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INTERNET TELEPHONY®
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VOLUME 9/NUMBER 2 FEBRUARY 2006
The VoIP Authority Since 1998™

Will Apple Finally Launch the VoIPod?
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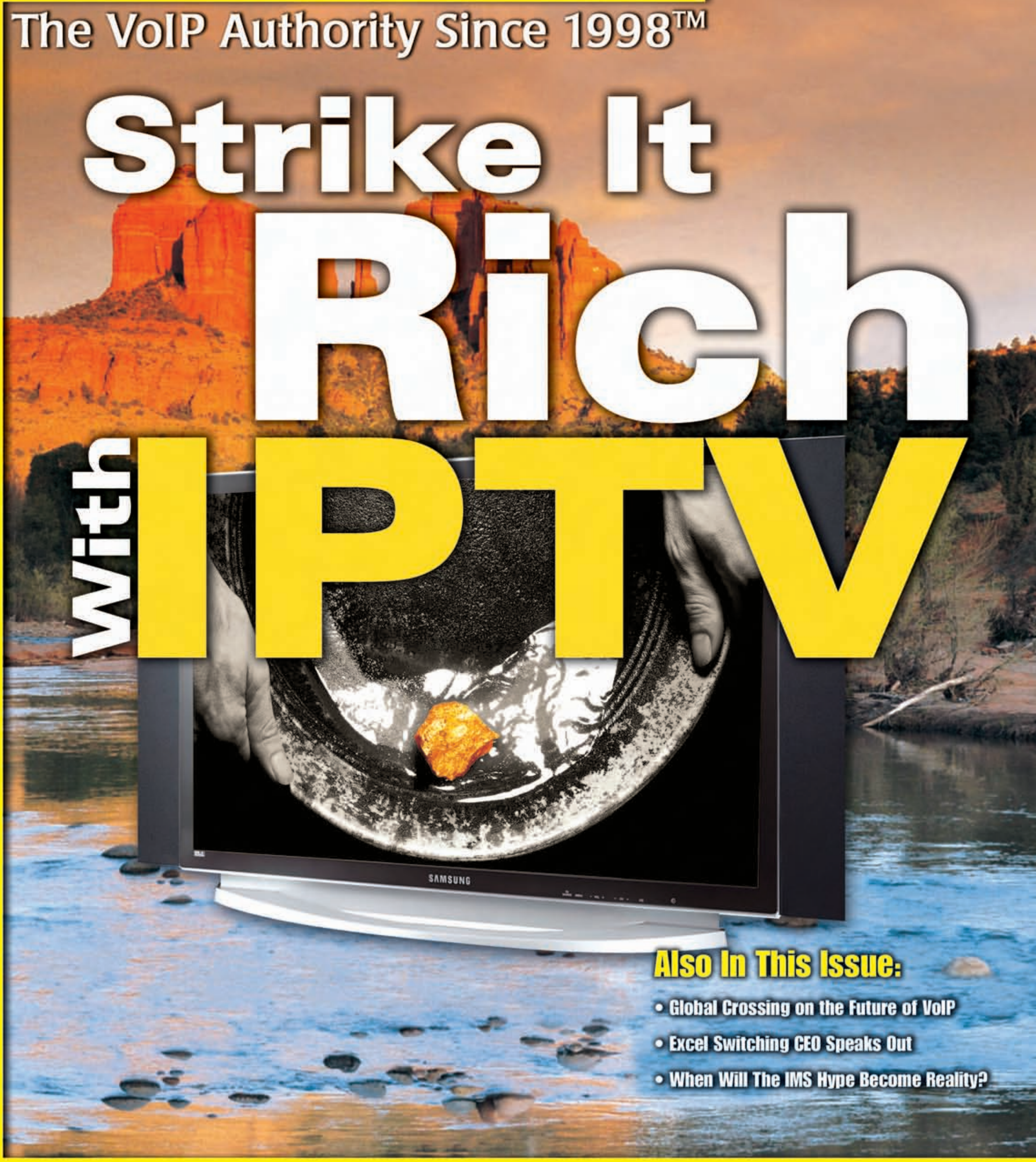
Strike It Rich with IPTV



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INTERNET TELEPHONY® magazine (ISSN: 1098-0008) is published monthly by Technology Marketing Corporation, One Technology Plaza, Norwalk, CT 06854 U.S.A. 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, One Technology Plaza, Norwalk, CT 06854 USA.

