimedia gateway functionality. For instance, some IP clients would be more capable technology-wise of handling a video message — i.e., a fully equipped laptop can handle more than a PDA. As such, the multimedia gateway would also be required to translate and resize the information, depending on the client. Also, since there are different types of video storage standards in the data world (different MPEG versions, for instance, AVI, RealVideo), the multimedia gateway would also need to be capable of transcoding the formats.

IPTV is another use case for video and IP. This is a very hot topic these days, with analyst reports showing huge growth potential, and it is very different from the types of use cases I described above. IPTV is using the IP network to watch real-time video. In your home, either the PC or your TV (through a set top box) would be the viewing client, or when you are mobile, your mobile client (laptop, PDA, etc.) would be the viewing client.

You might be saying to yourself, "I have cable and that's real-time video, why do I need to watch real-time video over IP"? That's a good question. Think about the parts of the world that don't have well-developed cable networks. Or you might be traveling and want to watch your favorite game. Or you might even think that 157 channels aren't enough, and you want to access more. I don't doubt there is a market here.

While there are interesting business go-to-market issues with IPTV and who the winners will be, the technical challenges relate to Quality of Service (QoS). While there are QoS challenges with VoIP, they pale in comparison to real-time video over IP. The bandwidth requirements are orders of magnitude more than VoIP and the packet-loss requirements are orders of magnitude less. This is one tough application to get right. I am not smart enough to know how the QoS issues will be resolved, but I have confidence they will be (as they were resolved for VoIP). I do believe though that this application will likely force more standards-based QoS discussions, and will likely force some kind of packet inspection at the edge network.

I started this column by talking about video phones and how they didn't succeed in the mid 1990's, even though video was possible over traditional phone networks. Because of the 2.5G and 3G networks' capability to handle video and the fact that cameras are now a normal part of a cell phone, video has entered the mainstream of telephony. The inexorable march towards convergence continues and we now are on the threshold of IPTV, something I only vaguely imagined 10 years ago. There is money to be made here as video use cases continue to develop. Keep reading Internet Telephony to keep up-to-date with the latest developments. IT

Jim Machi is senior director, Modular Communication Product Platform Division, Intel. For more information, please visit http://www.intel.com.

By Tony Rybczynski

Dual-Mode Wireless: The Simplicity Of One, The Power Of The Network

The vision is simple. Bring your office anywhere — ensuring secure, consistent, reliable unified communications anytime, anywhere, and over any device. But a critical business reality for you, as a mobile professional today, is managing the price/performance tradeoffs among the multitude of communications devices and networks at your disposal. You might already be carrying one or more of these devices: a cell phone, a PDA, a Blackberry, a laptop and a pager — and using a variety of networks to attempt to stay relatively connected.

Enter the dual mode mobility device. As a starter, it supports WiFi and next-generation public wireless based on CDMA (define - news - alert) 1xEVDO and GSM UMTS technologies, delivering broadband connectivity across the LAN and WAN. It allows you to have one integrated device in your pocket and leave some of these other devices at home. Whether evolving from a smart phone, telephony enabled PDA or messaging device, a dual mode mobile device comes in various form factors to meet your business needs — supporting voice, data and maybe video; with various screen sizes; with or without a keyboard or keypad; and with a range of accessories. It is built on one of a number of operating systems including Microsoft (quote - news - alert) Pocket PC, Blackberry (quote - news - alert) OS, Symbian OS, and comes bundled with personal productivity tools. While it carries an incredible amount of technology in a small handprint, the power of a dual mode device comes from secure seamless operation with network-based applications such as telephony, e-mail and ultimately unified communications.

The power of a dual mode device comes from secure seamless operation with network-based applications such as telephony, e-mail and ultimately unified communications.

Secure Seamless Operation

There are three aspects to seamless operation that are driving dual mode devices: secure enterprise telephony heading towards Unified Communications, public/private roaming, and managing price/performance.

Unified Communications integrates real-time and near-real-time communications, including telephony, conferencing, instant messaging and presence, and various forms of collaboration (including optionally video) into a rich consistent set of capabilities. A consistent user experience ensures seamless feature operation (e.g., for directory and voicemail access, call logs, call management) across desktop and mobile devices. The lingua franca of Unified Communications is the Session Initiation Protocol (SIP). SIP (define - news - alert) is a signaling and control protocol standard for initiating sessions between and among users, independent of media being used. In this new world, the user is known by a single address, and the network does the rest. In the short term, this could be his or her business phone number; in the longer term it could be a SIP address of the form user@corporation.com. The user, whether on the road or in the office, has a high degree of personal control as to how incoming calls are handled based on caller, media used, and time of day. These capabilities are realized on dual-mode devices by tunneling data and SIP signaling over an IPSec VPN (define - news - alert) connection into the enterprise, independent of the voice path (i.e., cellular or VoIP).

Secure mobility includes roaming from floor to floor in a building or campus, across the city and around the world. WiFi is exploding across the enterprise. Roaming across converged wireless LANs, that support fast handoff and user authentication, is becoming a reality across campus networks. At the same time, WiFi hot spots have emerged in hotels, conference centers, airports, and coffee shops creating an opportunity for remote wireless access using VPN technology. Finally, second- and third-generation broadband wireless services have been introduced and are being expanded across the country. While manually changing from WiFi to public cellular is an option depending on where the user is, seamless roaming implies non-disruptive voice, data, and multimedia session operation with minimum impact on the user as he or she moves from in-building to public environments, and vice versa. Solutions that are providing seamless WiFi/public wireless two-way roaming services are emerging for enterprises and service providers; these are based on some form of 'handoff server,' which acts as a signaling anchor point for SIP sessions.

Managing price/performance is a major concern across the business. A key opportunity enabled through dual mode mobile devices is that they allow users, whenever in the range of a wireless LAN access point, to leverage the higher bandwidth offered by WiFi for IP telephony and data. Given that often the majority of cell phone voice calls are made and received within an enterprise site, there is a significant opportunity to lower the number of cell minutes used, and minimize roaming and wireless long-distance charges. Together, these serve to address the escalating costs associated with the increased use of public wireless services, while optimally leveraging these services and WiFi deployments.

Dual mode mobile devices enhance productivity by enabling just-in-time communications across the virtual enterprise.

Simply And Securely Connected

Dual mode mobile devices enhance productivity by enabling just-in-time communications across the virtual enterprise, whether the user is a campus worker or road warrior. They open the door for seamless roaming between private and public domains and deliver a consistent user experience wherever the user is and whatever the device he uses. They ensure business continuity at the end user level and provide the means to maximize price/performance of telephony, data, and multimedia applications across private and public networks. Would the simplicity of one combined with the power of the network improve your employee productivity and enhance customer service? Absolutely. IT

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By William B. Wilhelm, Jr., Esq. & RA Dr. Axel Spies

Homework To Do In Europe

There is more than mere cleanup work to be done for the thousands of EU officials who return to Brussels after their August vacation period. Politically, the EU Constitution remains in limbo after being rejected by the French and Dutch. Britain suspended legislation to set up a referendum on the new EU Constitution, putting that country at odds with Germany and France, both of whom have called for the ratification process to continue. The political axis of Berlin/Paris is weakened, with Germany facing early Federal elections in September that incumbent Chancellor Schroeder will most likely lose. France is facing the highest number of bankruptcies it has experienced in nearly 10 years and is not very keen to support the current U.K. Presidency of the EU to open the markets "à l'americaine" (American style).

What does this uncertainty mean to the telecommunications industry and, in particular, to VoIP? It remains to be seen whether the anti-EU Constitution trend will actually slow the EU's economic growth. Some European experts argue that that it won't. Of course, it is the European Commission that has been the subject of criticism, rejection, and ridicule for years, particularly in Britain. That situation has created an anti-Commission groundswell that has taken its toll on performance areas that are unique to the Commission: the drafting of legislation and the policing of legislative implementation, i.e., acting as 'guardian of the treaty.' Today, telecommunications is an industry in dire need of the same kind of legislative resolve that marked the early nineties and which created a liberalized and almost single European telecommunications market. Now, legislative initiatives are being shelved. "Certain Member State attitudes," as one hears frequently from European officials, "do not allow strong European visibility at this time."

This combination of national interests and fears could prevent necessary intervention from the European Commission. New areas in telecommunications that could have become gateways to new and sustained growth could, instead be squandered due to Member State resistance. Such is the case with the well-intentioned Voice over IP initiative of the European Commission that has been languishing for months. Telecom is an area in which it is to Europe's economic advantage to adopt common approaches to new technologies, much as it did with the mobile standard GSM. In the 1980s and 90s the European Commission, first and foremost, drove the issues and the agenda, as well as the legislation.

Today, telecommunications is an industry in dire need of the same kind of legislative resolve that marked the early nineties.

Now, others seize the initiative and pass rules that will shape the VoIP (define - news - alert) industry globally for years to come. The FCC's E-911 decision will have a serious impact on many EU regulators who have not yet touched the issue of emergency access and VoIP. In its decision, the FCC does not distinguish between "Interconnected VoIP Providers" that are located in the United States and those that are located in Europe — both are covered. At this time, there is no "European" FCC in sight that can take on the task of developing EU-wide standards for VoIP. Instead, the European Commission is considering expanding the European "Television without Frontiers" Directive that promotes European industry (and suggests production quotas) to include services provided over the Internet. "What does this mean?" one observer joked, "Will we be required by the EU to make every second VoIP call in French?"

At the same time, the European Commission is losing its grip on another issue: mandatory data retention. In an effort to appease law enforcement officials in the wake of the London bombings a few weeks ago, the EU Ministers and Justice are trying to reach a compromise on the EU-wide retention of traffic data. In fact, the "compromise," if it is adopted, does not deserve this name.

It does not unify the retention regime within the EU. Since the individual EU Member States will be authorized to deviate from the general 12-month period that the Ministers of Justice suggest, there will be no EU-wide retention period. This will render it more difficult for U.S. carriers and ISPs to provide innovative EU-wide non-location-based services, such as VoIP. They must store "traffic data" (whatever the definition for this term will be regarding VoIP) much longer than generally needed by responsible operators for business purposes. It is also a question as to whether that "as a first step" only providers of telephony services will be required to store data, while later ISPs would also be covered by this obligation. This "two-step approach" is in contrast with the EU's principle of "technology neutrality" under

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the EU Electronic Communications Framework. It also raises several currently unresolved questions, such as if and when providers of VoIP will be classified as "telephony services." In theory, at least, the EU Council, the body representing the national Governments, could adopt a mandatory data retention measure at its meeting in October, thus bypassing the European Commission. However, if this happens, the European Parliament will likely step in, arguing that the measure needs the consent of the Parliament and cannot be adopted without it. The Parliament has argued that the Council lacks jurisdiction in the area of data retention to adopt retention periods via a Framework decision. If this conflict is not resolved by negotiations, the measure could end up at the European Court of Justice.

What we will probably see more often is a "turf battle" between the EC, the EU Council, the EU Parliament, and the national regulators. After the rejection of the EU Constitution in France and the Netherlands, the balance is tipping more towards the national regulators who might now see themselves in a stronger position. The rejection could also lead to a split between those countries seeking a unified approach to telecom policy and those wanting to go their own way — "Europe at different speeds." This could make it difficult to provide innovative services, such as VoIP, that cross the national borderlines. IT

"Will we be required by the EU to make every second VoIP call in French?"

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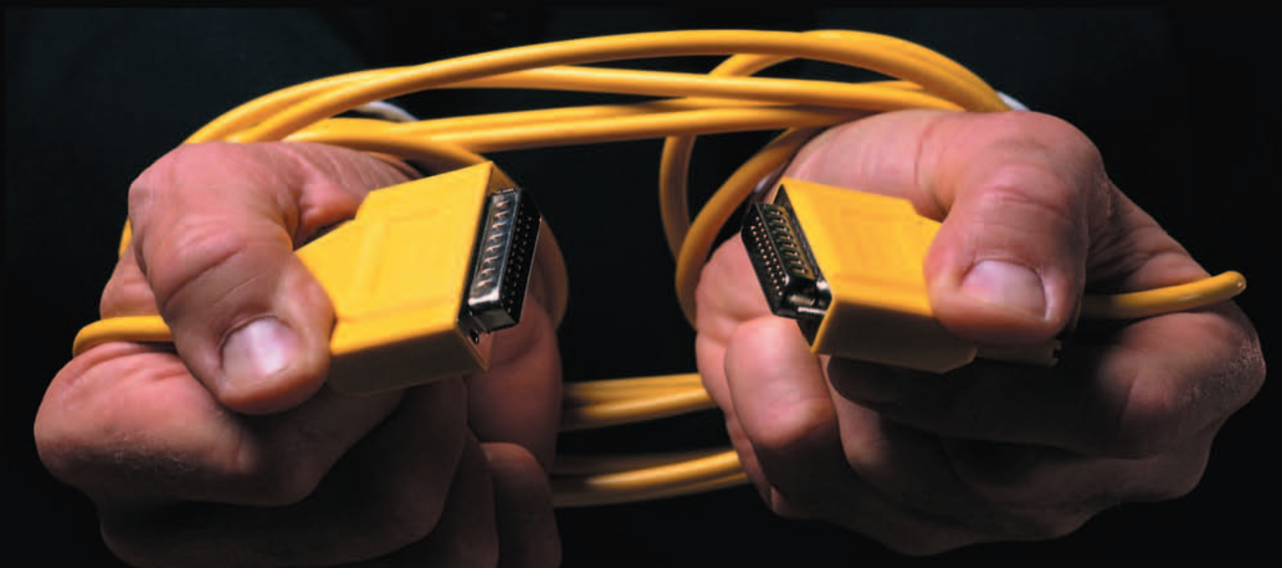

By Hunter Newby

The Classic Peering Model Evolves

The more VoIP changes the telecom business, the more predictable it gets. During a recent conversation with the heads of sales and operations for VoIP peering service provider, Infiniroute, I was reminded of the evolution of the classic ISP peering model. Just as technical necessity and economics drove the ISPs to directly connect through peering points the same holds true for the carriers now using VoIP. Access is the key to the marketplace. Interestingly though, many of the international incumbent carrier networks have been architected to not be openly accessible, but rather have one main gateway typically using H.323 that acts as the interface from their network (and user base) to the rest of the world. Let's see if IP, ENUM (define - news - alert), and SIP (define - news - alert) allow that architecture to survive.

The issue for the incumbents is that they need to move to IP, but don't want to change their revenue model just yet. They can somewhat accomplish this by using H.323 and establishing only direct peering relationships with other carriers on a bi-lateral basis. This design is a product of timing, limitations, and a touch of sunk costs.

Many of the international carriers established these VoIP gateways several years ago when VoIP was brand new as a means to leverage the benefits of IP for the purpose of provisioning voice circuits to other carriers. Back then H.323 was the dominant protocol. That's the timing part — they were early adopters in a sense. The limitation part of H.323 and subsequently the reason why practically everyone in the world is moving to SIP is SIP's ease of interoperation. SIP is a very "lightweight" protocol meaning there is not a lot of overhead, bulky code, or processing involved. H.323 has more of a traditional telecom background whereas SIP comes from the IP world. Most importantly it enables dynamic provisioning through SIP addressing. That makes SIP more efficient, easy to use, and ultimately cost effective.

This is not lost on the incumbent carriers either. What is very telling is that most of their internal architecture is based on SIP because it makes their own network easier for them to manage. So why the H.323? The lack of dynamic provisioning on the carrier/customer level for buy/sell relationships makes it difficult to connect, route, and reroute to them and requires in many cases personal negotiation for call termination on their networks. This keeps them in control and their wholesale business from being further commoditized. For most non-incumbent wholesale carriers SIP is dominant because they need to make it easy for others to connect to them since the middle-man business is so cut throat and margin-less these days. SIP is the next step in the path to ENUM and once that point is reached devices can communicate directly with each other. Eliminating long-distance puts a dent in the incumbent revenue model.

The sunk cost in the H.323 equipment for the incumbents is somewhat of a factor for things remaining the way they are as the incumbents don't want to remove it if it is generating revenue and works properly, but the provisioning limitation provides a convenient "technical reason" why open-access can't really happen and subsequently protects the revenue.

Keeping the past alive in this way presents an opportunity for Infiniroute, which provides route optimization over public and private networks as well as signaling information for direct routing between H.323 carrier gateways and H.323 to SIP conversion. Their model works because the incumbents own the end user access and therefore the traffic to and from those users. Getting all of these VoIP islands to directly connect and avoid any unnecessary hops enhances quality and lowers costs. Infiniroute's direct routing facilitates the layer 7 aspect of classic ISP peering by identifying the gateways and then it looks for the optimal media path for transport.

"Once the signaling request is made our system determines the best way to route the call based on quality, available capacity and cost," states Neal Axelrad, VP Sales for Infiniroute. "This may be over the public Internet, which is our strong point, but many of the incumbents already have direct connections between each other and they are typically Layer 2 Ethernet. Where quality is concerned a private network connection is usually always preferred, but we'll optimally route them either way."

SIP is the next step in the path to ENUM.

That says a lot about how ALL network operators think.

These private peering connections between the big incumbents are very similar to the "Seven Sisters" of ISP folklore. They privately connect, cut special deals, and don't open their networks. Although this pattern is repeating now in

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VoIP the fact is that it might not be that easy to keep change from happening, or even slow the pace. All of the technology the VoIP carriers are using is readily available for any medium to large enterprise. That coupled with pure-play transport (mainly Ethernet) carriers with no Layer 3 service model and all of the pieces are in place. This means the enterprise user base can step around the carrier, build their own VoIP network, peer with other enterprises and the minutes (and revenue) go away. It's not that difficult, it's already happening and as it begins to really take hold the addressable market for Infiniroute and other VoIP Peering service providers will expand by several orders of magnitude. Anyway, let's face it, back in the good old days of IP transit (1995) a wholesale meg of IP was $1,000. Today in 2005 it's $10. Oh well, so much for protecting the top line revenue with a closed network model.

In the grand scheme of evolution, whether it's VoIP, or a living species, eliminating inefficiencies is nature's way.

In the grand scheme of evolution, whether it's VoIP, or a living species, eliminating inefficiencies is nature's way. We are all tied to a logical genetic structure in what we create as it is the way we were created. As long as we are able to put aside conflicting political and legacy economic pressures to resist the evolution the end result will benefit society. Just because it is good for humanity though doesn't mean it's good for the incumbents' bank accounts, but in the end no one can stop the VoIP volcano from permanently changing the landscape.

Mismatched protocols and creating VoIP islands is not the optimal path. What this global voice organism wants is seamless interconnection with all of its parts, but will the enterprise VoIP wide-area networks eventually be interconnected to each other and eliminate that part of the business of the IXCs in the process? Since many enterprises are basically starting with a clean slate and have no interest in generating revenue from voice as they see it as an expense, there is a very good chance that that will happen. It's all a matter of time. IT

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By Ben Guderian

Getting Physical With WiFi Telephony

The most unique aspect of WiFi telephony compared with wired VoIP is the affect of using the airwaves as the physical media for carrying packets to and from the wireless telephone devices. A lot of technology goes into making that wireless connection as transparent to the user and the host telephone system as possible, but there are a couple of considerations and tradeoffs to deal with in choosing a specific WiFi network implementation based on the physical radio standard: 802.11a, 802.11b, or 802.11g.

These Physical Layer (PHY) standards describe the radio frequency (RF) spectrum used and how it is partitioned, the modulation type, and power levels. Some of these can be factors in selecting a particular WiFi product, such as the RF spectrum used, while others are only important to product developers to ensure interoperability, such as the modulation scheme used to turn binary data into radio waves. The specific implementation details aren't important, except for the ones that affect interoperability, performance and capacity.

The initial 802.11 standard had three PHY options: two using spread-spectrum radio technology, frequency hopping (FH) and direct sequence (DS), and a third using infrared optical technology. Both radio standards used the 2.4 GHz band, which was available for use without licensing restrictions nearly worldwide. The optical approach never took off so the standard now rests peacefully in the standards graveyard.

Even while the initial 802.11 standard was being ratified, work began on standards to increase data rates beyond 2 Mb/s. Two parallel efforts were initiated. The 802.11a Task Group took the approach of moving up in the RF spectrum to where bandwidth was more plentiful and could support data rates above 50 Mb/s. Meanwhile, the 802.11b Task Group stuck with the 2.4 GHz band and looked to build on the DS standard to increase data rates to greater than 10 Mb/s. There were also some efforts to improve on the FH standard outside of the 802.11 committee, but it never caught on primarily due to the widespread industry support of the 802.11a and 802.11b efforts. Following on to these was the 802.11g Task Group, which leveraged the higher speed technology of 802.11a for use in the 2.4 GHz 802.11b spectrum, which offered a lower cost alternative to 802.11a with backward compatibility with legacy 802.11b client devices.

The 2.4 GHz band used by 802.11b/g has a couple of advantages. First, it is now available throughout the world for WiFi networks, although it is also available for other consumer and enterprise wireless products such as cordless phones and RFID (define - news - alert) tags. The other advantage is that radio signal propagation is pretty good at 2.4 GHz, both in free space but also going through walls. But the biggest limitation of the 2.4 GHz band is that is only has about 80 MHz of radio spectrum available which is partitioned into three non-overlapping channels. Adjacent WiFi access points need to use different channels to prevent interfering with each other, so having only three channels to work with is a network design challenge, particularly in multi-floor, three-dimensional deployments.

On the other hand, above 5 GHz where 802.11a plays, there is enough radio spectrum available to support twelve non-overlapping channels. That makes deploying an 802.11a network much easier in terms of avoiding co-channel interference, but the tradeoff is that radio signal propagation is worse at higher frequencies, so the coverage area for an 802.11a access point is usually less than that of an 802.11b access point. But having more channels allows for denser network deployments and lessens the chance of running up against interference from a neighbor's WiFi network. Most enterprise WiFi infrastructure vendors support both 2.4 GHz 802.11b/g and 5 GHz 802.11a PHY standards, and in many cases both are available in a single access point (AP). Besides the advantages or disadvantages associated with the RF spectrum used, what is the impact of the higher data rates supported by 802.11a and 802.11g for WiFi telephony?

The initial 802.11 standard had three PHY options: two using spread-spectrum radio technology, and a third using infrared optical technology.

Counting Calls

The actual bandwidth used by a telephone call over a WiFi network depends on several factors, starting with the codec used to generate a digital data stream just as with any VoIP implementation. The codec determines the number of bytes that will be carried in each packet for each sample period. For example, full-rate G.711 coding which turns an analog voice stream into a 64 kb/s data stream requires 80 bytes for every 10 ms sample, while G.729 compression at 8 kb/s requires only 10 bytes in a 10 ms sample. Some efficiency can be gained by sending multiple samples in a single packet, reducing the amount of overhead by reducing the number of packets. The tradeoff with sending

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Most vendors can't extend your contact center technology resources to service remote contact center locations and, at the same time, enable your local business units to manage their own technology-driven business processes. As a result, multi-site organizations are challenged to gain economies of scale by sharing IP-based infrastructure and software licenses across sites.
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more information in packets less frequently is that it can affect voice quality by adding delay and echo, and the affect of lost or dropped packets is much more significant.

The overhead required to transmit a voice packet over the wireless LAN and through the wired IP network adds significantly to the packet size. Working our way up the OSI model, headers are required for the 802.11 media access control (MAC) and the logical link control (LLC) portions of the Data Link Layer, followed by the IP header for the Network Layer, and finally the Transport Layer which typically uses something like User Datagram Protocol (UDP) under Real-time Transport Protocol (RTP). Stripping off the 802.11 MAC layer, a wireless VoIP packet looks like any other VoIP packet once it hits the wired network. All together, the various higher layer headers added to a voice packet can more than double the packet size.

WiFi APs and client devices are designed to drop back to lower data rates as the signal quality degrades. This is advantageous for maintaining WiFi coverage into every nook and cranny, but it also means that more of the AP's bandwidth is used up when the data rate drops, since it takes longer to transmit the same amount of information. Most AP's allow the minimum data rate to be selected to prevent low-rate wireless connections from hogging bandwidth, but this requires the deployment to be done such that devices can always operate at the minimum rate or above, no matter where the user is.

Differing from wired Ethernet protocols, which detect and deal with packet collisions, WiFi uses a carrier sense multiple access/collision avoidance (CSMA/CA) protocol to minimize the chance of two devices transmitting over the air at the same time. This protocol requires devices to wait a certain amount of time and then listen to determine if the radio channel is free for them to transmit (referred to as listen before talk). These waiting periods are based on a fixed time followed by a randomly selected time within a certain range, and WiFi quality of service (QoS) mechanisms such as WiFi Multimedia (WMM) assign shorter waiting periods to higher priority applications such as voice and video. So what does this have to do with bandwidth utilization? Well, the waiting time has the affect of wasting available bandwidth. For example, even if just one WiFi device is associated with an AP so there is no chance of another device transmitting at the same time, that device still has to wait the time prescribed by the CSMA/CA (define - news - alert) protocol. A typical waiting time of 50 microseconds is equivalent to transmitting 550 bits at 11 Mb/s, or about 70 bytes. These timing gaps reduce the effective bandwidth of the AP and need to be taken into account when calculating the actual call capacity for WiFi telephony.

That is why it is more accurate to quantify voice traffic in terms of percentage of bandwidth utilized instead of packet size or data rate. This gives a more realistic measure of the maximum number of simultaneous calls supported by a single AP. For example, taking into account the total packet size and timing gaps gives us around five percent bandwidth utilization for G.711 at 11 Mb/s with 802.11b. So the theoretical maximum number of calls per AP for this example is 20. But that doesn't account for real-world conditions where collisions happen, users roam around and need to handoff to other APs, and other devices need to use the WiFi network. A more realistic maximum takes into account the headroom needed for these things, so capping the real-world maximum capacity to 12 simultaneous calls at 11 Mb/s is more reasonable. But if the WiFi coverage isn't good enough to maintain the 11 Mb/s data rate, the bandwidth utilization goes up significantly. At the minimum 802.11b data rate of 1 Mb/s, a G.711 call uses more than 20 percent of the AP bandwidth in an 802.11b network, dropping the traffic capacity to only three or four simultaneous calls.

Translating Calls To Users

Telecom managers know that you typically don't need a call resource for every telephone. An enterprise PBX has fewer trunk lines for outside calls than the number of telephone sets installed. The number of trunks is calculated using a probabilistic model of telephone usage that is based on the acceptable chance of a caller wanting to make an outside call and a trunk not being available. The theory behind telephone traffic planning has been around since the early days of the Bell System, and provides a means to determine the actual number of users that can be served by a WiFi access point based on the maximum number of simultaneous calls it can handle. Traffic analysis for WiFi telephony is a little more complex though for a couple of reasons. First, different geographic area may have different traffic requirements. Users may congregate in certain areas such as cafeterias and smoking areas, so the traffic engineering may vary by location and anticipated usage patterns. Second, WiFi telephone users will probably have higher than average usage profiles, at least for the initial users since their communication needs were sufficient to justify the investment in their wireless handsets in the first place.

There are a few different traffic models available that vary based on the number of users and how blocked calls are handled. A good source for more information on telephone traffic models, including online calculators, can be found online at http://www.erlang.com. Of course, the results from the calculations are only as good as the data you put in, so you need to have a good sense of what kind of traffic load to expect from WiFi telephone users.

Choosing which flavor of 802.11 radio technology requires looking at several factors, not the least of which is the bandwidth requirements of both wireless voice and data applications. Most enterprise WiFi telephony applications can be supported by the most widely deployed 802.11b standard, although its maximum data rate of 11 Mb/s may be insufficient for multimedia and other high-bandwidth data applications. Deploying an 802.11a WiFi network may require replacing existing wireless client devices, since most WiFi enabled laptops and PDAs only recently started offering 802.11a radios. A safe bet chosen in many enterprise WiFi deployments today is to opt for dual-radio APs, offering the high-capacity advantages of 802.11a along with broad device support using 802.11b/g. And leveraging that investment with WiFi telephony continues to deliver improved productivity, responsiveness, and mobility of employees in all kinds of enterprise applications worldwide. **IT**

Ben Guderian is vice president of market strategies and industry relations at Spectralink. For more information, please visit the company online at http://www.spectralink.com.

By Mark Collier

Voice over IP Security

As an application running on the shared IP network, VoIP inherits the network security issues common to IP, including viruses, worms, Denial of Service (DoS), eavesdropping, man-in-the-middle (MITL) attacks, etc. Attackers can now attack the voice application from locations on the shared IP network. This is true even if Virtual LANs (VLANs) are used, which only provide basic separation. Listed below are several reasons why VoIP can be vulnerable to attacks:

• **Multiple Networked Components** — VoIP (define - news - alert) requires multiple networked components, including IP PBXs, media gateways, IP phones, etc. An attack on any component affects the VoIP application.

• **Complex Protocols** — VoIP is controlled through various network-based protocols, which exchange signaling information to support call control. This includes proprietary protocols, H.323, MGCP, and SIP(define - news - alert). A large VoIP deployment may use several of these protocols. The greater the number of protocols in use and the more complex and dynamic a protocol is, the greater the chance its implementation has vulnerabilities.

• **Weak Authentication** — Most VoIP systems do not employ strong authentication. Without strong authentication, attacking software can use spoofing and other techniques to execute a wide variety of attacks.

• **Platform Vulnerabilities** — VoIP components run a variety of operating systems and supporting services — and can be attacked through these underlying services. This is especially true for IP PBXs, which often run a general-purpose operating system such as Windows or Linux, a database server, a Web server, and other software.

Vulnerabilities

The reasons above and its presence on the shared IP network, bring VoIP the following vulnerabilities:

• **Signaling DoS** — Every component of a VoIP system must process signaling exchanged over the shared IP network. If the software implementing the signaling has implementation flaws, it is vulnerable to DoS attacks, including transmission of malformed packets and request floods.

• **Media DoS** — VoIP media (audio), normally carried by RTP, is vulnerable to attacks that congest the network or slow the ability of a phone or gateway to process the packets in real time.

• **Voice SPAM** — Voice SPAM, or SPIT, occurs when an attacker sends many unsolicited calls to an enterprise. These calls may go directly to users or voice mail.

• **Vulnerable IP Phones** — IP Phones (and softphones) are the most common component in a VoIP deployment and can be a challenge to secure. IP Phones often come with network-based access enabled and no/weak passwords.

• **Eavesdropping** — Most VoIP media is not encrypted during transmission. An attacker who has access to the network segment where media is transmitted or who can execute certain types of MITL attack, can access, gather, and playback the media.

• **Firewall Issues** — VoIP creates a number of issues for traditional firewalls. VoIP uses separate connections for signaling and media. The media sessions use random ports, which is an issue for firewalls because these ports are normally closed. VoIP also embeds IP addresses in signaling messages, which are ignored by traditional NAT. Firewalls also add latency to VoIP media packets and tears down calls if they fail. Finally, traditional firewalls do not monitor all of the VoIP protocols for the many types of attacks that can occur.

Recommendations

While a VoIP network has vulnerabilities, the threat of exploitation can be managed by following these recommendations:

• **General Security** — Develop policies, maintain strong physical security, follow best practices for securing an IP-based service, monitor resources for new vulnerabilities, maintain patches, remove unneeded network services from all components, enable and review logs, and use standards-based security add-ons (TLS and SRTP) where possible.

IP Phones (and softphones) are the most common component in a VoIP deployment and can be a challenge to secure.

• **Secure the Network** — Build a fully-switched network, use VLANs for basic voice/data separation, use switches to provide the first line of security, and VPNs to secure traffic traveling over an untrusted network.

• **Secure the IP PBX** — Use secure operating systems, remove/lock down all network services, control administrative access, use host-based intrusion prevention, and use network firewalls/intrusion prevention systems.

• **Secure IP Phones** — Use phones that offer strong security, use strong passwords, disable unneeded network

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access, and secure the firmware upgrade process.

• **Deploy VoIP-Optimized Firewalls** — Monitor signaling and media for attacks, mitigate SPIT, provide call admission control, and perform protocol-aware NAT (define - news - alert).

VoIP's evolution will require a focus on VoIP security.

Conclusion

VoIP deployments will have vulnerabilities depending upon scope, vendor selection, configuration, and deployment scenario. Because VoIP deployments are still uncommon, generally separated from other data applications, use closed/proprietary protocols, and do not yet exchange VoIP with a public network, the threat of an attacker actually exploiting the vulnerabilities is moderate. This will change however, as more VoIP is deployed, as it becomes more closely integrated with other data applications, and protocols such as SIP are used to communicate with a public voice network. VoIP's evolution will require a focus on VoIP security, including deployment of VoIP-optimized firewalls at key network locations. **IT**

Mark Collier is the chief technology officer of SecureLogix Corporation. SecureLogix (news - alert) will participate in the "Reseller Live" Internet Telephony Conference & EXPO panel scheduled from 1:00–4:00 p.m. on October 26. If you would like to submit a question for the panel, please send to: maxschroeder@tmcnet.com.

TalkSwitch's Jan Scheeren

This is the first installment of TMCnet's "Executive Suite," a monthly feature in which leading executives in the Voice over Internet Protocol/communications industry will discuss their company's latest developments as well providing analysis on industry news and trends.

Technology Marketing Corporation President and Editor-in-Chief Rich Tehrani and assistant editor Ted Glanzer interviewed TalkSwitch (nee Centrepoint Technologies, Inc.) CEO Jan Scheeren last week.

Ottawa-based TalkSwitch (news - alert) manufactures affordable, all-in-one telephony systems for small and multi-location businesses. Products from its award-winning TalkSwitch line, which includes a hybrid PSTN/VoIP PBX, range between $695 and $1,795.

After a 30-minute conversation with Scheeren, an Australian native who is given to peppering his analysis with surfing analogies, one thing is clear: he understands the big picture.

Indeed, there is a refreshing pragmatism about Scheeren, which he revealed while discussing myriad topics, which ranged from the difficulty in identifying TalkSwitch's competition to his philosophy on hardware manufacturing.

Scheeren's down-to-earth approach is reflected in his company's recent decision to change its name from Centrepoint Technologies, Inc. to TalkSwitch (http://www.talkswitch.com).

To be certain, the decision was not taken lightly, considering that Centrepoint was founded in 1990 — well before the prevalence of the Internet and search engines — to provide small businesses with affordable telecommunications services.

Traditionally, Scheeren said, such solutions were beyond the reach of small and multi-location businesses for two reasons:

1. They were difficult to install, configure, and maintain; and
2. They were cost prohibitive.

Scheeren said that at the time of the company's founding, ILECs appeared "apathetic" to the small business telecommunications market, while almost everyone else viewed it as "saturated."

Nevertheless, Scheeren said that he saw an underserved market. Specifically, Scheeren honed in on the lack of innovation in the small business telephony space.

"At the end of the day, in the last 20 years, there had been no drop in price and zero improvement in terms of capability," Scheeren said.

Enter Centrepoint.

"[The name] represented what the company did — voice communications for small offices," Scheeren said.

However, once the Internet exploded and the Yahoo!s (quote - news - alert) and Googles (quote - news - alert)of the world took over how people obtain information, Centrepoint had a major problem. More accurately, the company had a major problem with the name Centrepoint.

"People had a hard time finding us," Scheeren said, noting the difference between how "centre/center" is spelled in the English-speaking world.

Also, according to Scheeren, changing the company's moniker made sense because the vast majority of resellers and consumers identified Centrepoint more through its popular TalkSwitch product line.

"Eighty-percent of the market saw us as TalkSwitch anyway. There is nothing unique in what we did."

What is unique, however, is how Scheeren identifies TalkSwitch's competitors. The way he sees it, TalkSwitch's biggest competitor isn't a specific entity, but the market in general.

"There are all kinds of competitors and it's an obvious question," Scheeren said. "The bigger factor is that our opportunity and specialty has been small businesses with 2 to 32 telephone users per company location."

Until recently, Scheeren viewed the competition as whoever was providing single-, two- or three-line connections to small businesses, and, perhaps, Panasonic Wireless.

Then along came a little thing called VoIP, which is turning telephony as we know it on its ear.

"The biggest thing… [VoIP] is doing is opening up the dial tone business to competitors," Scheeren said. "It creates all these new channels, and that's enormous. In the past, our competition was Panasonic or the phone system companies. Currently, it's hard to see the competition because it hasn't opened up yet. In the VoIP space, multi-location competition and opportunity are related to a point."

Regardless of their identities, Scheeren said that TalkSwitch separates itself from its competitors — present and future — with its longstanding market presence.

"We are a lot more established," Scheeren said. "We've got many thousands of customers [approximately 20,000], with resellers signing up at a rate of 150 per month. It's just enormous."

To continue to build on this success

for the future, TalkSwitch must figure out when the migration from the PSTN (define - news - alert)world to IP will be completed or, as Scheeren puts it, "when switches will actually go away."

"It won't happen overnight."

Just don't try to pin him down to a time frame. Five years? Ten years? How long?

"Pick any number, the forecast will change; I don't know," Scheeren said. "For any business, there is the need to get from here to any mark. We still need to get to the next quarter, and the quarter after that and the quarter after that."

Which leads us to his surfing analogy. "We picked our beach, we are working with our surfboard, we're paddling out, looking for the breaking wave," Scheeren said, adding that TalkSwitch is similarly looking for that break in the market. "For us, it's timing and structuring."

As for the immediate future, Scheeren offers a pragmatic view.

"The challenge right now is making money in the short term," Scheeren said. "We've been successful. We've created a hybrid solution that works equally well with VoIP and the switched world."

"Our challenge has been a lack of distribution channels."

Currently, TalkSwitch distributes its products through resellers and directly to end-consumers through its Web site.

"TalkSwitch has a service provider department mandated to go after that business," Scheeren said. "Timing is a big question regarding the incumbents and the ITSPs. We haven't seen which way that will play. We work with all of them. The incumbents of course are a tough nut to crack and, to a large degree, are not fully committed yet [to migrating to IP] or at least not publicly yet."

For TalkSwitch to continue to thrive in the small business market, Scheeren said that the company also must continue to tie the customer to the vendor and vice versa.

"The secret to me is, at the end of the day, in the small business market, churn is going to be a big thing. Customer retention will be [the] key. That's a channel we want to be plugged in to."

Toward that end, TalkSwitch does not cut corners on the production end. Indeed, the company designs from scratch its own hardware; it does not license anything from its competitors.

"It might cost more and take longer in developing," Scheeren said, "but at the end of the day, you have control, and control over your hardware is important."

Also, Scheeren has shied away from bottom line, cost-saving manufacturing measures, choosing instead to produce TalkSwitch's wares in North America and Malaysia.

Proof is in the pudding. TalkSwitch, according to Scheeren, boasts a return rate of less than one percent on its hardware.

"At the end of the day," Scheeren said, the consumer is "going to get the best rate of return on the hardware." IT

Pingtel SIPxchange SIP PBX

CT Labs was asked by Technology Marketing Corporation to evaluate the Pingtel (news - alert) SIPxchange PBX product. CT Labs (http://www.ct-labs.com) performed the tests and evaluation on the Pingtel SIPxchange PBX in the CT Labs Rocklin, CA facility, concluding the effort with this report. This report appears here in abbreviated form — the full report appears online at tmcnet.com/173.1.

Executive Summary

The SIPxchange PBX (define - news - alert) product was found to be easy to install as supplied to us in its typical form as a turnkey solution. CT Labs was able to set the system up and get it working with the preconfigured desktop phones in about 30 minutes. It was slightly more difficult, however, to get the SIP softphone to register with the SIPxchange server. Overall, though, the installation was found to be straightforward.

The SIPxchange features were found to be comprehensive for a basic SIP PBX. While this was not the focus of this test, we understand that the next release (due out in October) will include many new features, including Meet-Me Conferencing and ACD features. The sidebar entitled *Pingtel Version 3.0 New Features* provides a list of the new features that will be announced at the Internet Telephony Conference & EXPO in Los Angeles (http://www.itexpo.com) on October 24.

The documentation and integrated help were found to be well-organized, easy to comprehend, and provided much helpful information. The administrative and client GUIs were easy to use with a simple structure and clearly labeled options.