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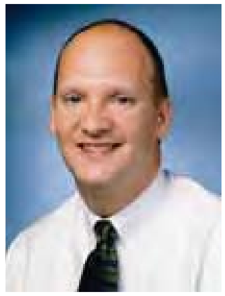


By Greg Galitzine

Mobile Me: Some Predictions Are Easier Than Others

This year is barely six weeks old, and already there's lots of speculation regarding the next "big thing." Who — or what — will be this year's Tech Darling? What will be the "it" that can stand in for the "it" in all those eBay commercials? Whatever "it" is, "it" will have to be more than practical and cool. "It" will have to have attitude.

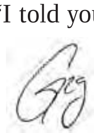
One possibility is that Apple will finally jump into the mobile communications fray with some combination video-iPod-phone-PDA type device. There's even talk in some quarters that Apple may become an MVNO. For those still unaware, MVNO stands for Mobile Virtual Network Operator. Essentially, this is a company, or perhaps better stated, a brand, that positions itself as a wireless operator, catering to a specific niche or community, or simply to a massive audience that might be persuaded to purchase service from that brand. They're not really a phone company with switches and gateways etc…, they partner with an existing carrier for wholesale minutes and all that. They are essentially a marketing machine, driving their brand further into the consumer consciousness. If anyone knows marketing, it's Apple.

So, in this case, Apple (quote - news - alert) might partner with a Sprint (news - alert) or Verizon (quote - news - alert) and promote 'AppleTalk' or 'ChattyMac' or some other service. Of course, that part is pure speculation. But, with Apple's recent filing for trademarks for the term Mobile Me, it seems ever more plausible that they might leverage their strength (the iPod) and move into the phone business.

Apple currently partners with Motorola and Cingular on the Rokr music player/phone, but the word on the street is that the limited capacity of 100 songs is just not enough to draw the interest of real iPod fans.

Our very own Rich Tehrani has been speculating on the impending release of what he calls the VoIPod ever since December 2004. I think this is a case where the prediction will live on long enough that when the device finally arrives, Rich will be able to proudly point to his prediction and say, "I told you so." According to Rich, "Service providers should be looking at the iPOD/iTunes model as how to sell bundled hardware and software and most importantly reduce churn… This concept is part of VoIP 2.0 and what I believe is the future of our industry."

I touched on this phenomenon in a blog entry back in August of last year and, in fact, Apple was near the top of my list as a candidate for becoming a MVNO. I've never been one much enamored with making predictions, but if this comes to be, and one day we're chatting on VoiVideoiPodPhones and buying minutes fro JobsTel, I'll be sure to IM you from what will likely be the "coolest device ever" while I'm watching the trailer for Rocky VII, and tell you, in stereophonic sound, "I told you so."



Errata:

Try as we might, we're not perfect. In the January 2005 issue, we neglected to include Spirent Communications on our Product of the Year list. Oops! Check out the list of winners with product descriptions starting on page 80.

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Top 10 Visitors to TMCnet.com Blogs (by U.S. State)

1. Virginia
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3. New Jersey
4. New York
5. Texas
6. Washington
7. Massachusetts
8. Illinois
9. Florida
10. Colorado

QUOTE OF THE MONTH:

Let's start by looking at history a bit more fairly. Most technological and economic development is, in fact, a series of hyped bubbles followed by sober re-examination and, more often than not, success. For example, speculation around railroads was far ahead of its ability to deliver real services and profits but, ultimately, the industry boomed, crashed, consolidated, and then enjoyed an era of dominance until it, too, was dethroned by the next generation of transportation — namely, automobiles and then airplanes.

Our own communications industry has experienced a series of similar hype cycles around broadband, the Internet, dot coms, NGNs, and 3G, leading many to question the next buzzword, IMS. But several factors indicate that the time is right for next-generation IP networks to mature.

– Grant F. Lenahan (page 94)

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 700,000 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.

Unified Communications for Argentina
Unified Communications provider Esnatech announced its Latin expansion with a partnership with Damovo Argentina to launch and market Esnatech's Telephony Office-LinX solution. http://tmcnet.com/235.1

Broadsoft Leads in Hosted VoIP
It seems like the explosion in companies selling hosted services has had an amazing impact on Broadsoft. The company is touting its accomplishments as of late; they are now delivering services for six of the top 10 and 11 of the top 25 global carriers (based on revenue). They are also delivering the first IMS-compliant application software to market through partners such as Ericsson, Lucent and Italtel. http://tmcnet.com/236.1

The Future of VoIP: A U.S. Evolution
The continued development of VoIP — and specifically wireless VoIP — technology has created a hopeful future for the U.S. market. According to a report that investigates the developments of the VoIP Market in the U.S., Visiongain has found that the incredibly low cost of VoIP is the "major driving factor for its adoption." With companies now capable of offering the benefits of VoIP technology at a fraction of the cost of traditional services, many consumers are buying in. http://tmcnet.com/237.1

Intel Unveils New Consumer Technologies
Intel Corporation has rolled out two new technology suites geared toward bringing digital entertainment to consumers' living rooms through a number of platforms. The company's Viiv technology features a choice of Intel processors, as well as a specialized chipset and software that enable high-definition video and surround-sound audio. http://tmcnet.com/238.1

VoWLAN is Inevitable and Available in 2006 Says Pyramid Research
WLAN has transformed from a network of stand-alone hotspots into a supporting technology for delivering convergent solutions, clearing the way for VoWLAN. According to a recent Pyramid Research report WLAN-Cellular Convergence: The Carrier Business Case for WLAN, UMA and VoWLAN, VoWLAN mid-tier handsets will come to market this year, placing MVNOs, fixed and cable providers in a position to make a strong play for mobile voice revenues. http://tmcnet.com/239.1

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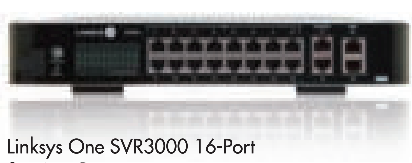

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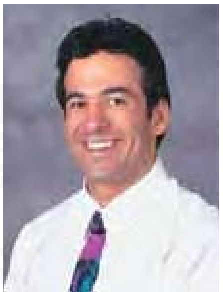

By Rich Tehrani

One Little, Two Little, Three Little Internets…

We live in a country that prides itself on being at the forefront of technology, yet we are behind in many ways. Interestingly, we seem to constantly lag when it comes to communications infrastructure. While we brag that our country invented the Internet and probably 90 percent of the world population realizes how important the Internet is in business and our personal lives, no one in the United States is stepping up to the plate to ensure we will have the best possible Internet.

First a word on cell phones. The U.S. along with a small handful of other countries have standardized on various wireless network technologies. In our country, Sprint (news - alert), Cingular (news - alert), and Verizon (quote - news - alert) all use different telephone standards. While we are used to the way things work domestically, in other countries that have standardized GSM, you can freely take a phone from one provider and use it on another network. You can even choose from a half dozen providers at will. Which system do you think is more competitive and better for consumers? The best cellular devices are routinely developed in other parts of the world. Even when a U.S.-based service provider like Verizon Wireless embraces something universally useful, like Bluetooth, they feel they can get away with crippling all Bluetooth services except those that allow you to connect to a headset. They force you to transfer data to the device by either paying for a cable or by using their network and paying a fee.

Verizon Wireless is holding us hostage. They have the best wireless network in the U.S., so many people don't mind using the Verizon network on the company's terms. As a point of fair disclosure I switched from a GSM provider to Verizon because they have such great service.

Holding people hostage is something you can do if you have little or no competition. Interestingly, there was a time, many years ago, when ISP competition was pretty fierce and hundreds of service providers wanted you to connect to the Internet through them. It got so competitive that companies, such as NetZero, gave away Internet access for free. One would surmise this was absolutely fantastic for consumers.

We are a competitive country. We shouldn't ever lose at anything, especially not the performance of Internet service.

Fast-forward 10 years and your ISP choices for broadband are limited pretty much to cable and DSL. You likely have at most two providers in you area. Take a guess. Do you think prices are decreasing for broadband delivery? No, they aren't.

We may marvel that certain parts of the country have access to broadband over cable and that people in these areas are able to receive broadband at speeds of 4 megabits per second. We call this innovation. We say, "WOW! This is so much better than dialup." What we may not consider, but probably should, is that, in Japan, 100 megabit service costs $25 per month and, in Stockholm, 1,000 megabit (or one gigabit) access costs just over $100 per month.