Our manual testing of features and functionality went well, with calls between the variety of supplied desktop and softphones connecting properly and providing a high level of voice quality.

The excellent performance results from our automated call-handling load tests showed that the SIPxchange PBX can easily handle heavy call loads of station-to-station as well as trunk-to-station calls.

Overall, the CT Labs found the Pingtel SIPxchange to be a solid, dependable system that is easy to install and use, and provides a solid level of performance. Product evaluation scores of the CT Labs evaluation of Pingtel's SIPxchange can be found in Table 1.

General Goals Of The Evaluation

The "guiding light" questions that we are attempting to answer as a result of performing the test procedures and analysis for this product are as follows:

1. How easy is it to install and set up (a) the PBX server components; (b) a client workstation; and (c) an administrative console?
2. How easy is this system to administer and maintain from (a) a local console; and (b) a remote console?
3. How easy-to-use are the product's graphical user interface programs (administrative as well as client programs, if available)?
4. How do the integrated voice messaging and auto attendant features compare with typical enterprise voice mail systems in terms of feature depth?
5. How do the supported SIP Phone devices perform with respect to perceived speech quality and call handling?
6. How effective and helpful is the product's documentation package?
7. How effective and helpful is the product's on-line help for the product's graphical user interface programs?

Vendor Equipment Provided

The vendor was asked to provide CT Labs with the following:

- A fully turnkey solution with processor and memory support chosen to correctly match the production requirements for the trunk and station line density of the unit provided.
- Support for standard auto attendant and voice mail features.
- At least three desktop SIP phones and one Pingtel softphone.
- Telephone line support requirements:
    - Trunk-side lines. Systems with trunk line support for T-1 CAS (robbed bit).
    - Station-side lines. Support for a minimum of 32 stations.
- Provide CT Labs with optional installation and setup assistance by a trained system installer for one to two business days.

Test Setup

Pingtel supplied CT Labs with a turnkey SIPxchange system installed on a Dell PowerEdge SC420. This was connected to a Cisco Catalyst 3524-PWR XL switch, which was used for connecting all the internal phones and our workstation (used to access the administrative and client Web-based

GUIs). An AudioCodes TP260 gateway card was already installed in the Dell server to provide us with T1 trunks using ISDN PRI signaling.

For the automated call-handling tests, an Empirix Hammer FX-TDM was connected via T1 to the Audio-Codes gateway on the Dell server so that the Hammer FX-TDM could place PSTN (define - news - alert) calls into the system. An Empirix Hammer FX-IP was also connected to the system via Ethernet through the Cisco switch and registered as 96 SIP station phones. For the station-to-station tests, the

Pingtel Version 3.0 New Features

The following list of features was provided by Pingtel as new features that will be implemented in the SIPxchange system in their next release (version 3.0, due out in October, 2005). According to CT Labs, the implementation of these features should "move the Pingtel system up a notch from basic PBX solution to full featured PBX solution."

Reflecting the increased functionality, PingTel is renaming the solution "ECS" or Enterprise Communications Server, which will feature the option of installing all the components on separate serves, greatly increasing scalability. There will also be a Business Edition, which comprises a single server installation. All existing Pingtel customers will receive the feature upgrades as part of their ongoing subscription.

PBX Features

Call pick-up. The call pick-up feature allows a user to dial a feature code and retrieve a call that is ringing at another phone or group of phones.

Call park/retrieve. This feature will allow users to park calls. The user would then page or notify a user which extension the call is parked on. The call can be retrieved by dialing the call park extension. Each call park extension can have a unique music on hold source or greeting that the administrator can import.

RHEL 4 Support. SIPxchange will support Red Hat Enterprise Linux 4.0 with this release.

Scheduled Backups. Administrator can schedule backups of system configurations and/or voice mail messages daily or weekly.

Auto Attendant Features

Multiple Auto Attendants. Administrator can create up to 100 auto attendant menus. Menus can answer calls with unique greetings and call routing rules. Menus can be nested to-gether to create custom call flows.

Voice Messaging Features

Voice Mail Distribution Lists. Voice Mail users can create custom distribution lists.

Change PIN with TUI. User can now change their voicemail and web portal PIN through the TUI. Previous versions would not allow this.

Administrative Interface Features

SIP Phone provisioning. This release adds provisioning support for Snom and Grandstream phones

User Configuration. Enhanced user management by sorting by User ID, First Name, Last Name, Alias or Group.

Client Interface Features

Soft Attendant Console. The Soft Attendant Console is a PC application that is used for managing calls. The application includes softphone capabilities and can operate on a MS Windows platform with a USB headset.

Meet-Me Conferencing Bridge Features

Secure Conferencing. Administrator can set up a PIN for the Conference Organizer and the Participant.

Configuration Options. Each conference can be configured to start or not start without the Organizer.

Entry and Exit Tones. Tones are played when each caller enters or exits.

Mute. Each participant and organizer can mute and unmute by dialing a code.

Configurable with SIPxchange PBX or stand-alone. Bridge can be installed on same server with SIPxchange PBX or installed by itself with a third-party proxy server.

ACD Features

Intelligent Routing. Ring All, Linear, Circular and Longest Idle is supported. This is programmable on a per queue basis.

Welcome Greeting. A greeting is played to caller before they enter the queue. This greeting can be disabled or can be interrupted as soon as an agent is available. This is programmable on a per queue basis.

Queue Message. A message or music is played to callers that are in queue. This is program-mable on a per queue basis.

Agent Sign-in/Sign-out. Agent can sign in or out through a feature code, Web interface or custom API

View of all callers in queue. An API is provided to view all calls in queue. Caller information like ANI, DNIS, time in queue, ringing etc is available.

Overflow. Call can overflow to a secondary destination. The destination can be any extension or SIP URL. Each queue can overflow the last call in to queue or the call that has been queued the longest.

Hammer FX-IP used 48 of its registered SIP station channels to call the other 48 registered SIP station channels on the Hammer. For the trunk-to-station tests, the Hammer FX-TDM placed PSTN calls into 23 of the T1 trunk lines on the system and placed calls to the 96 registered SIP station channels on the Hammer FX-IP.

Installation And Configuration

A total of 30 minutes was spent connecting up all the items. From there the server was started up and station calls were placed between the pre-assigned Cisco and Polycom desktop phones. About an hour was then spent installing the Pingtel SIP softphone and registering it with the Pingtel server. While it was simple to install and set up the Pingtel SIP softphone for peer-to-peer mode, it took longer to get it set up as a registered phone on the Pingtel system. After trying to accomplish this using the Pingtel SIP Softphone documentation, CT Labs had to ultimately contact our Pingtel support representative. The rep then walked us through the registration process using the Web-based interface on the softphone itself, which was not an option that we saw in the documentation.

Pingtel noted that the Installation Guide is mainly used by resellers and system integrators, although some large enterprise customers prefer to install the software on their own networked Linux servers.

Overall, we rated the installation and configuration of the SIPxchange an 8.5.

CT Labs found this product to have a good feature set comprising all the features that most small businesses would need in a SIP PBX (with the exception of call park/pickup).

Overall, the current set of Pingtel SIPxchange features was rated an 8.7.

To read the full report online, please visit http://tmcnet.com/173.1. IT

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

Table 1.

Product Evaluation Score (each rated out of a possible 10)	Product Score	Relative Weight (%)	Weighted Score
Ease of Installation & Configuration	8.5	10	0.85
Features Evaluation	8.7	10	0.87
Product Documentation, Online Help	8.9	10	0.89
Graphical User Interfaces (Admin And End-User)	8.9	20	1.78
Manual Product Exercise	9.4	25	2.35
Automated Call-Handling Load Test (No VM)	9.6	20	1.92
Technical Support	9.6	5	0.48
	Total	100	9.14

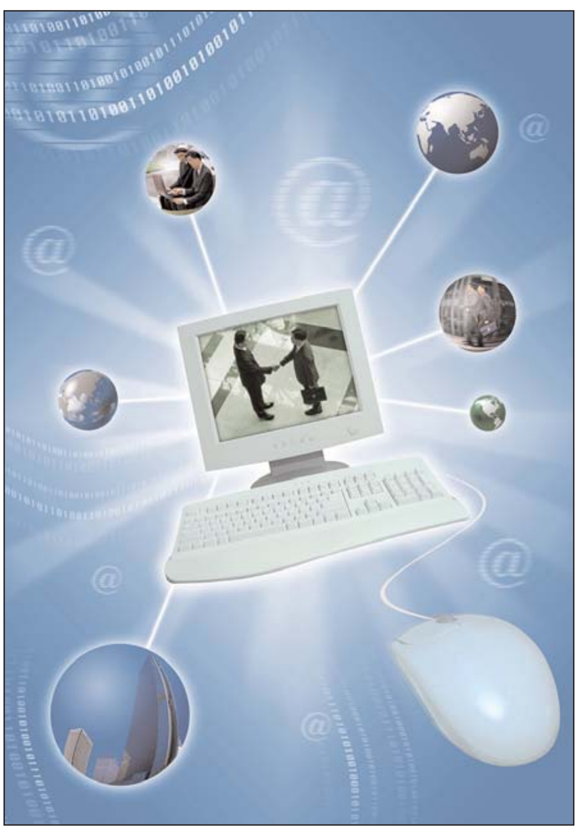
Table 1: Product Evaluation Scores. The weighted score is calculated by multiplying the score for a given category by the relative weight (e.g., a score of 6 times a weight of 10% would yield a weighted score of 0.6). Note that the maximum weighted score for a given product for the sum of all evaluation categories is 10.

IP PBX Vendor Approaches:

There's A Wrong Way… And A Right Way

Contrary to what some vendors in the IP telephony arena would have you believe, not all IP PBXs are alike. And the fundamental reason they aren't comes down to software versus hardware. On the software side, vendors who targeted the IP market early on started with IP telephony as a core base, and followed the applications and open standards path that IP communications were intended for. Most major proprietary telecom vendors, however, tried to launch their IP strategies from the same rigid hardware architectures they've been recycling for years — and for the most part are still fighting an uphill battle.

While the hardware strategies from proprietary vendors haven't been entirely fruitless, they haven't exactly reached the level of functionality for IP telephony that software has. Nor will they. Hardware simply doesn't offer the IP flexibility that open standards software does, and in fact often requires bolting on additional servers to fit a proprietary IP PBX into a voice over IP network. So it's somewhat surprising that the proprietary folks claim their IP offerings are just as functional and easy to implement as the ones from the innovators offering software and standards-based solutions. You should therefore look under the hood to determine exactly what you're getting beyond dial tone and any claim that says "Of course it's IP-enabled."



Here are some critical issues to consider in choosing your organization's IP PBX solution, as well as the vendor you get it from.

Migrating to IP telephony over time. In an ideal technology world, implementing an IP infrastructure works best if it's a straight-up installation enterprise-wide. Let's say that's the right way — yet not always realistic since most organizations that already have TDM-based PBX phone systems can't afford to just take them out of commission and start from scratch: Cost-wise, or downtime-wise.

The best approach to migrating to IP telephony over time, then, is to develop a migration plan that blends existing traditional TDM (define - news - alert) resources and IP to get started. Scheduling your migration in phases for various departments, workgroups, etc. also lets IT teams work out issues on a small scale and set a better stage for extending voice over IP across an entire organization. Which brings us to a wrong way and right way for IP PBX vendors.

It's imperative to work with a vendor whose IP telephony offering allows you to install a new IP PBX on traditional telephony interfaces and ultimately migrate to VoIP with no forklift upgrades. Moreover, a vendor should offer VoIP (define - news - alert) for the telephony interface using SIP for all incoming calls and stations, or using SIP only for stations still connected to the PSTN (define - news - alert) via traditional trunk interfaces. With SIP and potential forklift upgrades playing such crucial roles, the right way to go is with a vendor whose IP telephony solutions are open and application-driven, since the SIP standard itself is software-based and open. Conversely, while most proprietary vendors advertise open solutions for VoIP, they're still trying to adopt SIP to their traditional hardware products to leverage IP networks.

Open standards, or the same old proprietary approach? If a vendor is indeed trying to adopt SIP into a lineup of traditional hardware products, it's safe to say they're also taking a traditional proprietary approach to VoIP. That means such vendors will likely try to lock you in to using add-on hardware recycled for VoIP, and to costly end-user devices and soon-to-be-extinct PBX products. Which, again, is the wrong approach for IP communications.

Getting back to SIP (define - news - alert) as a prime example, standards-based IP PBX software is made for SIP, in that the SIP standard is software-based, open, and lightweight so it can support IP phones along with soft phones, analog phones, desktop PCs, and mobile devices. Yet because most proprietary vendors are still struggling to support the SIP standard in their hardware, they often try to compensate by saying SIP lacks sufficient interoperability for VoIP as an inherent flaw in the SIP standard. But the truth is, SIP's relative simplicity makes interoperability much easier than with older protocols such as ISDN that are infamous for system compatibility problems, which has driven IP technologies themselves to become primarily software-based since calls are directed to application servers on a data network the same way as e-mails, Web chats, and other media.

Applications, applications, applications. Along with real-time voice communications, SIP also provides the perfect blueprint for text messaging and application sharing. In conjunction, VoIP enables a business to use applications that share information between the data side and the telephony side using a common network and infrastructure. But if the applications aren't developed and available, your business will definitely miss out on the potential of an IP PBX phone system and multimedia applications that go well beyond traditional voice mail and IVR.

In a wrong-way multimedia applications scenario, major proprietary vendors have had to

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acquire media applications such as fax servers, IVR systems and e-mail response management products from smaller vendors. Select an IP PBX from such a vendor and your business will often have to purchase all those proprietary hardware applications separately for IP-based multimedia capabilities. You also take the risk that your new multi-box systems won't pre-integrate to one another, let alone to your data systems — leaving your business with no multimedia routing flexibility and much higher implementation costs.

With pre-integrated IP telephony application suites such as those from Interactive Intelligence and its Vonexus subsidiary, you can take the right approach to IP by replacing disjointed communications hardware with a single software-based application server on your network. Rather than proprietary third-party hardware, your business is able to handle all media types using one set of business rules integrated to your business systems. The depth of functionality in IP software also leads to less implementation for more features, like inbound multimedia ACD (define - news - alert) ; voice response; pre-integrated screen pop; fax server; Web services; recording and quality monitoring; multi-site routing; and even outbound predictive dialing.

IP PBX system administration. Re-read that last sentence … and consider that all those applications and their features are managed with a single administrative interface and IP network for voice as well as data communications. Simplified system administration is another way a software-based IP PBX does IP telephony the right way. And going one step further, the IP application suites from Interactive Intelligence and Vonexus take a Microsoft-centric approach to IP network management via SIP to streamline administration even more. On the other hand, proprietary hardware and third-party applications for IP telephony tend to cobble together administration processes, even with a voice and data network, since each hardware box often requires its own admin interface.

Co-existing with TDM systems located at other locations. As we said earlier about migrating to VoIP over time, finding a balance between an existing TDM-based PBX phone system and a new IP infrastructure is the key to early IP communications success.

Choose your vendors wisely.

It's no different for an organization with branch offices.
Again, in order to marry a new IP PBX and existing TDM systems, especially in multiple locations, it's important to choose a vendor whose IP offering can be installed on traditional telephony interfaces and migrated to VoIP later without forklift upgrades, no matter the location. Also remember the importance, and flexibility, of SIP. Vendors who take the right approach to their IP solution allow distributed organizations to use SIP for offices where stations are still connected to the PSTN via traditional trunk interfaces, and still extend VoIP to the telephony interface for incoming calls and new IP stations in offices that make the move to IP first.

An IP PBX equipped to leverage SIP also makes it easier to bring branch offices together over a SIP-enabled LAN or WAN — and in a single administrative interface — whether offices are using IP or TDM systems. Or, you can always take your chances with proprietary IP systems that attempt to bring multi-site offices together with third-party applications, separate administration interfaces, and a converged voice and data network that most times isn't converged at all.

VoIP for more than just cost savings. IP PBX solutions for VoIP can certainly lead to cost savings such as lower long-distance costs via toll bypass and reduced maintenance expenses over traditional PBX equipment. However, there are other tangible benefits to consider in making a business case for VoIP; for instance, its ability to make a mobile workforce more responsive from wherever they are. Many IP application suites offer features such as real-time presence management, Find-Me/Follow-Me, Web-based voice mail access and other availability tools. Also unlike proprietary hardware, IP telephony makes it easier to add functionality by simply adding applications, and also to customize and automate interaction processes for customers. In short, user and system capabilities that simply weren't available before IP communications came along.

Consider the right factors, not the wrong vendors. If a new IP PBX and voice over IP is on your business's radar, one essential factor can make your decision a right one or a wrong one: choosing flexible standards-based IP application suites from a true IP PBX vendor, or boxes of rigid hardware that some proprietary supplier is trying to convince you is "open" and equipped for VoIP. Because IP telephony and the SIP standard take a decidedly software-oriented approach — and because most proprietary vendors are still trying to adopt SIP to their traditional hardware for IP — deciding on the right vendor should be easy. Just look for one that combines the multimedia IP software innovation with the experience of being an IP industry leader to bring the least risk and best value to your business.

Joseph A. Staples is Senior Vice President of Worldwide Marketing for Interactive Intelligence Inc., and the company's wholly-owned subsidiary, Vonexus Inc. As 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 along with Vonexus is a leading industry innovator in the IP telephony, VoIP and SIP movement. For more on their suite of IP telephony, contact center and enterprise solutions, contact Interactive Intelligence at 317-872-3000 (http://www.inin.com) and Vonexus at 888-817-5904 (http://www.vonexus.com).

Things To Consider

- **Develop a migration strategy.**
- **Keep an eye towards open standards and interoperability.**
- **Make sure applications are fully integrated.**
- **Seek simplified system administration.**
- **Don't forget: You may need to integrate with legacy TDM systems.**
- **Choose your vendors wisely. There's a right way, and a wrong way.**

Announcing The Internet Telephony® Excellence Awards

The editorial staff of Internet Telephony magazine is proud to announce the winners of the first-ever Internet Telephony Excellence Awards. These companies are as varied as there are products that fit under the IP telephony umbrella. But the one thing that binds them all together is that they all excel at delivering solutions that solve their customers' needs.

As varied as the products are the companies that sought out these solutions. Medical outfits like the Detroit Medical Center. Not-for-profit groups such as AMIDEAST. Financial institutions, Universities, ILECs, small businesses, foreign mobile carriers, VoIP service providers… the list of companies implementing VoIP (define - news - alert) into their everyday life is as impressive as the list of award winners who supply them the tools they need to help their businesses grow and run smoothly.

Winning companies each submitted a case study, illustrating how their clients were able to take advantage of the product in question. These case studies documented the installation at the client company, and included background on the client company, their particular problems, and the reasons they chose the award-winning product or service, as well as the results.

The VoIP space is hot these days, and it's easy to get caught up in the hype associated with such a buzzing marketplace. However, the winners of the first-ever Internet Telephony Excellence Award are not merely "me-too" players. These companies all have proven products and services, and most importantly, customers that are willing to speak up and offer themselves as references.