As if we weren't behind enough already in the U.S., we now need to deal with a new wrinkle in telecom pricing and for lack of a better term, "hostage taking." The LECs are now in negotiation with content providers to sell them access to a second tier of Internet service. In other words, if Google wants to stream video to your computer, they will have to pay service providers to ensure acceptable video quality. What will Google be charged? It is unclear. When you have no competition, you can charge what you want. In fact, the LECs can decide to charge enough so that Google (quote - news - alert) decides it makes more sense to not send you video at all. By the way, this sort of business practice would have killed iTunes before the service got off the ground. The contrarians may think the cable companies will naturally allow Google to stream video without restriction. Yeah right. Are they going to watch Google become a mega-TV broadcaster and eventually watch them take away their cable TV business? I can't see that happening.

The principles that govern giving all content providers fair and equal access to Internet bandwidth is called Net Neutrality, and there are debates taking place right now among government officials about whether this system makes sense for our future. Or is a better solution to be found by allowing a tiered Internet system?

Our government is heavily influenced by lobbyists and incumbent providers are some of the best lobbyists around. There is a great deal of concern by those 'in the know' that the government will allow two levels of Internet to develop.

Imagine for a second that every company that makes a living by doing business on the Internet has to pay fees to ensure their Web pages come up in less than 10 seconds. The potential for abuse is staggering. Will Amazon have to pay massive fees to ensure their customers can still shop online? Would you shop online if every page on a site you tried to reach took 10 or more seconds to load?

Will customers have to start using dialup to access their favorite sites? This is probably far fetched, but what safeguards are in place to ensure service providers won't abuse their powerful monopoly positions?

Service providers argue that they have spent considerable money on their networks and they need to recoup their expenses. The irony is that they also told the government that sharing their lines with the CLECs was unfair because they are spending so much money building their networks, it isn't fair to share. So, the FCC decided to more or less demolish the CLEC market, which now sets the stage for cable and phone companies to exclusively control our access to the Internet. Now we have a duopoly with the potential for abuse.

The phone companies are in the process of developing their own content and also signing distribution deals with content producers to ensure they can become competitive with the cable TV companies. The more content they produce, the more money they will be able to make. They also are in a position to ensure that content providers are at a disadvantage in providing service while their own services receive priority.

How does anyone monitor how much priority traffic actually gets? If your Vonage (news - alert) calls start to sound lousy, you may switch to a competitive service from the LEC… Is this not a compelling reason for LECs and cable companies to tinker with the quality of packets you receive?

Indeed, would services like Vonage or Skype have ever gotten off the ground if these restrictions were in place five years back? The answer is likely no.

We know that any sort of restriction on bandwidth and the holding of content providers hostage is just bad business for the Internet community as a whole. It is beyond my understanding how our government can even entertain such conversations.

If service providers are allowed to threaten content providers, we will enter a new world of Web extortion where everybody loses, especially consumers.

I hope there are people in the government monitoring these issues very closely and that these same people realize how well the Internet works today. Any changes to our system that will allow service providers to hold content providers hostage is a giant step backwards. If anything, in order for service providers to be allowed to keep their government-sanctioned and actually encouraged monopoly status, they should be forced to provide consumers a better price/speed ratio than any other country in the world.

This is what we desperately need. We are a competitive country. We shouldn't ever lose at anything, especially not the performance of Internet service. If the government were to

New Internets

While we focus here in the U.S. on having multiple Internets of varying quality, beyond our shores the trend seems to be having different Internets.

Many countries are looking for alternatives to being forced to use technology that is controlled in some way by the U.S. government. A while back, Europeans decided to launch a multi-billion dollar GPS competitor into orbit called Galileo.

Now, other countries are looking to compete with America's ever-growing Internet dominance. For example, China, the Arab League, and a Dutch company have all started to build their own mini-Internets that are not necessarily available to those on the real, or should I say "first," Internet.

These newer Internets will differ from the one we use today because they will use suffixes different from the 264 that are approved by the Internet Corporation for Assigned Names and Numbers (ICANN). You know these suffixes as .com, .org, .jp, etc. The central root basis for the Internet is what allows it to work so well. The single root, which is replicated for security and redundancy, is necessary so that anyone can get to any Web site easily. When you type in Tehrani.com, the central root communicates rapidly with administrator of your domain and returns an IP address that, in turn, connects you the correct site.