And as I always like to remind readers whenever we publish any type of awards, always check out those customer references. These days, that's becoming ever easier to do. IT

– Greg Galitzine

COMPANY	WEB	PRODUCT NAME
AccessLine Communications	http://www.accessline.com	SmartVoice Service
Alcatel	http://www.alcatel.com	OmniPCX Enterprise
Aspect Communications	http://www.aspect.com	Aspect Uniphi Connect
Atreus Systems	http://www.atreus-systems.com	Atreus xAuthority
Brix Networks	http://www.brixnet.com	BrixCare Self-Service
Brooktrout Technology	http://www.brooktrout.com	SnowShore IP Media Server
Cisco Systems, Inc.	http://www.cisco.com	Cisco Wireless IP Phone & Cisco Aironet WLAN
CopperCom	http://www.coppercom.com	CopperCom CSX
EagleACD	http://www.eagleacd.com	EagleACD
Elluminate	http://www.elluminate.com	Elluminate Live! Version 6.0
Emergent Network Solutions, Inc.	http://www.emergent-netsolutions.com	E-REV (Emergent Residential Enterprise VoBB)
Empirix, Inc.	http://www.empirix.com	Hammer NXT
Ensim Corporation	http://www.ensim.com	Ensim VoIP
FacetCorp	http://www.facetcorp.com	FacetPhone
Intertex	http://www.intertexdata.com	Intertex IX67 Including SIP Switch Functionality
Interwise, Inc.	http://www.interwise.com	ECP Connect
Juniper Networks	http://www.juniper.net	M-series Routers
Lucent Technologies	http://www.lucent.com	Lucent's IP Telephony Assessment Solution

Table of winners continues on next page

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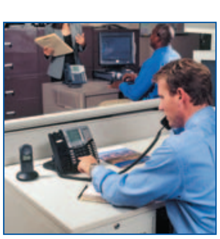

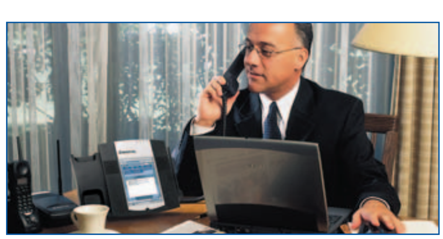

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COMPANY	WEB	PRODUCT NAME
M5 Networks, Inc.	http://www.m5net.com	M5's Outsourced IP Phone System
MERA Systems	http://www.mera-systems.com	MERA VoIP Transit Softswitch
Meru Networks	http://www.merunetworks.com	Meru WLAN System
NetCentrex Inc	http://www.iplay3.com	U-Play3
New Global Telecom	http://www.ngt.com	6DegreesIP — Turnkey Wholesale VoIP Services
Nominum	http://www.nominum.com	Nominum Caching Name Server
Nortel	http://www.nortel.com	Communication Server 1000
Nortel	http://www.nortel.com	Nortel Multimedia Communications Server 5100
Pandora Networks	http://www.pandoranetworks.com	Worksmart
Pingtel Corp.	http://www.pingtel.com	SIPxchange
Protus IP Solutions	http://www.protus.com	Protus Virtual Fax
Psytechnics, Inc.	http://www.psytechnics.com	PESQ & Psytechnics Speech IP Monitor
Siemens Communications, Inc.	http://www.enterprise.usa.siemens.com	HiPath OpenScape
Siemens Communications, Inc.	http://www.enterprise.usa.siemens.com	HiPath ProCenter Suites
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Vodavi Communications Systems, Inc.	http://www.vodavi.com	XTS-IP with PathFinder and Discovery ManagerPlus
Vonexus	http://www.vonexus.com	Enterprise Interaction Center (EIC)
Witness Systems	http://www.witness.com	Witness Systems IP recording solution
XO Communications	http://www.xo.com	XOptions Flex
Xten Networks	http://www.xten.com	eyeBeam

A Conversation With Aculab's Alan Pound

Alan Pound
Founder and Managing Director
Aculab

Back in June, Aculab announced the release of its next-generation board, Prosody X. Designed around an IP core, the card offers many innovative benefits to developers. I ran into Alan Pound, Aculab's Founder and managing director at the top of the Sears Tower back in June, and that led to some follow-up discussions, which eventually led to this interview you now have before you. Aculab will be exhibiting their latest and greatest at this month's Internet Telephony Conference & EXPO in Los Angeles at the LA Convention Center.

– Greg Galitzine

GG: What is happening in the VoIP marketplace today?

AP: As VoIP (define - news - alert) goes mainstream with the proliferation of broadband access for consumers and enterprises alike encouraging the development of new IP-based services, there is a need for a new, IP-centric offering from within the communications building block vendor space.

Telcos — (news - alert) national and international carriers, both fixed and mobile — are under increasing competitive pressures from a variety of sources, which is pulling them towards VoIP and other IP-based service offerings as a means of differentiating themselves and, perhaps not to put too fine a point on it, ensuring their future. Service providers, including the telcos, are in a similar position, with innovative competition being the dominant factor — change is the only constant, and never was this adage more apt.

The landscape is changing, with access-independent VoIP providers challenging the old paradigm — having turned the hype into reality — and creating new opportunities. Peer-to-peer networking has emerged as an alternative for communications and, miles away from geeky consumer VoIP, is potentially going to revolutionise the enterprise applications space. There is a new ecosystem evolving, in which developers should get busy. Their innovation is needed to think up and deliver the new offerings that consumers will buy and the new productivity services that enterprises will choose to spend their money on.

Aculab (news - alert)is continuing to innovate and to introduce new products like Prosody X that will provide developers and systems integrators — and the development organisations of the telcos and service providers themselves — with the platforms with which to build these new offerings.

GG: Aculab has recently launched Prosody X. Tell us about it.

AP: Prosody X is an open standards circuit card designed for media processing in IP-based networks with TDM connectivity as an option. It offers a new approach to card design that acknowledges the reality of IP in today's telco network infrastructures. Designed around an IP core, its architecture makes the product distributable amongst different chassis platforms offering resilience and scalability as well as helping future proof solutions as they move to IP. Preliminary channel counts for fully configured cards with basic speech processing are running at 600 for the PCI product variant. Its rich mix of basic and advanced speech, fax, and data features coupled with the price makes Prosody X an extremely attractive option for any developer looking to create large scale VoIP applications from simple IVRs to complex blended contact centers.

Prosody X highlights:

- On-board Ethernet means IP is inherent in the design;
- Ethernet connectivity not only conducts VoIP traffic, but also enables the control of cards in remote chassis offering benefits of resilience and scalability;
- Rich media processing DSPs give large channel counts for VoIP, transcoding and media processing tasks like IVR, echo cancellation, conferencing and fax;
- With the Aculab SIP Bridge there is no restriction on the number of SIP signalling channels available for third-party call control;
- Optional connectivity for up to eight E1/T1 trunks with digital network access protocols from CAS to ISDN and SS7 (ISUP and TCAP).

GG: What is this 'new approach to card design' that you mention?

AP: The fundamental change behind

the next generation hardware of Prosody X is the *principle* around which the product is built. It is constructed around an IP core, while maintaining the ability to (optionally) interface with E1/T1 connections, which are still in widespread use.

IP is, therefore, inherent in the design — there are no PCI drivers and the card appears to the host PC as a NIC — and this architecture means it is the ideal platform for creating large-scale, cost effective, revenue generating applications for solution providers and telcos offering VoIP-centric products.

Along with the change in environment goes a parallel change in the available DSP technology we have used to construct our next-generation hardware. Using the latest DSP families (created with VoIP in mind) allows the choice of devices that are, by design, application-friendly to both VoIP and rich media processing applications.

Notwithstanding this, there continues to be a need to interface with traditional telephony connections like E1 or T1. Previous open standards telephony hardware was designed around these legacy interfaces with VoIP as an add-on, but Aculab's next generation of hardware has a beating heart of IP while continuing to service E1 or T1 connections by clever modular design.

Prosody X leverages Aculab's core expertise in combining complex technologies into a powerful and flexible proposition that assures developers a simple, clear migration path. Prosody X offers 'mix and match' modules to scale a wide range of applications including low and high densities suitable for enterprise and public networks. Together with higher densities and lower costs this is most certainly a compelling proposition.

GG: Can you expand on what the Aculab SIP Bridge is?

AP: SIP (define - news - alert) adoption is increasing steadily, as expected. Many solution providers are looking to leverage the power and capabilities that SIP can bring — more efficient use of communications hardware and software being a key driver.

With Prosody X, solution providers can take advantage of a number of SIP enhancements reflected in the Aculab API, presenting greater flexibility in application design — namely third-party call control.

In an IP environment, in instances where calls are simply being connected between RTP endpoints (User Agents), only signalling is required, for example, by a server-based IP PBX. Phone calls between people may not require any media resources (unless they need to be recorded, or an intelligent device needs to 'listen' into the call). It is only when a call is being terminated by an application, such as voicemail, that media resources are needed. This means that some systems will need more signalling channels than media channels and to be able to handle third-party call control to manipulate connections.

Aculab's new SIP implementation, SIP Bridge, makes it possible to use Aculab's highly integrated SIP protocol stack in a much more powerful manner — to directly implement an IP call center or an IP PBX, for example, using third-party call control.

With some media processing resource card vendors, although signalling and media are separated, a signalling channel will depend on a media channel on a 1:1 basis. With Aculab's Prosody X card and SIP Bridge, this dependence is removed, presenting the ability to more effectively manage and allocate resources, with third-party call control becoming an exciting option.

GG: What sets Prosody X apart from other products available?

AP: Alternative products offer a fragmented set of functions focused toward one application or one application environment only, Prosody X is a new benchmark offering the ultimate flexibility to developers who can select the functions required to scale a wide range of low- to high-density applications. It is a highly configurable card that combines Aculab's proven media processing resources, IP telephony and TDM (define - news - alert) digital network access functions and offers unprecedented value per channel. With an unlimited number of SIP call control sessions available, 600 channels of rich media processing resources and the option to add up to eight E1/T1 trunks, the product's feature set is unparalleled in the market.

> The landscape is changing.
> — Pound

GG: What are the benefits of Prosody X for developers?

AP: Customers and prospects have been requesting extreme density and a broad range of functionality. By being selective with the component technology employed, we have been able to bring such a product to market at a barely believable price when compared to competitors — so the component cost of their large scale solutions is set to shrink.

The possibility is opened up not only to create large-scale VoIP applications like IVRs and unified messaging, etc., but also compete with large scale proprietary products like ACD and contact center systems by building equivalent solutions with Prosody X.

Developers now have a high-density, multi-functional building block at their disposal allowing them to add functionality to platforms without necessarily adding more hardware — thus moving to a more software centric approach.

Remember that the number of these cards that can be built into a given solution is no longer limited by the size of a host chassis since additional cards can reside in a lower cost expansion chassis, while remaining under the control (via Ethernet) of the host processor. IT

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Hybrid IP: The Best Of Both Worlds

By Greg Galitzine

When considering the move to IP telephony, enterprises have a number of choices to make, not the least of which involves the decision to migrate slowly towards a VoIP (define - news - alert)solution while retaining applications and business processes, or rip out the existing phone system and go completely IP right out of the gate.

According to a recently published Dell'Oro Group report, IP telephony has become the technology of choice in the PBX market. Sales of IP PBX (define - news - alert) and Hybrid IP/TDM PBXs are forecasted to reach $6.1 billion in 2009, an 11 percent compound annual growth rate. Dell'Oro Group projects IP PBXs and Hybrid IP/TDM PBXs, which represented only half of the lines shipped in 2004, will comprise almost 88 percent of lines shipped by 2009, taking share from traditional TDM systems. "Hybrid IP/TDM PBXs are currently the most popular since they allow businesses to take advantage of IP telephony capabilities, while preserving their existing investment in TDM-based lines and digital telephones," said Steve Raab, Director of IP Telephony Research at Dell'Oro Group.

Infonetics recently released a report as well, where they predict that worldwide hybrid PBX revenue will nearly triple between 2003 and 2008, growing to 69 percent of the total market.

Many companies that are looking to VoIP enable their communications systems are still wary of the relative novelty of the technology. They feel that a measured approach to IP telephony is best. Rather than face the enormous cost and uncertainty of ripping their current telecom solution out completely, and replacing it with a relatively unproven solution from a vendor that might not traditionally be considered a telecom specialist, customers are opting for a more conservative approach. By deploying a hybrid IP PBX solution, customers can retain many of the features of their existing digital/TDM-based systems while leveraging the advantages VoIP systems have to offer.

Of course, pure IP proponents point out that in a hybrid design, IP telephony is essentially bolted on to a TDM architecture; therefore, only the IP-enabled endpoints have access to the rich capabilities of IP Communications.

But still, advantages abound.

A key benefit is the ability to migrate an enterprise one group or department at a time. This lowers the risk associated with a potential catastrophic systems failure, allowing a business to slowly ramp up its comfort level with a new technology. Also, as enterprise employees learn to take full advantage of an IP telephony system, leveraging all the so-called soft benefits such as efficiency and application integration, future departments will be more willing to embrace a new telecommunications system if their colleagues have had a positive experience.

Corporations can retain their existing applications, (e.g., call center or enterprise directory), and use IP trunking between locations to extend the applications to far-flung workers or home-based teleworkers, thus allowing them to reap the same benefits as their office-bound colleagues.

Companies do not have to spend their fortunes on retraining employees to use new equipment or to learn new processes that are part and parcel of deploying any new technology, especially systems as important to the day-to-day operation of a business as a phone system.

IP telephony has become the technology of choice in the PBX market.

And what about all those existing analog phone sets and fax machines? What about lobby phones or courtesy house phones in hotel hallways? In a pure IP system, you would either have to replace them all with expensive IP sets, or be forced to scrap them altogether. You could of course install gateways to translate TDM (define - news - alert) to IP and back to TDM if need be, but the addition of costly conversion equipment is simply a non-starter for many businesses.

When considering the move to IP telephony, companies are faced with myriad decisions. Today's research shows that hybrid IP telephony solutions are increasingly the more popular choice. If you are in the market for a new phone system, you should seriously consider a hybrid IP telephony offering from any of the number of vendors that provide such a solution.

Greg Galitzine is the editorial director of Internet Telephony magazine.

Laying The Groundwork For IP Telephony

By Eric Yerina

You've studied the promising ROI for IP Telephony and researched vendor solutions that appear to fit your company's needs. If you've done your homework, you also know that merging voice and data networks into one integrated network is a complex undertaking that requires careful analysis of existing infrastructure. An objective evaluation of network capabilities is essential to any IP Telephony implementation for several reasons:

• Knowing the **existing traffic demands** on your data and voice networks will help you determine Quality of Service (QoS) requirements and potential increased bandwidth needs for your new IP Telephony network.

• Detecting and resolving **underlying network issues** such as bottlenecks or frequent retransmissions of data packets will clear the path for a smoother implementation.

• Having an accurate picture of your **current network topology** points to areas that require improvement prior to implementation.

• Establishing a **baseline** of existing network performance offers a solid basis of comparison against future expected levels.

Areas Assessed

The success of a new IP Telephony implementation depends on the capability of its existing network components. From the data network side, every device that is expected to carry application traffic should be tested and benchmarked. These include routers, switches, servers, gateways and VPN.

More often than not, these devices are not IP Telephony-ready and will require upgrading, modifying or replacing. Testing confirms and validates this need. This critical first step is to determine what kind of traffic patterns occur throughout the day, as peak utilization times might warrant increased amounts of bandwidth to support IP Telephony. Other factors studied are type of users and traffic, their priority, packet types, application usage and transmission problems.

Maintaining good voice quality is a key consideration in VoIP implementations, so it is important to know whether your current communications system and voice gateways can adequately handle the converged network's traffic. To assess this, a voice quality-testing device measures the network's current voice quality. VoIP traffic is then injected and measurements are taken again to determine any improvements that must be made to ensure good voice quality in the new converged environment.

The Assessment Process

While it is beneficial to know what voice quality issues exist with IP Telephony prior to implementation, it is equally important to analyze the health of the data network — since this is the platform on which VoIP will run, and ultimately depend on. Conducting voice quality tests will tell you how much delay you have, but not where network congestion — a major culprit behind packet loss — is occurring. Knowing what issues are occurring on the data side is as important as understanding those on the voice side, since each impacts the other.

This assessment process begins with gathering detailed information about your network infrastructure. Business expectations are also important to note and figure prominently in the subsequent recommendations. After information has been gathered, the Network Engineer will conduct and document a physical survey of the site(s) to determine whether the physical environment, i.e., cabling, circuits, security, is helping or hindering the current network in its journey to IP Telephony.

Using a protocol analyzer, testing of various network segments (LAN, WAN) for variables such as peak usage, dropped packets and retransmissions provide additional details of the network. Statistical reports are then generated and analyzed for causes of network issues such as congestion or packet retransmissions.

At this point, voice quality tests of the existing voice infrastructure are conducted to establish a baseline against which to measure IP Telephony performance. VoIP traffic is then injected in the network flow for measurement and detailed statistical reports are generated for analysis.

Using the combined information collected from interviews, site surveys and test reports, a final assessment report is issued, containing analysis, findings and recommendations, outlining any necessary improvements in network infrastructure that should be made prior to implementation.

The decision to outsource the critical services of a Network and IP Telephony Assessment is an important one. There are many factors to consider including: Does my staff have the tools and expertise required to perform the assessments? Does my staff understand voice technology? If I want to build the expertise in-house, have I budgeted for the expenses associated with building the expertise?

When making this decision, consider your business requirements, timeframe to deploy and internal core competencies. Deploying a successful IP Telephony solution for your organization will depend as much on the planning and services you select prior to the implementation as the IP Telephony products you choose.

Eric Yerina is the General Manager, Services Marketing, NEC Unified Solutions, Inc. For more information please visit http://www.necunified.com.

INTERNET TELEPHONY EDITORS' CHOICE AWARD 2005

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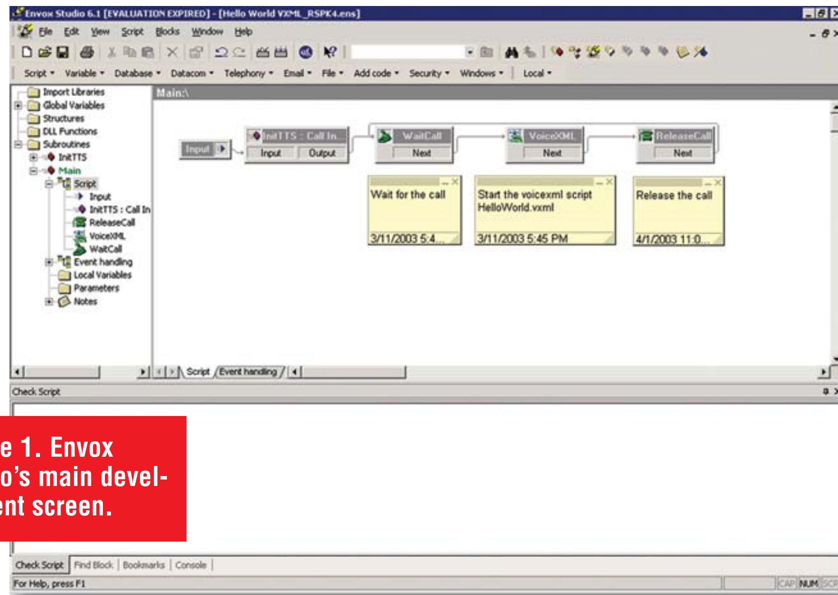

Figure 1. Envox Studio's main development screen.

TMC Labs has been testing application generators for over 10 years; many — such as Artisoft's Visual Voice, Parity Software's VOS, and others — have come and gone. Through all that time however, Envox (news - alert) has continued to penetrate the market, improve their product, and even acquire competing app-gen companies such as Show N Tel.

Each time we examine Envox's communications development platform we are impressed at how much more functionality they add and how easy they make it to develop VoIP (define - news - alert) applications, IVR/autoattendant/voice-mail applications, voice portals, unified communications, mobile Internet, and more. Heck if they make it any easier to develop complex telephony applications we may as well tell all programmers to pack their bags and move to India. Oh wait, that's already...never mind!

In any event, Envox 6.1 is an open, standards-based communications development environment that supports VoiceXML, speech, Web services, and VoIP capabilities all on a single platform. We took Envox 6.1 for yet another test drive and one again we were very impressed.

First we examined the user interface of Envox 6.1 to see if they made any usability improvements over prior versions. Once again it had the same friendly drag-and-drop capabilities. All of the various blocks are broken into menu categories letting you quickly find and place a new block on the screen. The categories include: script, variable, database, datacom, telephony, e-mail, file, and more. The development screen also featured the ability to put "yellow sticky notes" next to each block describing what it does (Figure 1). We also liked that the development screen featured context-sensitive help, which allowed us to double-click a block to open it, then put our cursor into a field we had a question about, next press F1 (for help), and the appropriate help screen was automatically displayed.

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

You have two choices for development. You can use the renowned Envox Studio 6.1, a proprietary development platform that supports just about any standard you can think of, or you can develop standard VoiceXML applications using Envox VoiceXML Studio. We developed several applications and tested them. We should point out that Envox includes several sample examples both for Envox Studio 6.1 and Envox VoiceXML (define - news - alert) Studio which made it much easier to get up to speed in developing applications of your own.

After you create your application, you simply assign the application to one of your licensed channels from within Envox Console (Figure 2). We should mention that Envox has both a hardware and software simulator that lets developers emulate telephony hard-

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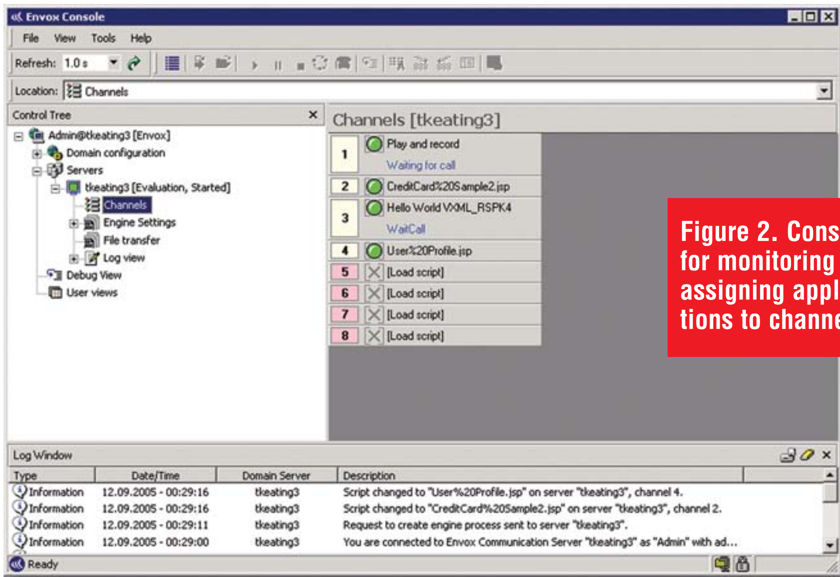

Figure 2. Console for monitoring and assigning applications to channels.

ware/software without having to invest in hardware in order to test their applications. We tested the emulators and they worked quite well. In fact, we set up a VoIP application to accept incoming SIP (define - news - alert)calls and then used the SIP-based SJPphone softphone client to call the Envox application server using an IP address. The application server answered and played the prompts and accepted our DTMF digital clicks entered via the softphone client. We also tested Voice XML applications, which executed via an application server hosted by Envox, and these applications too performed flawlessly.

Envox claims that their graphical programming environment that reduces development time by 50 percent or more versus coding by hand. The Envox suite has extensive debugging and logging capabilities. You can set breakpoints in your application, which is useful to debug specific parts of your application. You can then step through the application and view various variables and parameters. Source code version control allows the use of Visual Source Safe or CVS directly from the development environment to facilitate change management and version control in large, collaborative development projects.

Finally, extensive logs and a hierarchical tree-structure for browsing the logs made debugging applications a breeze.

Envox has an extensive feature-set. It features an embedded VoiceXML browser for executing VoiceXML scripts, numerous integrations for leading telephony and speech products, a runtime environment that supports up to 240 ports per server, support for Web services, and powerful management tools for easily configuring, monitoring, and managing large-scale deployments. The built-in speech APIs offer native integration with ScanSoft and Nuance speech recognition, text-to-speech, and speaker verification technologies. Envox supports a plethora of communications and IT standards including ODBC, XML, SOAP, WSDL, HTML, HTTP, TCP/IP, SNMP, LDAP, SS7, USSD, RADIUS, X.25, IMAP4, POP3, SMTP, and more. In addition, you have the ability to integrate existing IT assets into voice solutions by calling Web services or external applications, hosted scripts and DLLs written in C, C++, VB, Active X, JScript, and VBScript.

Envox supports remote, secure administration across LAN/WAN via TCP/IP. It also supports SNMP agents and alerts. In addition, the company also provides a Watchdog Service so you can monitor parameters you select for fault tolerant operation. Fault tolerance is further enhanced via their domain server, which lets you choose a primary and a backup server.

Other features include:

- H.323, SIP;
- T1; E1; IP; SS7; analog; USSD;
- Station Channels;
- ISDN (BRI/PRI); QSIG; DPNSS; R2;
- Intel Global Call 4.2, 4.0;
- Intel Host Media Processing 1.1 FP1, 1.1; MP3;
- Intel Dialogic SR 5.1.1 FP1 and SR 6.1 PCI for Windows.

Speech-related

MS SAPI 4

Speech Recognition products supported:

- Nuance 8.5, 8.0 SP3;
- ScanSoft OpenSpeech Recognizer 3.0, 2.0;
- ScanSoft PhoneticOperator 5.8;
- ScanSoft SpeechPearl XML 1.1.

Text-to-speech products supported:

- Nuance Vocalizer 4.0, 3.0;
- ScanSoft RealSpeak 4.0, 3.51 CS;
- ScanSoft Speechify 3.0, 2.1, 2.0.

Speaker verification products supported:

- Nuance Verifier 3.5, 3.0;
- ScanSoft SpeechSecure 3.0.

Conclusion

Envox has over 1 million ports deployed worldwide and for good reason — this is one of the premier communication development platforms on the market today. It is very easy to develop applications such as IVR, unified messaging, VoIP, directory assistance, speech applications, call center applications, credit card fraud alert applications, and more. With its ease of use resulting in rapid application development time and support for more telephony and networking standards than you can shake a stick at, Envox 6.1 earns high marks from TMC Labs. IT

The State Of Softswitch:

Adding Value To Carrier Networks

Since the early days of VoIP, softswitches have been the principal network component and the key enabler of VoIP service delivery. A regular buyer's guide will give you fifty something companies that have softswitches in their product portfolio, but when you take a closer look at all those products you will find that a softswitch is indeed a very elusive term, and each vendor is implementing their own understanding of the softswitch concept.

Potential buyers — as they look for a turnkey switching solution and make their choice between available vendors — should realize that the softswitch is no straightforward replacement of legacy "black box" TDM (define - news - alert) switching gear. It is more about a system comprised of single elements, each of which adds to the overall functionality of the whole. It can be viewed, and is often arranged, as a number of logical components built around a single core. Such components can be incorporated in one hardware platform or run independently on different servers across physically distributed locations. The principal function of signaling and media control is thus supplemented by add-on yet indispensable tools, which constitute the major value of a given softswitch. These are the tools to check for if you compare softswitch products of different vendors, since they are the ones that actually make the difference. Routing tools provide smart routing of calls based on predefined criteria; border control units ensure interoperability of incompatible networks; signaling and media proxy provides for topology hiding and IP address concealment; billing and accounting modules are responsible for CDR generation and invoicing; and last but not least — QoS management tools serve to provide constant quality of calls depending on the user needs and preferences. In terms of functionality, softswitches stand really close to session controllers. The basic difference between them actually lies in the intrinsic inability of session controllers to provide VoIP to PSTN (define - news - alert) interfacing, but this gap is easy to bridge by means of additional modules with PSTN support.

Gasping For A Second Wind: New Value From The Softswitch

Softswitches have been and remain the key enabler of VoIP wholesale switching, the driving force of the IPT industry of the past few years. However, as the industry gets invaded by an ever increasing number of players, the wholesale VoIP (define - news - alert) market becomes way too competitive to generate significant revenues. To keep above the waterline in the sea of VoIP, carriers keep looking for ways to drive down costs and create new revenue streams. And similarly, the role of softswitch evolves to provide the technology that meets new carrier demands. This is evolution in the proper sense of the term: an old solution acquires new functions to keep the lead of the today's industry and create a springboard to the converged networks of tomorrow.

Let us briefly outline the functions that a softswitch should perform as a revenue-generating tool on a carrier-grade network.

Optimize Network Management

To optimize wholesale traffic exchange, carriers need to keep their costs low and margins high. A mature softswitch delivers intelligent routing capabilities — by switching between routes based on their cost to quality ratios, load, QoS, and other parameters

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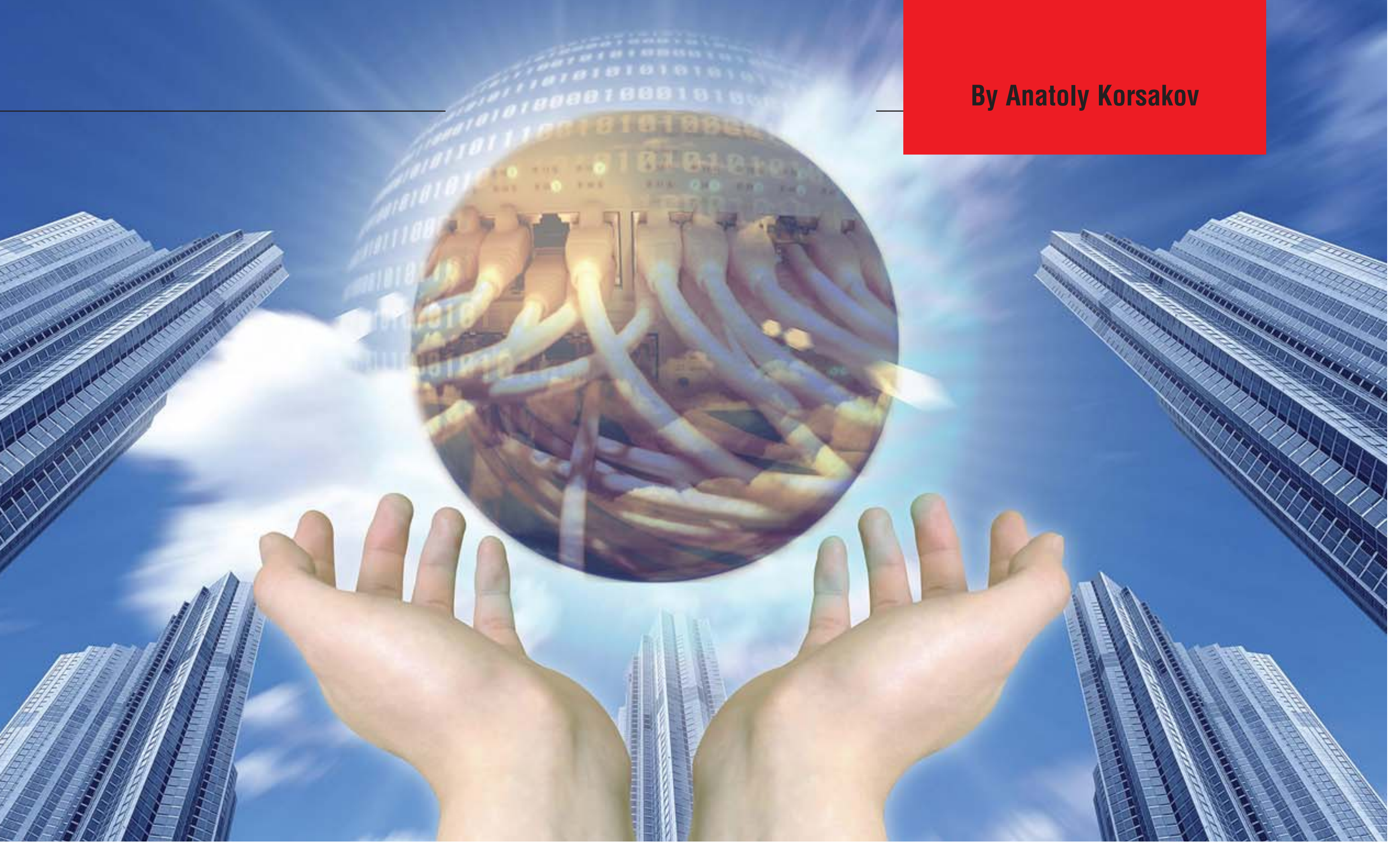

By Anatoly Korsakov

in real time. Such routing tools can increase profit margins by as much as 30 percent, according to carriers' testimonies.

Another critical feature that helps increase operational efficiency is the availability of advanced monitoring, analysis and reporting tools that enable carriers to keep the finger on the pulse of their network. These tools allow telcos to receive accurate real-time data on route performance and profitability, quickly react to the changing market conditions and take appropriate business decisions.

A powerful tool that will keep carriers' costs on the low end is on-line profitability control, backed by automatic redirect to alternative routes. If a carrier works with several terminating partners to a particular destination (NY, for instance), it is highly advisable to have an automatic engine in place that will monitor profitability dynamics, distribute the calls to the routes with maximum profitability, and present real-time views to the network analysts in the form of intuitive graphs. In other words, profitability control tools ensure that the carrier is not losing money at any given moment in time.

Furthermore, integration of softswitches with billing solutions enhances carriers' ability to collect data and increase marketing flexibility by means of embedded discounting and real-time invoicing systems — a special value for carriers working with enterprise users.

Support A Variety Of Business Models

Along with overall cost reduction and network optimization, industry leading softswitches provide another way for carriers to survive in the competitive market environment. A truly next generation softswitch enables carriers to diversify their wholesaling models and thereby drive new revenue-generating opportunities. A popular business model today is leasing softswitch partitions to Tier 2 and Tier 3 operators that do not possess their own infrastructure. These smaller carriers use the leased softswitch capacity to resell minutes on a wholesale basis or offer VoIP services to enterprises and residential customers.

Another business case for a softswitch is offering VoIP exchange services, where customers can buy and sell minutes online from multiple operators, based on optimal price/quality ratios. Advanced routing tools play a vital role in this business model — the softswitch sends calls traded on the exchange to available routes based on specified cost or quality criteria. The carrier profit is generated from the fees paid by traders for every minute of traffic coming through the exchange, the fees ranging from USD 0.0015 to USD 0.005 per minute depending on destination rates.

Softswitch Takes On New Roles

The past few months have demonstrated that the wholesale VoIP market has reached its climax and is likely to see a gradual yet constant reduction of revenues in the coming years. It is evident that the true potential and most profitable application of converged networks is in the many features and value added services delivered over IP. This is why many wholesale carriers start to add diverse value added services to their offerings.

At the same time, the boundaries between fixed and mobile networks tend to dissolve: researchers expect full-fledged fixed-mobile conversion somewhere in 2010–2012, meanwhile customers already benefit from the first steps in this direction. VoIP architecture will play a key role in the establishment of converged networks by supplying the missing link between the fixed and mobile domains.

Actually, softswitches are already an important part of today's FMC scenarios. For example, many mobile service providers offer cheap ILD calling to their subscribers, as they partner with VoIP carriers for long-distance termination of mobile traffic. Another popular business model for a VoIP telco is to operate as a Mobile Virtual Network Operator (MVNO) and resell services of a mobile carrier under its own brand, purchasing airtime at wholesale rates. Convergence of the wireless and wireline domains opens new and exciting ways to serve the end user, and the use of softswitches in this framework provides firm ground for development of new IP based services.

Take, for instance, this year's hot spot — IP Multimedia Subsystem (IMS). It is actually built around the IP technology with SIP in the core network. What's more, it heavily relies on the softswitch technology, whose parts or functions are dispersed within the IMS architecture levels. Here are the main requirements for the softswitch to fit in the new IMS world: its components must be distributed, it must be agnostic of the underlying networks and protocols, and support standards-based interfaces with other network elements. Some of today's softswitches that comply with the above requirements are already being utilized by IMS vendors.

Despite the obvious recognition of IMS benefits, there is a long way to go before it becomes commercially deployable — it is safe to estimate commercial implementation between 2007 and 2009. Carriers will continue to use softswitches to maximize the efficiency of their networks today — and will definitely rely on them as the FMC surge finally hits the market tomorrow. IT

Actually, softswitches are already an important part of today's FMC scenarios.

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Open Shelf Architectures Provide Foundation For Next-Generation Softswitches

As telecom OEMs (TEMs) retool and replace their traditional Class 4 and Class 5 switches with packet-based platforms, they must grapple with the age-old question of what to build in house, and what to outsource. What many TEMs are finding is that they can no longer remain competitive from a cost or time-to-market perspective developing custom systems from scratch in house. And even if they could, they lack the manpower. It takes a large staff to design, develop, manufacture, maintain, test, and support proprietary architectures in house, and TEMs are reducing their workforces, not expanding them.

To facilitate outsourcing, many softswitch OEMs are adopting open architectures with standard interfaces such as PICMG 2.16 and AdvancedTCA. These platforms reduce design, test, and manufacturing costs by enabling TEMs to utilize standard, pretested, off-the-shelf components, which enjoy larger economies of scale. They reduce stocking costs by enabling TEMs to standardize on a smaller number of blades and modules that can be reused across multiple product lines. They enhance functionality by giving TEMs easy access to best-of-breed third-party products. And they enhance product differentiation by enabling TEMs to focus on value-added application software and services.

Open Platforms Evolve

PICMG 2.16 and ATCA are the first open platforms that can provide the telecom-specific features that TEMs need to build scaleable, field upgradeable, high-availability infrastructure systems. They're also the first open platforms to provide the flexibility and headroom that TEMs need to keep ahead of the fast-changing softswitch landscape.

Both PICMG 2.16 and ATCA provide the bandwidth needed to support large number of blades working together at the shelf level to perform compute- and data-intensive control and media processing functions. They also provide the packet- and TDM-based transport mechanisms that are needed to implement softswitch functions such as signaling and media gateways, which facilitate interoperability with the PSTN.

PICMG 2.16 supports packet-based Ethernet transfers over the backplane at speeds of up to 1 Mbit/sec per channel. It also provides a dedicated H.110 telephony bus for acquiring and processing TDM data. ATCA provides a redundant (up to full mesh) switched fabric that scales up to 10 Mbit/sec per channel and supports multiple packet based protocols (Ethernet, PCI Express, RapidIO, etc). Both PICMG 2.16 and ATCA blades are hot swappable, enabling the blades to be upgraded and replaced in the field without disrupting overall service, a key requirement for maximizing availability.

To enhance flexibility and make outsourcing even easier, both PICMG 2.16 and ATCA are available with mezzanine expansion options. By combining a generic PICMG 2.16 or ATCA blade with application specific modules for media processing, signaling, packet processing, network management and control, TEMs can outsource a broad range of functions at the blade or module level. This modular approach not only speeds time to market and provides

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By Stuart Jamieson

access to low-cost mass market products, it reduces the number of unique blades that TEMs and service providers have to design, manufacture, and stock. For field replaceable mezzanine solutions like AdvancedMC, it enables service providers to service, provision, and spare their systems with a finer degree of granularity, thus reducing cost.

Most PICMG 2.16 systems use PMC (PCI Mezzanine Card) for mezzanine expansion. PMC is also available in telecom-friendly configurations such as ProcessorPMC (PrPMC) and PTMC (PCI Telephony Mezzanine Card). These mezzanine formats can support TDM data transfers (in addition to PCI) and enable processors residing on the module to act as the master processor for the PICMG 2.16 blade.

ATCA systems can also utilize PMC mezzanine expansion. However, PICMG has developed a module specification specifically for ATCA known as AdvancedMC, which is optimized for ATCA's switched fabric, physical size, hot swap, and system management facilities.

Both PICMG 2.16 and ATCA platforms support Integrated Peripheral Management Interfaces (IPMI), a system management interface that enables network management systems to directly provision, control, and program all softswitch subsystems. Through IPMI, shelf management and network management systems can monitor and control individual blades, negotiate power allocation, detect faults, and coordinate failover. As befitting the next generation of mezzanine modules, AdvancedMC modules also support IPMI, giving shelf managers even greater visibility into and control over blade and module operation.

Softswitch Architecture Evolves

Today's most advanced softswitches offer packet-based wireline and wireless call control, switching, and application services. Distributed throughout the packet network, softswitches work together to establish connections and facilitate the transport of user data between end points. They also control, and in some cases contain, the signaling and media gateways that bridge packet networks with the PSTN and maintain compatibility with traditional PSTN terminals, while supporting the latest IP-based (SIP and H.323) end points.

The term softswitch is often used interchangeably with MGC (Media Gateway Controller). Increasingly, however, the term softswitch refers to a more highly integrated platform that combines the MGC with the media gateway, signaling gateway, shelf management, network management system, and interfaces for billing and provisioning. Collectively, these systems replace and interoperate with the Class 5 switches that are used throughout the PSTN to provide trunking, SS7 networking, translations, routing and network services.