Recently, there was controversy over the addition of a .xxx domain name when the U.S. government twisted the arm of ICANN to squash the new name. Other countries cited this example of how the U.S. controls the Internet and have subsequently pressed for ICANN to be under the UN's control. As the Internet becomes a bigger part of every country's daily lives and economy, the fear of having U.S. control over such an important network is growing.

In response, the U.S. says that countries like China, Libya, Syria, and Cuba, who complain about U.S.-based Internet control, don't have democracies and, as such, taking control of the Internet for them means they will use their power for censorship.

Alternatives to ICANN are also popping up in Europe, where the Open Root Server Network (ORSN) mirrors ICANN and is there almost as a safeguard in case ICANN starts to behave badly. In other words, this root can be used as leverage to ensure ICANN operates in a fair and equitable manner.

Another example is UnifiedRoot in Amsterdam, which allows customers to purchase a domain name with any suffix they choose for $1,000 and a recurring annual fee of $250. ICANN is responding to these threats by becoming more accommodating to foreign languages.

I wonder… If this is the beginning of a slew of new Internets being built, will we be able to easily use VoIP across these networks? In other words, we not only have to have the same VoIP provider, but now we will also need to be on the same Internet?

This whole situation has the ability to confuse users and any confusion is bad. Furthermore, anything that makes it more difficult to communicate is not beneficial for the global population. For more on this topic, visit http://tmcnet.com/240.1

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Did Cable Miss the VoIP Boat With OCAP?

News Analysis By Robert Liu

As my distinguished colleague Hunter Newby notes in his commentary this month, the cable industry is obviously taking VoIP Peering seriously by issuing its Request for Information to learn more about standards and deployment options. But how much is the cable industry really embracing voice ... or Internet Protocol for that matter?

Consider the announcement in January from industry consortium CableLabs. At a press conference at the International Consumer Electronics Show gala, the industry's top honchos banded together to throw their hefty weight behind OpenCable Application Platform (OCAP), a Java-based middleware that CableLabs has been working on with the European-based Digital Video Broadcast (DVB) group since 2001. Executives like Glenn Britt of Time Warner Cable and Brian Roberts of Comcast plan to deploy the middleware later this year into head-ends of their cable systems in a handful of markets, giving software developers the ability to write applications for cable systems for the first time.

The goal isn't just to unify disparate systems but also simplify them by bringing together various functions onto the same platform. That could result in the elimination of the need for a separate set-top box or the evolution of a truly universal remote control, "There's a fight in the consumer's mind as to whose operating system is running it," Roberts explained at an investors' conference a few days after CES.

But in its current form, OCAP still lacks the capabilities to unify voice with data and video services despite the bundling efforts of the cable multi-system operators (MSO). And that could represent an opportunity for IPTV providers to gain a small foothold in the ever-changing bundles known as Triple- or Quadruple-play services.

OCAP is based on DVB's Multimedia Home Platform (MHP) specification and runs in the Java 2, Micro Edition (J2ME) environment. In order for developers to port over their pre-existing Java-based applications onto OCAP, applications need to be re-engineered slightly with an "Xlet" — think, Java applet — so that it can work on the OCAP stack and the platform has the ability to manage the application.

As such, industry officials believe the cable industry's endorsement of OCAP represents a significant win for the entire Java development community. Following that logic, it also is a big loss for Microsoft ASP or .NET developers, who had been making inroads pushing Windows out to the IPTV community.

"That seems to be the way it's playing out in the cable field," said Dilshan De Silva, Director of Marketing, Browser Business at Espial, a Canadian developer of IPTV solutions. In fact, companies like Espial, which has come up with a browser software developer kit built for embedded designs, could benefit by porting its application over to the OCAP platform to support Internet TV services.

OCAP is brought up occasionally as a possible middleware standard for the IPTV community as well. But Telco operators and service providers have not made the same commitment to the MHP standard as the U.S. cable industry has. "The area we're interested in is OCAP being brought up in the IPTV space," De Silva said

"It's a little too early to tell [about OCAP's viability]. You're not seeing the same level of adoption as you're seeing with the cable industry. In IPTV, there's more proprietary middleware," he explained.