To see how softswitch components interact with each other and the PSTN, it is helpful to trace the call control process for two cases, one in which both endpoints are IP based, and the other in which one of the endpoints is PSTN and the other is IP. When both endpoints are IP based, the source end point (such as a VoIP phone) initiates the call by sending a request to the MGC, which locates the destination endpoint and responds to the source endpoint (either number not found or trying). The MGC then sends a request to the destination endpoint, which, if available, starts ringing. The MGC also sends a message to the source endpoint to start ringing. Once the call is answered at the destination endpoint, and the MGC receives a pick-up message, it sets up a bi-directional voice channel through the media gateway, which performs (if necessary) the VoIP voice data processing (e.g., echo cancellation, codec conversion).

The call process is more complex when one or both of the end points are PSTN (phone, modem, fax, and so on). Consider the case, for example, where the originating end point is a PSTN phone, and destination is a VoIP phone. After the PSTN (define - news - alert) phone picks up and starts dialing, the PSTN switch sends an ISUP (define - news - alert) message to the MGC (via the SIGTRAN signaling gateway), which sends a request to the VoIP phone. If the VoIP (define - news - alert) phone is available, the MGC tells it to starts ringing. The MGC also sends a ringing sound (via the media gateway) to the PSTN switch, which relays it to the PSTN phone. When the VoIP phone picks up, the MGC informs the PSTN switch, which works with the MGC to set up a bidirectional voice channel through the media gateway.

Conclusion

Open platforms like PICMG 2.16 and AdvancedTCA provide an excellent platform for building scaleable, high-availability softswitches, from dedicated MGCs, to complete softswitch systems with integrated gateways, network management, application and accounting functionality. These platforms reduce time to market and cost by enabling TEMs to outsource substantial portions of their softswitch design. They also reduce capital and operational expenditures by providing a modular, flexible, field upgradeable framework that can be serviced and provisioned with a fine degree of granularity, and readily adapted to support next-generation softswitch services and architectures. IT

To facilitate outsourcing, many softswitch OEMs are adopting open architectures with standard interfaces such as PICMG 2.16 and AdvancedTCA.

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Balancing QoS & Security In A VoIP Environment

Do a search on the Internet for "VoIP Security" and your search will come back with thousands, if not millions of hits. Voice over Internet Protocol (VoIP) security is fast becoming a number one concern for those deploying this relatively new technology. Hackers are taking notice of this expanding, previously untargeted area as millions of users begin sending conversations over IP networks.

VoIP security presents risks that organizations must address as they move to IP Telephony environments. VoIP can bring organizations tremendous benefits — cost savings and increased productivity to name just two; but VoIP (define -news - alert) will also undoubtedly bring security risks. While you will need to add features to your environment to secure VoIP, you'll want to add these features without hindering Quality of Service (QoS). Voice has real-time characteristics, which have very strict requirements for network performance. Delay (latency), lost packets, and jitter are the key network impairments that can cause degradations in VoIP call quality. As you address VoIP security, be aware of the potential performance impacts.

VoIP Performance Factors

A telephone conversation depends on interaction between the caller and the called party. The higher the level of interaction between the two parties, the less they can tolerate delays in the conversation. Most industry experts agree that 150 milliseconds is a good target for the end-to-end delay in a VoIP call. A total end-to-end delay of less than 150 ms is required for toll-quality calls. Delays greater than 400 ms can effectively render the service unusable.

Firewalls and other security applications that work well in data network environments are often not as effective in VoIP environments because they can exacerbate delay. In a data environment, a delay in transmitting information via e-mail likely won't be noticed by the sender or receiver of that data. However, if packets in a phone conversation are delayed even a fraction of a second, it can significantly reduce the quality of a call.

IP phones send VoIP packets at consistent intervals, typically every 20 or 30 milliseconds. When the packets arrive at varying time intervals, the resulting problem is called jitter. Jitter buffers smooth out the variations but add delay, and if the variation is too great, the jitter buffer can drop packets. As new security features are implemented, be careful to check their impact on jitter.

Security features that cause potential bottlenecks can lead to congestion and lost packets. When packets are lost, portions of the conversation, such as a word or a syllable, can be lost. Sophisticated packet-concealment algorithms can minimize some of the impact of randomly lost packets. However, when the packet loss occurs in bursts, or when multiple consecutive packets are lost at a single time, the loss can have a dramatic impact on the quality of the VoIP conversation.

Security mechanisms that are imple-

By Jeff Hicks

mented on VoIP servers can also potentially cause performance issues. For example, everyone knows the value of anti-virus software to scan your computer and protect from viruses. It is important to run anti-virus programs on your key VoIP servers. However, real-time scans of frequently updated trace or log files can cause CPU usage and disk access times to increase. You may need to exclude certain directories or files from the virus scan to prevent a VoIP performance hit.

Other security agents on VoIP servers can intercept the kinds of harmful functions that can be accessed at the API level. These security agents can prevent some attacks by controlling what the malicious applications can do. But if you implement such agents, you need to consider the performance overhead for detecting intrusions at this level. Testing is essential. If you turn on all of the protection features, what is the impact on CPU and memory usage? These are some of the tradeoffs you may have to consider.

Encrypting VoIP traffic is a good way to stop hackers from eavesdropping on your conversations. One caveat, though, is the possible impact on network performance as you encrypt traffic. Encrypted traffic requires more device processing and can increase delay, so you may have to deal with declining quality as you secure your calls. Monitoring your call quality as you implement new security features is important so that you don't get caught off guard by user complaints.

A layered security approach is recommended to protect a VoIP deployment. Different network layers have different security features. At the networking Layer 2 (switches), a good recommendation is to protect against ARP poisoning and other spoofing attacks. At Layer 3 (routers), firewalls are a must to harden the perimeter and protect against external attacks. Firewalls often offer different levels of protection, such as deep packet inspection, a method that can decode the VoIP protocols. But the performance implications of layering security features must be considered. Does the feature increase delay, increase the likelihood of congestion, or add variability to packet delivery?

Security and performance management are closely inter-related, creating a delicate balance between security and VoIP quality. You may have to consider tradeoffs between a security feature and the quality of your phone calls. Having good tools to help you assess and monitor the security risk, while monitoring your performance on a 24x7 basis, is critical.

As you closely monitor VoIP quality and security, keep evaluating VoIP from a service perspective. Each element has its own security features, but what you really want to know is how well the overall telephone service you're offering is performing, and how well it's protected.

Managing Security Of An IP Telephony Service

Managing security needs to follow the same approach as managing the performance and availability of the IP telephony service itself. Just as you can't manage VoIP availability by monitoring a single element of the service, you can't ensure security by securing just a single element of the system.

Performance, availability, and security are closely interrelated. Providing trusted services relies upon continuously managing and balancing performance and security with the cost required to provide each. As we've seen in the previous section, these three factors are often competing: optimizing one factor generally comes at the expense of the other two. IT providers must work with business customers, legal advisers, compliance teams, auditors, and others to determine the proper balance. Creating effective, efficient, and repeatable IT management processes is critical to achieving the proper, agreed-upon balance.

Some organizations are beginning to utilize best-practice service management frameworks such as ITIL, the IT Infrastructure Library, to help define and implement service and security management processes. In ITIL, these processes are divided between service support — the day-to-day operation of IT services — and service delivery — the long-term optimization of IT services. Unfortunately, while ITIL is starting to gain popularity, most IT providers still employ a very fragmented approach, even to individual service-management processes like incident management. The result is that no one quickly understands incidents, their causes, the business impact, or the appropriate response. Instead, the incident management personnel scramble to gather all of the necessary information, often involving distinct groups

Enterprise Security Issues

By Faizel Lakhani

More enterprises are considering a migration from legacy circuit-switched PBXs to IP PBXs (VoIP) in order to capture the benefits of IP telephony — lower telecom costs and value added convergence features, such as voice and data integrated call center applications. However, this migration leads to concerns over performance and security, since unlike circuit-switched infrastructures that physically separate voice services from data networks, IP telephony merges traffic on the same IP network.

The current enterprise "best practice" to address this issue is to place IP telephony resources (call managers, media gateways, and Ethernet handset phones) on separate VLANs (virtual LANs) from those used exclusively by desktop/laptop PCs and data application servers. ACLs (Access Control Lists) are then used to tightly control which devices (and TCP/UDP ports) can access resources on the other VLANs (e.g., communications between voice mail servers on the voice VLAN (define -news - alert), and e-mail servers on the data VLAN).

While this solution does, to a certain extent, provide logical separation of voice and data networks, there are considerable security and scalability limitations inherent with this solution that will inevitably inhibit the widespread deployment of IP telephony. First, the management of MAC addresses and ACLs may be feasible for a few select users/locations within the organization; however, tracking the MAC address and physical location of each and every phone in a 50,000 user network poses a significant management challenge. The likelihood that an enterprise has and maintains a complete, up-to-date list of all the MAC addresses of user PCs and laptops throughout their network is very low. It's easy to see how this problem literally doubles with an IP telephony rollout.

With the expanded use of Windows-based softphones [enabling voice communications directly from the end system itself], such as in call centers using voice-integrated customer support applications, an alternative to separate VLANs for voice traffic is required. This is due to the fact that existing network infrastructures are unable to distinguish the specific application type that is being sent from the PC since voice and data traffic are now merged into a single pipe. As a result, voice related traffic is treated with the same level of priority and security as all other traffic, opening the enterprise up to increased risks of performance degradation and security threats for their voice traffic.

A emerging "best practice" that is currently being deployed by enterprises is to implement a new layer of control in the LAN. This solution consists of a high-speed, in-line device called a LAN Controller that resides between the access switches that provide connectivity to PCs, softphones, and IP handsets and the network core where the IP telephony infrastructure resides [Figure 1]. These LAN Controllers work with an existing network infrastructure to provide a very high degree of security and performance assurance required by IP telephony applications.

LAN Controllers guard the IP-PBX server infrastructure from denial-of-service [DOS] attacks. Positioned close to users in the LAN, these systems can effectively track call session and protocol flows while also identifying and blocking malicious packets before DOS attempts can infiltrate the network.

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that do not have operational responsibilities. This consumes valuable time and effort needed for the response and often leads to other issues.

While a close focus on all the specific security incidents that could occur in a VoIP environment is important, managing services from an integrated performance and security perspective is even more important for ensuring the overall and ongoing health and availability of those services. And as we've stated in the previous section, VoIP is fundamentally a service, a set of linked and interdependent components. It's a highly critical service that's practically useless if it's not performing well.

Summary

VoIP security presents challenges, but these challenges are manageable with careful planning, security awareness. Just like any other area of security, you can make things totally secure, but at what cost? For VoIP, the tradeoff could be an incredibly secure phone system that no one uses (they uses cell phones instead) because the call quality is always poor.

Because VoIP is more sensitive than most applications to small fluctuations in the performance of its key components, it's a good idea to view VoIP as a service, rather than a system of small components to micromanage. Instead of separating your VoIP deployment into different elements and trying to secure each one separately, you need to manage the performance, availability, and security of your VoIP deployment as an integrated service, wherein performance and availability — linked as they are to the health of the entire data network — must be balanced with security concerns. **IT**

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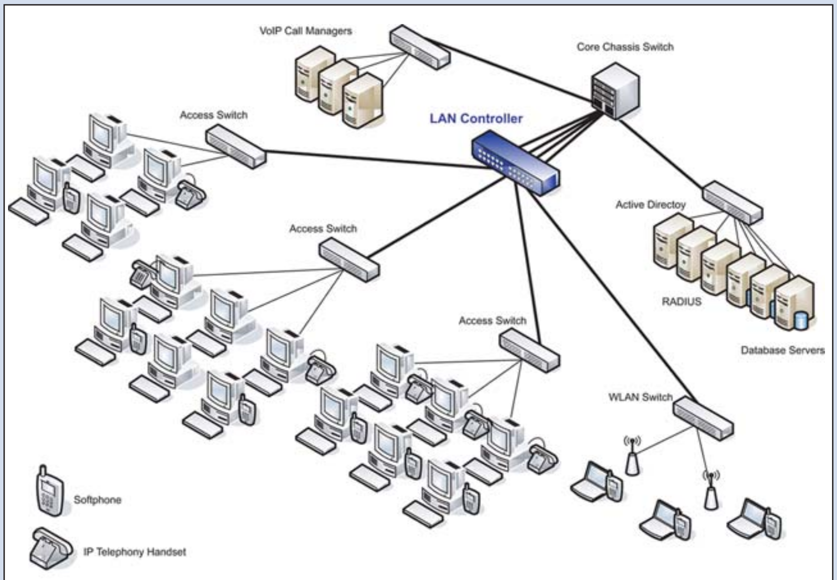

LAN Controller provides a layer of security and performance control for IP Telephony

Stopping threats close to the source is especially vital for IP telephony communications emanating from a branch office to headquarters over a frame relay WAN connection. Once the malicious traffic traverses the WAN, identifying the exact user or even location causing the security breach becomes considerably more difficult. A LAN Controller deployed at corporate branch offices can effectively identify the exact source of the attack and prevent any disruption of service for telephony traffic within the branch office or calls destined for the corporate headquarters.

LAN Controllers also ensure that only authorized users are able to access the IP telephony infrastructure by effectively adding a layer of port-level Network Access Control [NAC]. Before a voice-enabled system is allowed to communicate on the network, the LAN Controller will query the existing enterprise user directory [LDAP, Active Directory, RADIUS] to authenticate the user and authorize which applications may be used over the network. This will stop various types of illegal activities, such as rogue handset or softphone users from piggybacking calls over the corporate telephony infrastructure.

Finally, as LAN Controllers are application-aware, they can provide a high degree of quality of service [QOS] for IP telephony while providing an effective alternative to complex MAC-based VLAN management or an even more expensive, dedicated IP telephony network. This same solution works even for softphones. By differentiating between IP data traffic and IP telephony traffic [e.g. SIP or H.323 data and control traffic], LAN Controllers provide both preferential voice traffic forwarding that virtually eliminates jitter and latency issues and a heightened level of security protection, regardless of whether voice and data traffic are traveling on the same physical network port. This sets the stage for IP telephony to finally become ubiquitous. **IT**

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Turning IP Triple Play Into A Home Run

Convergence is finally happening. For years this idea of convergence was only a vision — driven by fanciful notions rising out of technical possibility rather than true business needs. Today, that has changed. Business drivers and consumer behavior have combined to effect a "perfect storm" accelerating the adoption of bundled voice, video, and high-speed data solutions across the Americas market. In this article, I will discuss how this new world of triple play convergence has become a reality, how successful IP triple play platforms can be created, and how to make the right technical and business decisions to ensure success in your own triple play deployments.

Five years ago, some of the largest triple play service providers in the world today, like FastWeb, barely existed. Incumbent telcos were only slowly investing in next-generation technologies such as Voice over IP (VoIP) to migrate away from expensive and inflexible TDM-based equipment. After the crash of the technology bubble of 2000, it took some time for incumbent service providers to begin to see the business benefits of these new technologies. And it took a while for the average consumer to understand the true value of these new services.

But that is what is happening right now. Telcos are actively investigating or rolling out IPTV (also called "telco TV") and IP Video on Demand (IP VoD) services. The more innovative incumbent MSOs, like Comcast, are heavily marketing video calling services such as video voice mail. These video providers are also enjoying success deploying traditional primary line voice services. Wireless providers are now looking at video services. And even municipalities are getting in on the act, deploying greenfield fiber or broadband wireless networks with triple play service bundles. Today's success stories show us that a converged network has both business and psychological advantages. On the business side, IP-based networks are less costly to finance and operate, and new service bundles generate increased revenues. But even more important is the view that these new service bundles foster long-term relationships with customers, reducing customer churn (and therefore reducing the costs of attracting, keeping, and bringing back customers) and increasing adoption rates for new services.

IP triple play, which is the combination of voice, video, and high-speed data services from one provider, is the first example of myriad potential services that demonstrate the value of an integrated bundle. But more than just getting these three services from one provider on one bill, it is important to understand the key value points that make this integrated package so attractive to consumers. Below I list the critical ingredients of the solution and then discuss them one at a time:

1. Integrated subscriber experience.
2. Converged and scalable network equipment.
3. Simple, yet sophisticated management system.
4. Service creation capabilities.
5. Multiple access technologies.
6. Flexible business models.
7. Aggressive marketing and consumer education.

1) Integrated subscriber experience. The biggest change in the mindset of service providers today is how a service needs to adapt to consumer behavior, rather than consumer behavior adapting to new technologies. The best example of this is the new "TV Portal" or "Home Kiosk" products being offered by triple play service providers. Combining the best features of a menu-driven digital TV display with a cell phone-like interface,

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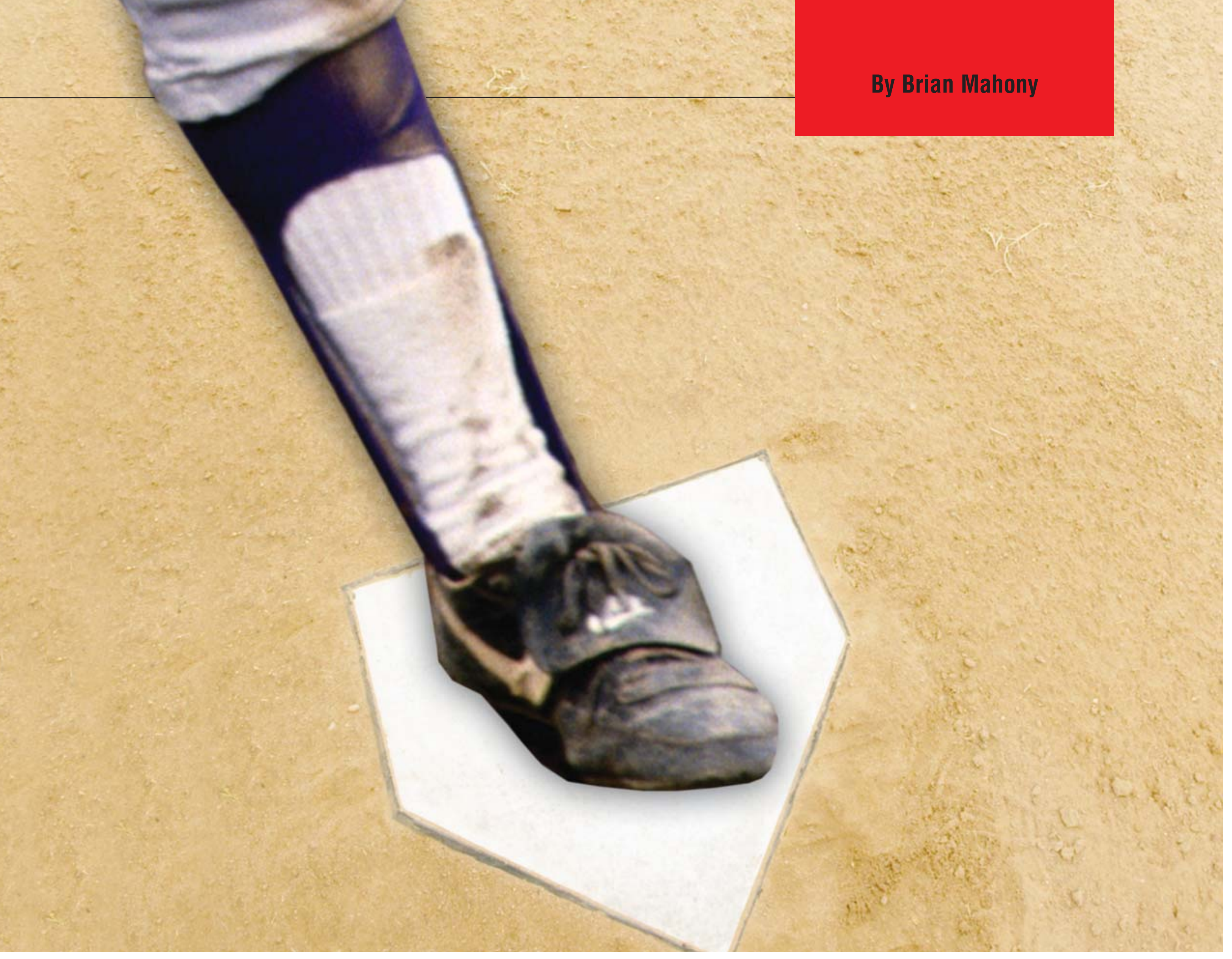

By Brian Mahony

these TV portals provide an integrated subscriber experience that consumers understand within seconds. Instead of being forced to get up from your favorite TV show every time the phone rings, the TV portal displays the caller ID on the screen, and gives you a chance to send unwelcome callers directly to voicemail. Subscribers can also check called or caller lists and even dial from the TV using a normal remote control (the calling party then picks up a nearby connected phone). All of the communications and entertainment equipment (TV, phone, remote control) remains the same except for a new IP set-top box connected to a broadband access device. To extend the capabilities of the TV "family interface," a similar subscriber portal is offered on the PC "individual interface," which takes advantage of the resident mouse and keyboard to offer a greater level of control of the service (for example, setting up call forwarding rules to send calls to your vacation home; or programming the phones not to ring past a certain hour).

2) **Converged and scalable network equipment**. The experience of the network operator should also be integrated. Today's new multi-protocol SIP/MGCP/H.323 (define - news - alert) softswitches are much easier to manage than legacy TDM switches. The provisioning interfaces and dial plans have much greater flexibility too. Combine this with IP video equipment and media servers, and you have a next-generation "head-end/central office" that serves all of your voice, video, and data needs. A legacy TDM switch upgraded with an IP gateway card will not suffice. And neither will a call control agent imbedded into a media gateway. To be able to deploy future IMS and TISPAN compliant services, the transport, call/media control, and application layers must all be disaggregated and have the ability to be distributed without regard to any geographic barriers. This is the only way to deliver a scalable multi-service platform that can grow to millions of subscribers, and across many different access technologies.

3) **Simple, yet sophisticated management system**. Equally important to converged network equipment is an Operations Support System (OSS) that can effectively manage all the different services in an integrated way. This means single data entry for voice, video, and data subscriber services, plus rating and billing for these services. Also, the service provider should have no limits to cross-bundling programs (for example,

providing one free VoD movie for every 100 minutes of VoIP talk time). San Jose State University, rolling out triple play campus services this Fall as part of their "Campus Village" project, is a good example of this. SJSU defined three service levels (basic, silver, gold) that includes increasing levels of cross-bundling (e.g., "gold" service includes high-speed Internet, unlimited local and long-distance calling, a wide selection of IPTV channels, a package of free VoD movies, and unlimited access to the gaming center). Studies show that the more you bundle services, the less price-sensitive your customers become. With this in mind, the OSS can become a powerful weapon that uses different billing schemas to further cement customer loyalty. In our experience, an easy-to-use but sophisticated management system is as important as the equipment that enables the service.

4) **Service creation capabilities**. Related to the OSS system is a service creation environment that allows you to rapidly define and roll out new services. This requires a flexible object-oriented environment that allows you to create decision trees and workflows to define a new service simply by dragging and dropping and linking known service objects. Some of the better service creation tools also have service simulation programs that can help you predict profitability for new services before you actually deploy them. And since this service creation environment is tied to the subscriber portal, new service promotions can be delivered as TV menu "pop-ups," which subscribers can select and provision through the push of a button on the remote control. Studies have shown that subscribers are ten times more likely to sign up for a new service if they are not forced to write down a number, call a call center, or search on the Web. Finally, an IP service creation platform should allow you to easily roll-out a range of subscriber services that go beyond the basic triple play (e.g., gaming, video calling, distance learning, e-medicine, etc.). We have even seen some municipalities in the United States already begin to use their city-wide IP network to intelligently control traffic lights, security cameras, and utility meters remotely.

5) **Multiple access technologies**. One of the key lessons from successful deployments is that the choice of access technology does matter, but not to get too committed to just one form of broadband access. An example can be drawn from our European neighbor,

Statement of Ownership, Management and Circulation

(Required by 39 U.S.C. 3685)
Publication Title: **INTERNET TELEPHONY**®
Publication Number: 1098-0008
Filing date: 9/16/05
Frequency: monthly
Number of issues published annually: 12
Annual subscription price: requester (none to qualified US, $29 nonqualified US, $39 Canada, $60 Foreign)
Mailing address of Know Office of Publication and Headquarters or General Business Office:
Technology Marketing Corp., 1 Technology Plaza, Norwalk, CT 06854
Publisher: Richard Tehrani, Technology Marketing Corp., 1 Technology Plaza, Norwalk, CT 06854
Editor: Greg Galitzine, Technology Marketing Corp., 1 Technology Plaza, Norwalk, CT 06854
Owner: Nadji Tehrani, Technology Marketing Corp., 1 Technology Plaza, Norwalk, CT 06854
Issue Date for Circulation Data Below: September 2005

Extent and Nature of Circulation	Average # of Copies each issue during Preceding 12 months	No. copies of single issue published nearest to filing date
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Today's success stories show us that a converged network has both business and psychological advantages.

FastWeb, where the original business plan called for FTTx as a high-bandwidth medium for triple play. Today FastWeb is about half fiber and half DSL. There are some trade-offs of course. Fiber provides almost unlimited bandwidth and therefore does not force you to choose among competing high-bandwidth service bundles. Fiber also has the advantage of allowing the service provider to deploy lower cost MPEG-2 video compression (IPTV for about 4 MB/s; HDTV for about 20 MB/s). On the other hand, while DSL requires more costly MPEG-4/H.264 video for multiple IPTV channels (IPTV at about 2 MB/s and HDTV for about 4 MB/s), it can also take advantage of existing telco copper and save millions on fiber last mile construction projects. Similar lessons can be applied to wireless last mile technologies, as WiFi/Mesh/WiMax technologies could be considered replacements for or enhancements to today's fiber/DSL/cable architectures.

6) **Flexible business models**. Another lesson learned is that triple play service providers should be open to different business models. For example, you may or may not decide you want to own all of the next-generation equipment. In the United States, some innovative application service providers (ASPs) are providing triple play services across a fiber network built and run by the city of Provo, Utah. With triple play, there exist a multitude of different business models when you consider who owns the network, the application servers, the access technologies, the customer premises equipment, the marketing relationship with the subscriber, customer care, etc. The supply-chain possibilities are endless. In addition, triple play service providers targeting the residential market may decide to offer business services such as IPCentrex to businesses passed, and vice versa.

7) **Aggressive marketing and consumer education**. A final recommendation for deploying successful triple play solutions is to not overlook the marketing and education required to get subscribers' attention and compel them to use the new service. Marketing ensures the greatest possible uptake of new services and allows you to lock in new customers before your competitors do. Further, together with targeted education programs, it gives new subscribers the confidence and comfort to work through the inevitable kinks that will arise in any new triple play deployment. A good example is the program iProvo put together, where they combined advertising, DVDs promoting the new services, community question and answer meetings, and door-to-door "get the word out" campaigns for maximum visibility and customer acceptance.

IP triple play services are viable and profitable today. However, in this new open competitive environment were traditional rivalries are intensifying and new entrants are threatening incumbents, making the right technical and business decisions will be the difference between success or just survival. **IT**

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How Communications Service Providers Serve Up IPTV

Since Burger King launched its famous "Have it Your Way" campaign in the 1970s, companies of all shapes and sizes have claimed the same idea — target the right customers with the right products at the right time. Today, wireline communications service providers (CSPs) facing stiff competition from cable operators, are focusing on the triple-play. However, simply delivering these services is not enough. CSPs must seamlessly integrate IPTV and offer consumers what they want — a fully integrated bundle of services tailored to meet their specific needs. To successfully deliver the triple-play, CSPs must adopt an integrated customer management approach with the ultimate goal of creating an intentional customer experience that forges stronger, more profitable customer relationships.

The Race For The Triple Play

In response to growing competitive pressures, many CSPs have deployed bundling strategies. The hope is that by bundling services together, consumers will perceive greater value resulting in decreased churn and increased customer lifetime value. But despite their efforts, CSPs are finding themselves behind in the race for the triple-play. The large cable operators in North America are already delivering the triple-play with bundled cable telephony, broadband over cable, and cable video services. In order to counter this 'triple-threat', CSPs must differentiate their offerings. Traditionally, the telecom bundle included fixed voice, wireless, and broadband services. Recognizing the gap, many CSPs have added satellite delivered video services to their bundles. In most cases, however, satellite video only offers a service that is equal to cable and fails to differentiate the telecom bundle. As a result, many CSPs are now looking to IPTV to evolve their bundle and deliver a more compelling offering that will take them beyond the triple-play into a quadruple or multi-play environment. The CSPs who will win this race are those that are prepared to take advantage of the opportunities created by their IPTV and IP convergence strategies. These include opportunities such as fixed-mobile convergence where a mobile call can be continued from a subscriber's mobile phone to the fixed line in the home office, and feature interaction between networks including the ability to display an IPTV EPG on a mobile phone and set recording times for TV programs using the same mobile device.

Will Adding IPTV To The Telecom Bundle be Enough?

Despite the promise of more innovative and personalized video services, the nagging question is whether or not simply adding IPTV to the telecom bundle will be enough to gain competitive advantage and satisfy the growing expectations of consumers. To answer this question, we must look at bundling from the customer's perspective.

By Eran Wagner

Although bundled services are perceived to deliver greater value, the majority of customers have less than optimal bundling experiences. For example, a leading North American CSP offered a highly attractive service bundle that included fixed voice, wireless, satellite TV, and high-speed internet with a common monthly bill. An independent sampling of this service found that for one customer it took a total of three weeks to switch from his existing service to this bundle. During this three-week period, the customer had over 21 interactions with the CSP including four technician visits, more than six hours on the phone with CSRs and seven different welcome letters! The problem — the CSP operated along line-of-business silos in which the organization structures, business processes, and front and back office systems were all network centric and operated in isolation of one another. As a result, each line of business interacted separately with the customer to configure provision and support its service without insight or regard to the other services in the bundle or the customer's behaviors and preferences. This fragmented approach meant that the CSP had no insight into their customers and any customer experience was merely a by-product of the CSP's operating activity. From the customers' perspective, even though they were actually only dealing with one CSP, their perception was that they were dealing with at least four different companies. Needless to say, any perceived value of the bundle was negated by a customer experience that was inefficient, impersonal, and extremely frustrating!

It is not enough to simply be able to put all services on one bill or even to achieve rapid time-to-market for new IP services like IPTV within an existing bundle. For a bundling strategy to work and successfully deliver a multi-play, a CSP must be able to create the intentional customer experience. This means that the customer experience becomes the driver of all operations and business decisions. Creating an intentional customer experience requires a fundamentally different business approach — an integrated customer management approach. This is a way of doing business in which all corporate resources are agile and aligned to deliver an intentional customer experience. To achieve an integrated customer management

> **Subscribers place a premium on personalized content that is tailored to their specific needs and preferences.**

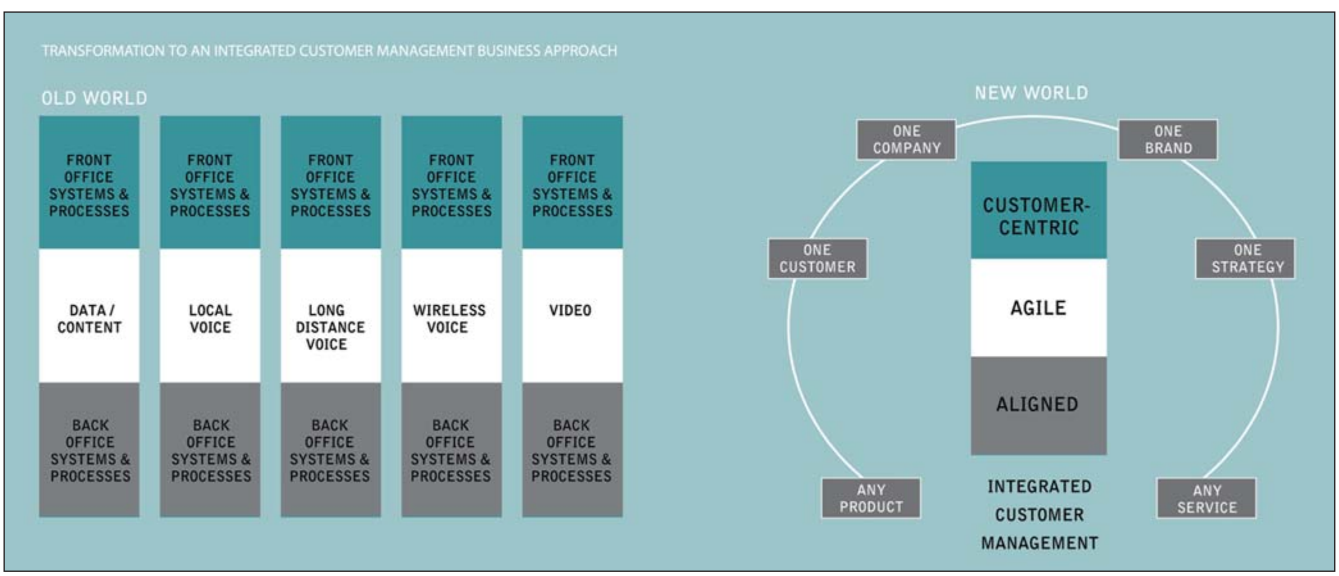
approach implies a move away from the 'stovepiped,' legacy systems, organizational structures, and business processes that characterize most CSPs. This old model of network centricity applied to the new needs of convergence means that the customer experience across services, functional departments and interaction channels is a by-product of, rather than the focus of, the business. In order to achieve integrated customer management, CSPs must place the customer at the center of their businesses.

Unlocking The Potential Of IP

By adopting an integrated customer management approach, a CSP's resources are aligned around the customer, providing visibility across all services supported by the IP platform and insight into how and when customers use their services as well as their personal preferences. This creates unlimited opportunities to leverage the customer experience to maximize value for both the CSP and the customer, particularly in the areas of personalization, self care, and content management.

Subscribers place a premium on personalized content that is tailored to their specific needs and preferences. Having a complete view of a customer across all services and devices allows for a much greater level of personalization and the ability to create a differentiated and intentional user experience. For example, providing a sports fan the ability to access real-time updates of his favorite football players while watching a match on TV, and viewing everything on the same screen. Or a customer watching a cooking program receives on-screen links to Web pages featuring related content and program scheduling notices. How a customer pays for services and goods purchased through their IPTV service can also be personalized. For example, providing a customer with an advice of charge for a transaction that reflects any bundling or promotional offers that apply specifically to that individual and the services he subscribes to. Likewise, a CSP can also provide multiple options for customers to pay for goods and services that meet their individual preferences; this includes the options to charge a transaction to their credit card or monthly bill using their TV remote control.

By adopting an integrated customer management approach, a CSP is also able to leverage the full benefits of self care as a strategic interaction channel. With an integrated customer management approach the CSP is able to provide a single, personalized user interface that can be replicated across multiple devices, interaction channels and applications. Such robust self care functionality is critical to the success of the multi-play strategy where the expectation of the customer will be to have anytime, anywhere access. For example, a customer who subscribes to an IPTV bundle will expect to be able to use the self-care functionality of his mobile phone to order a pay per view TV program and set the program to be recorded while he is at work so that he can view it at a later time that is more convenient for him. When using his mobile phone to order this service he wants to be able to access all of his account and IPTV service information and have complete visibility of the order process and receive confirmation that the order was successfully placed and billed correctly. This customer will expect his user interface, profile, and experience to be exactly the same regardless if he placed the order through his TV, PC, or mobile phone. Similarly, he expects that if he had a problem with the order or the billing of the program, when he contacts customer service the CSR would have complete visibility of his profile and



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provide consistent support across all touch points. Providing customers with the ability to self manage all aspects of their accounts, not only empowers the customer but also helps the CSP to significantly lower support and call center costs.

In the multi-play environment customers also want an unlimited choice of content. As a result, CSPs must take on the role of a retail content and entertainment super store where customers can browse, discover, and buy any content service they chose on demand. This requires CSPs to rapidly roll-out new content and services (much of it from third parties) at a dramatically lower cost than they can today. To do so, involves automating the complex tasks of delivering content services and managing complex business models with any number of partners. To proactively deliver high-value content and a differentiated customer experience, also requires understanding what content consumers are using, how, and when. Such insight can only be achieved through an integrated customer management approach.

To remain competitive and meet growing consumer expectations, CSPs must be able to support the multi-play. But as we have already seen in the market, it is not enough to just rapidly bundle IPTV and other digital content services together and present a common bill. To achieve market leadership, requires CSPs to undergo a significant business transformation from legacy voice platforms and network focused business models to an agile, customer-centric business model that facilitates a supermarket of digital, multi-media services and an intentional customer experience at all touch points — true integrated customer management. Only by taking this new approach to their business, can CSPs ensure their customers can have it their way! IT

To remain competitive and meet growing consumer expectations, CSPs must be able to support the multi-play.

Eran Wagner is vice president, IPTV at Amdocs. For more information, please visit the company online at http://www.amdocs.com.

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The Real Killer App In The Enterprise:

Converged Conferencing

For the past decade, carriers and OEMs have invested billions of dollars in building IP networks as a way to reduce costs — and they continue to do so. Cost, however, is no longer sufficient justification for deployment of IP communication networks. IP networks are attractive not only because they're less expensive to develop and maintain, but because they enable convergence of different media types, different communication methods, and different devices… all in the form of new valued-added applications. Consequently the greater value of Internet communications has yet to be realized. Simply put, by focusing primarily on reducing costs, carriers and OEMs are missing out on opportunities to exploit existing multi-billion dollar markets and grow revenue significantly.

Piecemeal Offerings Abound

While carriers continue to invest billions of dollars in IP networking because of the cost savings associated with VoIP, they have yet to tap the potential of converged applications. Inexplicably, these applications still take a backseat as the primary driver for investment.

But what's most surprising is that the multiple discrete services are being offered without leveraging the convergence opportunity of multi-service IP networks. Some carriers recently came to this realization when they renamed their VoIP (define - news - alert) programs "CoIP" because the value is in the communications, regardless of the type of media type being transported — voice, data, video, conferencing, and so on.

End users want the ability to tap into information and communication services whenever and wherever they want. The problem is that compelling communication services — such as presence, location, conferencing, and messaging — are often locked-up in "stovepipe" silos of functionally within a carrier's network. These services are, in turn, delivered to customers piecemeal, making them difficult to use and costly to support.

Consider the enterprise environment. Today most carriers and network equipment providers are deploying a piecemeal strategy for their different service offerings. This approach not only provides a poor and fragmented end user experience, but is expensive and yields poor margins for the retail provider. Carriers, for example, market separate VoIP, audio conferencing, video conferencing, and Web conferencing, yet typically each one of those applications have separate user interfaces, separate billing and back office support, even separate networks, thus costing both the carriers and their customers incremental capital and operating expenditures.

In most cases, newer services, such as Web conferencing and collaboration, continue to ride separate application specific "media tone" networks of their underlying ASP vendors. These ASP vendors are obliged to demand hefty margins to pay for their standalone application-specific media tone networks. The result is that these services only skim the market instead of encouraging widespread adoption, particularly

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By Eric Weiss

in the SME market.

This piecemeal approach is a strategic dead end for it fails to provide the carrier and network equipment provider with a migration path to unified communications. Furthermore, maintaining separate application specific networks was precisely what IP networks were supposed to eradicate! Yet they persist and are even garnering further investment.

While it's understandable that in the early phases of a market (e.g., Web conferencing) a carrier might offer a service in a "side-car" approach, non-integrated with their network and back-office, once that market has proven acceptance, it's time to provide it as a converged offering. Web conferencing, now a $500 million market with potential to integrate or even subsume another $9 billion in adjacent services, is clearly at that stage. Moreover, the openness of the Internet as a platform for innovative communications should make it much easier to try out new services quickly and easily in an integrated fashion from day one.

Let's consider a collaboration and conferencing example. Some participants are on PCs communicating via VoIP or a traditional public switched telephone network (PSTN), and collaborating via data and video. At the same time, other participants are joining the call via their cell phone or PDA, but they want to join the conference. With true converged communications, carriers can make this possible. The Web conference participants can use location services to pinpoint mobile users, and dial them into the conference, thereby delivering voice, data, video, and mobile services all working together, sharing context and content in a single interface. The ubiquity of Internet bandwidth access across multiple edge devices with Web interfaces — PCs, cell phones, PDAs, etc., now makes it more feasible to integrate such offerings. Equally important, using platforms and tools such as Flash that were once relegated only to the Web, it is now possible for the provider to present its brand in a rich and differentiated manner via a consistent user interface across these different devices, networks, and applications.