The cable industry's goal with OCAP is to come up with applications (like voting/polling, eBay-like e-commerce services, online bill payment, caller ID on TV, etc.) intended to add to the stickiness of the video product. But while the OCAP-based application can operate independently of the networking protocols used to carry it, it is still not going to replace voice-over-IP as a conduit into the home — at least not in its current form.

That's because, while Sun Microsystems — the keeper of all things Holy in the Java realm — has clearly outlined telephony call controls, none of the Java Telephony API (JTAPI) is included in OCAP/MHP. The Quality of Service mechanism still relies on the PacketCable Multimedia Terminal Adapter, according to Don Dulchinos, senior vice president of advanced platforms at CableLabs.

"You'll still need an IP channel," De Silva continued.

Officials believe the cable industry's endorsement of OCAP represents a significant win for the entire Java development community.

So much for bundled subscriber services.

That means cable operators will still need to provide bundled services through two devices, the TV (or set-top box) and the modem. And, even though Microsoft now faces more of an uphill battle getting its runtime environment ported onto cable systems due to wide-scale Java deployment, its developer base has still made more progress bundling voice and data with video, opening the door for IPTV.

To be sure, Dulchinos said CableLabs is still working out the kinks with OCAP. Voice (as well as a number of other features like HTML-support) is clearly under consideration. "It hasn't been written out yet. It's on our roadmap for '06," the CableLabs official said.

"They are trying to get the interactive services out faster than the Telco operators do. It will be a bit of a race," observed De Silva. IT

Robert's 15-year communications career spans from the print world to television and to the Internet. He has covered business and technology writing for Dow Jones, Bloomberg Business News, CNN, and Jupitermedia's internetnews.com. He has served as a producer at CNN, Headline News and A&E Television Networks. You may contact Robert at rliu@tmcnet.com.

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Apex Voice Communications Adds Multi-User Interface to Omnivox AES

APEX Voice Communications, (news - alert) a leading supplier of multi-service platforms for enhanced services and real-time billing solutions to telecommunications carriers, service providers, developers, and enterprises worldwide, announced that it has added a Multi-User Interface to its OmniVox AES multi-service platform for enhanced services solutions. Key features include enhanced account administration, new object control functions, and the greater use of permissions for various account actions.

The new OmniView Operation, Administration, Management and Provisioning (OAM&P) module of OmniVox AES gives systems administrators the ability to create Domains, Groups and Users. For example, Domains are used to partition systems when services are being "sub-leased" to third-parties. Each Domain in return has its own domain administrator that can create User and Group accounts and assign them various permissions and object access capabilities. Groups are used to define a specific set of permissions that can be applied to Users. Also, password storage and verification now uses 160-bit encryption for increased security.

"We have had extremely positive feedback from our customers who have started to implement the new multi-user interface of OmniVox AES," said Elhum Vahdat, executive vice president of APEX Voice Communications. "The new multi-user interface, along with the added security features, continues to keep OmniVox AES at the forefront of multi-service platforms."

http//www.apexvoice.com

RTX Intros Hybrid VoIP Cordless Phone

By Johanne Torres

Wireless product developer RTX America (news - alert) has introduced its PORTALphone, a hybrid, Web-enabled cordless phone that allows consumers to view customized Web-based content on a color screen.

The new phone bundles VoIP-based calling with standard telephony and personalized info delivery in a household cordless telephone. The phone is optimized for Casabi's platform to offer rich Web-based content services that have traditionally been tethered to a PC. Customers can now access news, local traffic and weather reports, stock portfolios, entertainment, personal address books, buddy lists and instant messages (IM).

The RTX PORTALphone allows customers to make and receive calls anywhere in and around the house. The included PORTALphone base station simultaneously connects to a traditional telephone socket and a broadband Internet connection. The device supports SIP-based VoIP telephony as well as standard landline (POTS) telephony.

"We believe that the PORTALphone gives carriers a unique opportunity to differentiate their VoIP services and add value beyond just voice," said Curtis Schmidek, vice president of marketing at RTX America. "And customers get the best of both worlds. They can have personalized rich Web content and VoIP capabilities on a telephone that seamlessly integrates with their traditional telephone line."

http//www.rtxamerica.com

Siemens Enters Router Arena with Acquisition of Harbour Networks Products

By Patrick Barnard

Siemens, (quote - news - alert) the world's third largest mobile telecom network equipment supplier, has reportedly acquired core assets of Harbour Networks Co., a broadband IP network equipment provider in China, for $110 million in cash.