The Killer App: Right Under Our Noses

It's not that the carriers and OEMs aren't thinking about convergence. They are. They need, however, a killer application that will drive deployment of integrated applications over their IP networks or equipment. But first let's define "killer app." This simple definition from http://hostingworks.com may be particularly accurate: "The application that actually makes a sustaining market for a promising but under-utilized technology."

The "killer app" in the enterprise is sitting right under our noses…it's converged conferencing — the convergence of Web, audio, and video conferencing, along with some aspects of the collaboration and VoIP markets, as well as rich on-demand media. Converged

The real opportunity lies in converged applications and media types over the Internet and over a variety of different devices.

conferencing is often confused with Web conferencing, a subset of the converged conferencing market. Web conferencing is, however, an important application because, as its name implies, it is the first to leverage the Web as it's communications fabric, thus enabling the integration of the others, as well as broad market accessibility. It is therefore one of the most strategic applications for a carrier to host within its own network and for a network equipment provider OEM to integrate within its solutions.

According to Wainhouse Research, telecom and audio conferencing service providers will double their market share of Web conferencing services sold at the expense of Web conferencing-centric ASP vendors between now and 2007 by using integrated solutions as a lure. Similarly, OEMs that bundle web conferencing technology as a feature of their larger CPE-based applications will increase their market share from two percent to 30 percent of all CPE-based Web conferencing technology sold because enterprises want integrated solutions, instead of standalone Web or audio conferencing.

Now let's test whether converged conferencing meets the definition of a killer app in terms of market size. If one accepts the hypotheses of the industry analyst experts, it's merely a question of quantifying the likely candidate ingredients of a converged conferencing offer. Table 1 illustrates a summary of relevant market sizes based on the author's analysis of the consensus of a few different major market research firms. Readers will find that any major market research firm will not vary much. The order of magnitude is sufficient to support the main argument that it's a big market for those who can provide an integrated offering.

Table 1.

	2005	2007
Audio Conferencing	$3.0 B	$4.0 B
Retail VoIP	$0.5 B	$1.0 B
Collaboration*	$0.3 B	$0.5 B
Web conferencing	$0.5 B	$1.0 B
Video conferencing	$1.5 B	$1.5 B
Rich Media on Demand	$0.2 B	$1.0 B
Total	$6.0 B	$9.0 B

**assumed to be 25–50% of the collaboration market.*

Converged conferencing doesn't fit the definition of a killer app just because of its market size. So is it also a "promising, but underutilized" technology? To be sure, with two-thirds of the market for Web conferencing dominated by North America, it's easy to argue that one of the core offerings of converged conferencing is underutilized. Audio conferencing, while more broadly used, is also underutilized in international environments. The promise of all of these applications is enormous considering the growth rate of globalization, offshore outsourcing and international commerce, and the demands for collaborative communications to bridge geographies.

The beauty of converged conferencing is that given the right platform it leverages existing markets and products… it's just the integration that is new. That makes it easier and more credible to make the claim that it's a viable market worth investing in.

So what's the "right platform"? The platform for converged conferencing has to use the Internet as its communication backbone. Clearly nothing is as ubiquitous or accessible. Private dedicated networks, while acceptable as an on- or off-ramp for the Internet, cannot be effective if operated as closed application-specific media networks. The beauty of the Internet is that if you connect anywhere you connect everywhere. All businesses today need to collaborate instantly and easily with a complex web of supply chain partners and customers that are outside the firewall. Moreover, the openness of the Internet encourages innovation that can be integrated with conferencing solution, further satisfying the end-users' demand for integrated solutions.

Conclusion

The initial phase of IP communications was driven by cost reduction. While compelling, this is no longer sufficient to justify the billions being invested. The real opportunity lies in converged applications and media types over the Internet and over a variety of different devices. Yet carriers and OEMs have been slow to capitalize on this opportunity, lacking focus on a "killer app." That killer app is a combination of several existing conferencing-related applications that end users want as an integration solution. In aggregate this represents a multi-billion market opportunity. To win in this market, providers must leverage a platform that is Internet ready, uses a ubiquitous rich client and is flexible enough to support third-party integration. Converged conferencing paves the way for an even more exciting range of unified communication services. IT

Eric Weiss is vice president of Telecom Solutions at Macromedia (quote - news - alert) For more information, please visit the company online at http://www.macromedia.com.

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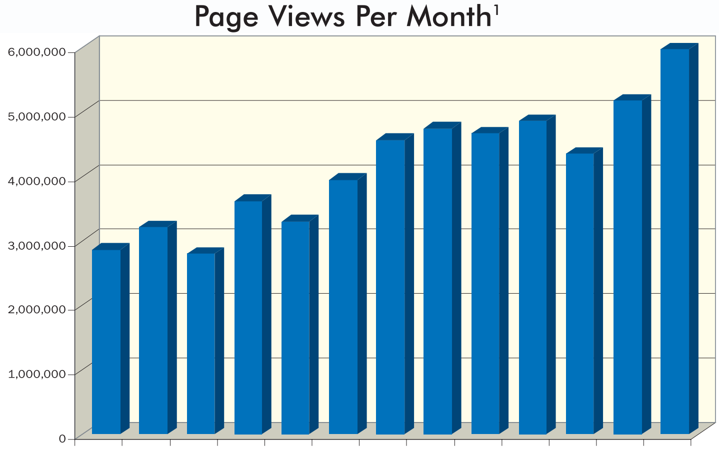

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The IMS Blueprint:

An Introduction To The IP Multimedia Subsystem Architecture

For most of the history of telephony, networks were comprised of circuit switches interconnected by (of course) circuits. In the late 90s, however, service providers began to realize the advantages of shifting their networks to packet technology. At that time, what would actually emerge as the dominant architecture was far from a foregone conclusion. The Internet's IP and the more circuit-like ATM both had their proponents, and carriers were still primarily concerned with how to emulate circuit switch functionality instead of how to improve on it.

Since then, an industry consensus has emerged. Service providers now realize that IP Next Generation Networks (NGNs) delivering bundled, innovative services are what will help them break away from the pack — and the IP Multimedia Subsystem (IMS) has emerged as the standard framework for providing those IP-based communication services.

Wireline Providers Make A Wireless Standard Their Own

Originally defined by the Third Generation Partnership (3GPP) and 3GPP2 wireless standards bodies, the focus of IMS was to provide a new mobile network architecture to enable the convergence of data, speech, and mobile network technology over IP-based infrastructure.

When wireline standards bodies began developing standards for NGN networks supporting voice over broadband, they realized that IMS was the perfect foundation for these efforts, for despite IMS' wireless roots, there is actually very little in IMS that is wireless-specific. IMS can in fact work quite well over any IP access technology.

This agreement on a single set of IMS core principles has implications and represents an important industry trend. Since IMS is now regarded as the underlying architecture for both fixed and mobile networks, it provides a natural base for Fixed-Mobile Convergence (FMC).

Telephony, Meet The Internet

IMS represents a fresh start for the telephony industry. All previous standards grew out of telecom's circuit switching legacy. Softswitching and gateway control protocols like H.248 were simply a way of "decomposing" a circuit switch into several components. Signaling protocols like Bearer Independent Call Control (BICC) were simply a way of carrying SS7 information over a packet network.

In contrast, IMS is based on Internet principles and standards. Intelligence is distributed throughout the network instead of being centralized in a few circuit switches or softswitches. Subscriber and routing information is kept in easily managed central databases instead of being scattered among the devices that might use that information. All of the call signaling is based on the Session Initiation Protocol (SIP).

Standards will play an important role in the industry's transition to NGN. Service providers have learned from experience that deploying networks using proprietary approaches leads to an uncomfortable dependence on one vendor, which explains why according to a recent report by Infonetics research, standards compliance was among the top three criteria network operators used when making purchasing decisions. IMS is a relatively complete set of standards that show how a service provider can construct a complete network. In theory, a carrier with an IMS network could buy any component from any IMS-compliant vendor, and it would all work together.

The IMS Core Network Architecture

It is useful to think of IMS as consisting of three layers:

• The **Applications** layer provides end-user services. Interacting with the

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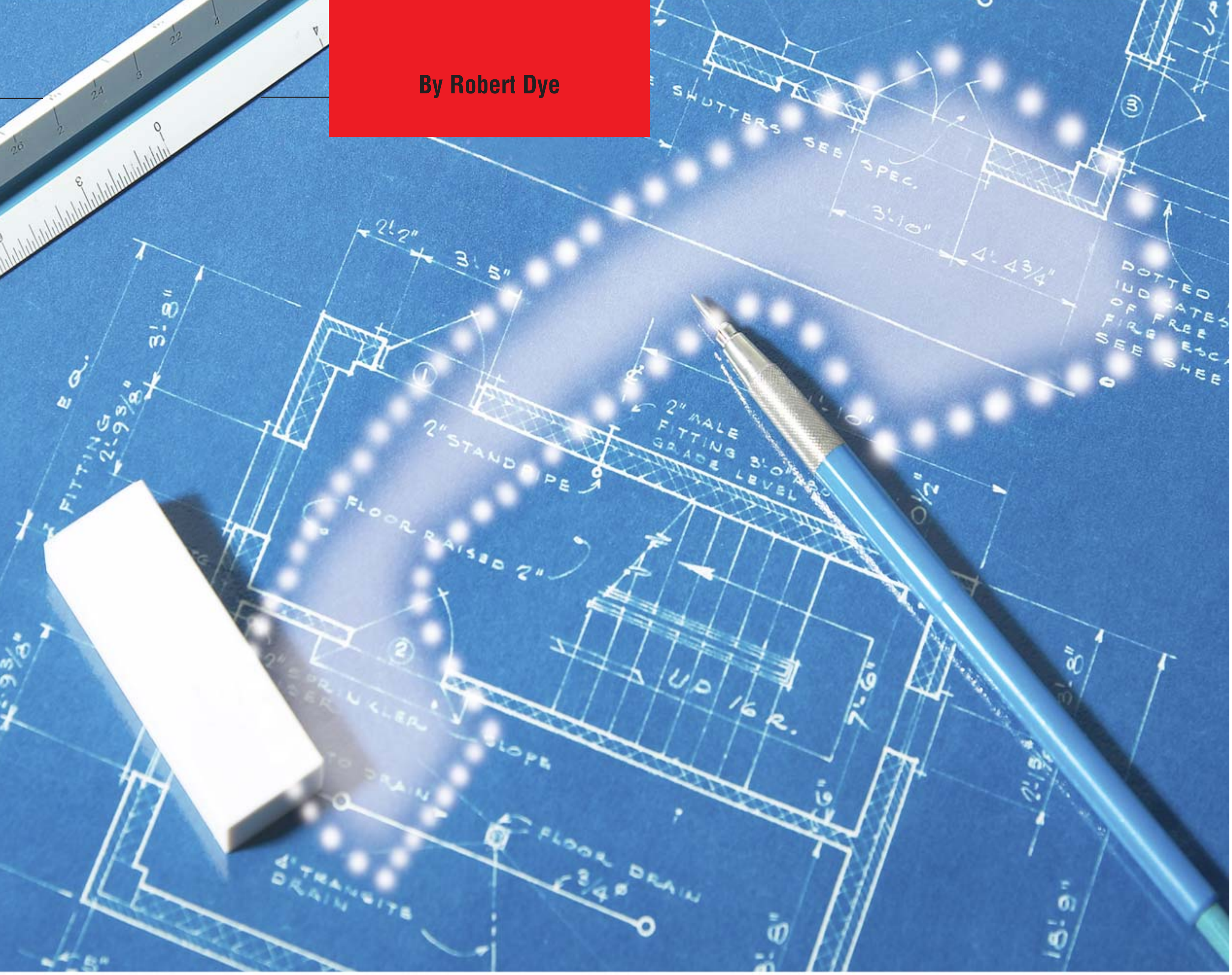
By Robert Dye



rest of the IMS network via SIP, the Applications layer is designed to promote rapid application development and deployment.

• The **Session Control** layer contains the components necessary to maintain the relationship between endpoints, or between applications and endpoints.

• The **Border/Media** layer includes all of the elements that form the border of the IMS core network, interacting with subscriber devices and other networks. It also includes media processing functions, such as systems that provide announcements, conferencing, etc.

The isolation of all outside interaction in the Border Control Layer is one of the fundamental strengths of IMS, as it provides the foundation for convergence. True convergence means a common core network supporting all access technologies, not simply the interconnecting of various silos. The functions in the IMS Border Control Layer can interwork between the native formats of external devices or networks and the common SIP and IP media formats used internally.

The Economic Advantage Of IMS

The economic advantages of an IMS network over a legacy circuit-switched network are clear. Traditionally, carriers have developed and deployed an application-specific network for each new major service. Clearly, operating many separate networks has a significant negative impact to profitability.

In addition, this model makes it extremely difficult for carriers to achieve other important business goals. Carriers have come to recognize that subscribers react very positively to the idea of common services across all types of access. IMS facilitates the implementation of common applications, while the application-specific approach tended to result in application-specific services.

Wireline And Wireless Come Together With IMS

Perhaps the biggest advantage of IMS is that it sets the stage for FMC. Most large, incumbent wireline carriers have a wireless arm, own a substantial interest in a wireless operator, or at the very least have a Mobile Virtual Network Operator (MVNO) service. With IMS, the advantages of a common core network and common applications can even be extended to cover both fixed and mobile networks. Many observers

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feel that common services, such as a shared voicemail box for business, home,and wireless, will be a powerful differentiator in today's much more competitive industry.

What's Next For IMS?

According to a recent report by the Yankee Group, most major carriers expect to deploy an IMS architecture in their networks in the next 18 to 24 months, citing the ability of IMS to promote faster time to market services and a simplified integration process for introducing new services as the primary motivating factors. The Yankee Report also stressed the importance of service continuity and interoperability as two key points of consideration in the migration to an IMS architecture. Failure to maintain service continuity threatens a carrier's ability to hold onto its subscriber base. As carriers begin to seriously evaluate different IMS solutions, one of the most important considerations is to ensure that a vendor's migration path is both seamless and cost efficient. By smartly employing an IMS strategy, carriers are in the position to address some of their biggest business challenges, such as the creation of new revenue streams and the establishment of competitive differentiators. IT

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Understanding The IMS Nomenclature

CSCF — Call Session Control Function

PROXY-CSCF — The P-CSCF is the first entry point for the user equipment (UE) into the IMS.

INTERROGATING-CSCF — The I-CSCF is used for initially choosing the correct S-CSCF.

SERVING-CSCF — The S-CSCF handles call session state in the IMS, including coordinating the use of Application Servers for enhanced services. Also, it is the subscriber registrar and is always in the home network.

BGCF — The Breakout Gateway Control Function routes calls to the public switched telephone network (PSTN) or any other circuit-based network.

MGCF/MGW — The Media Gateway Control Function/Media Gateway handles media to and from the PSTN.

SGW — The Signaling Gateway passes circuit-based signaling to and from the PSTN.

AS — An Application Server (i.e., SIP AS, OSA AS, or CAMEL IM-SSF) offers value-added services and resides in the user's home network or in a third-party location. An AS based in the home network can access the HSS as needed.

HSS — The Home Subscriber Server is the master database for subscriber data, containing subscription-related information to support call processing network entities. The HSS interfaces with call control servers in order to complete routing/roaming procedures by authentication, authorization, naming/ addressing resolution, location dependencies, etc.

VoIP MARKETPLACE

VoIP MARKETPLACE

CEO Spotlight

Craig Rauchle
President and Chief Operating Officer
Inter-Tel, Inc.

In the CEO Spotlight section in *Internet Telephony*® magazine, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Craig Rauchle, President and Chief Operating Officer of Inter-Tel, Inc.

GG: What is Inter-Tel's mission?

CR: Inter-Tel's (news - alert) mission is to continue its legacy of providing advanced technology to businesses, enabling them to improve their most fundamental business challenges, such as increasing revenue, improving operations and controlling costs. Our company has built a solid reputation as having the ability to deliver a tangible and practical return on investment to our customers. We do this by developing applications that improve their critical business processes, and by demonstrating the real economic benefits these technologies provide. We believe our mission — embodied by our ability to deliver value to our customers — will become even more important to the market, particularly as technology itself becomes more complicated, and businesses face new competitive pressures and dynamics.

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

CR: Inter-Tel believes advanced applications that improve business processes will drive the enterprise communications market. As "thought leaders" in developing many of these solutions, such as collaboration and presence management tools, we often times see our competitors feverishly working to deliver their own technologies and duplicate our success. So, we have recently taken several strategic steps to solidify our position.

Among these was the acquisition of Linktivity, Inc., which was highly regarded for their Web-based conferencing and collaboration tools. We have already integrated Linktivity technology into Inter-Tel's communications platform, with the release of Inter-Tel Web Conferencing and Inter-Tel Remote Support. The market is starting to embrace these solutions as a cost-effective alternative to pricey ASP conferencing services like Webex and Centra, so we see a lot of potential here.

Another strategic move was the acquisition of Lake Communications, a company that manufacturers business communications systems for the SOHO (define - news - alert) environment. Lake has been very successful selling its products into the major global service providers like British Telecom, Telecom Italia, and Australia's Commander. From a distribution and product strategy perspective, Lake is a very good fit.

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?

CR: Among the trends we are watching closely is the ongoing debate revolving around standards and interoperability. Inter-Tel is a strong proponent of standards-based technology. We were the first major vendor to integrate SIP at the core of our system. We are operating under the premise that at some point in the future, customers will demand that they can access each of their business-specific applications in whatever infrastructure they choose-whether it is an IP, TDM or wireless network.

Our industry has to be cognizant that we are most successful when we deliver solutions that meet market expectations, and not try to force feed technology down customers' throats. The unfortunate reality is that many in this industry have reputation for doing just that. If VoIP is to continue to gain mindshare and market share, we must make sure we do not repeat the mistakes of the past.

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

CR: There are a number of technologies that we are interested in leveraging, particularly as we look to expand our product portfolios.

Certainly, we continue to see next-generation IP-powered applications as an exciting opportunity for Inter-Tel. Solutions like presence management, collaboration, conferencing, and mobility are changing the way people work as well as communicate. Through these tools, companies are beginning to understand how they can expand their market penetration, compete on a larger stage, capture more revenue, improve customer service, and most notably, reduce expenses. These applications provide a compelling argument to companies that have shown an interest in Voice over IP.

Identifying new vehicles that can deliver these applications throughout an enterprise is extremely interesting to us.

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We're taking a very close look at the potential of WiFi and WiMAX (define - news - alert) technology to determine how we can better serve mobile workers and remote locations through wireless connectivity.

VoIP is constantly re-defining itself to the enterprise.

In addition, security is another area that has drawn our attention. The challenge for Inter-Tel, and the industry as a whole, will be to reinforce security while at the same time allowing for the proliferation of open, interoperable solutions that leverage standardized protocols. We are working very diligently with our partners to develop a mechanism that allows customers the freedom to choose the solutions that make sense for them, without compromising the level of security their businesses require.

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

CR: As we've seen in recent years, VoIP is constantly re-defining itself to the enterprise. When it was first launched, VoIP's primary benefits were articulated to be significant reduction in both toll charges and network management costs. What wasn't understood at that time was that in order to deploy VoIP, most companies needed to invest large sums of money to upgrade their data network. Any reduction in toll charges was negated by price-dropping from carriers and service providers.

Today, VoIP (define - news - alert) is rightfully earning a reputation as a conduit to deliver advanced applications that improve business processes. We can point to VoIP's revenue-generation capabilities as well as its well-documented benefits on the cost side.

In my mind, we will continue to see applications as the primary force driving VoIP growth in the years to come. IP technology provides us with a strong foundation to develop tools that can tangibly impact business performance. SIP, WiMAX, and other technologies will be important, but what will really matter is how well we address customer challenges. That power lies in applications.

While the next few years will be very exciting for Inter-Tel in terms of new technologies and solutions, we are very much committed to staying true to our roots by providing tools that are as practical as they are advanced. IT

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