According to a news report, Siemens officially signed the purchase agreement with Harbour Networks and, as part of the deal, Siemens will acquire the Beijing-based company's technologies and patents for high-end broadband products, as well as about 100 of Harbour Network's engineers. The profit generated from these assets reportedly accounts for approximately 60 percent of the total.

The deal marks Siemens' entry into the router arena. The company sells carrier Ethernet switches, but has relied on a reseller deal with Juniper Networks Inc. for its routers. Some industry experts have said the relationship between the two companies may become strained if Siemens starts to sell its own router products.

http//www.siemens.com

ShoreTel Marries Converged Conferencing Solution with IP Telephony Solution

By Erik Linask

California-based ShoreTel (news - alert) is one of the fastest growing providers of IP telephony solutions in the United States. Not coincidentally, it also happens to be the industry leader in customer satisfaction. ShoreTel was the pioneering force that first introduced Converged Conferencing to the IP telephony world back in 2003.

Now, ShoreTel has introduced Converged Conferencing 5.6, the latest in its line of innovative conferencing and collaboration platforms, designed to work in conjunction with the ShoreTel 6 IP telephony system. Together, the two will provide integrated audio and Web conferencing, enterprise instant messaging, and document sharing. The solution, hosted locally, rather than as an Internet-based service, helps ShoreTel's customers reduce costs, while providing a stable telephony infrastructure and improved employee productivity.

"ShoreTel Converged Conferencing gives enterprises the freedom to abandon expensive conferencing services while enhancing collaboration and productivity across the enterprise," said Steve Timmerman, vice president of marketing for ShoreTel. "By extending ShoreTel's IP telephony system to include Converged Conferencing, we are addressing customers' key business needs and delivering a compelling return on their investment."

With Converged Conferencing, ShoreTel provides a next-generation conferencing platform combining voice and data in a single, unified medium. Now, because ShoreTel Converged Conferencing is fully integrated with the ShoreTel's phone system, it removes the cost barriers traditionally associated with in-house collaboration solutions. It also ends the reliance on and costs associated with external conference service providers.

http//www.shoretel.com

Whaleback Systems Releases SMB1500

Whaleback Systems (news - alert) has released the SMB1500, a groundbreaking business phone solution that operates over a broadband connection instead of a traditional phone line to maximize cost savings and deliver cutting-edge capabilities.

Whaleback Systems reduces companies' telecommunications expenses and eliminates fluctuating monthly costs with an affordable, all-inclusive, flat rate per station service package. The package includes unlimited calling throughout North America, an IP multi-line executive handset, voice mail, video calling, desktop messaging, remote access, caller ID, speed dial, call forwarding, direct inward dialing, local number portability, auto-attendant, first-class sound quality, e-911 coverage and U.S.-based technical support. In addition, Whaleback Systems provides a free server to support the network and free upgrades to prevent obsolescence.

"Whaleback Systems developed a service package to ensure our cutting-edge technology is immediately affordable for businesses of all sizes," shared Ken Stess, Whaleback Systems' vice president of business development.

"We integrated advanced technologies to engineer a system with easy to use features that improve staff productivity and streamline internal and external communications," said Wray West, Whaleback Systems' co-founder and vice president of engineering.

http//www.whalebackystems.com

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PowerDsine Announces Next-Generation Power over Ethernet Midspan Series

PowerDsine (news - alert) announced the launch of its next generation 6500 PoE Midspan series. The new IEEE 802.3af-compliant 6500 PoE Midspan series is reportedly the first PoE midspan to offer advanced secured Network Management System and the only PoE Midspan to ship with a lifetime warranty.

The PowerDsine 6500 PoE Midspans will be available in 6-, 12-, 24- and 48-port versions. The 6500 Midspans are designed for enterprises, which are deploying IP phones, wireless LAN access points or IP surveillance cameras. Midspans allow networks the added functionality of PoE without the need to replace existing Ethernet switches. This optimizes PoE port count, improves the ROI of networks and saves costs associated with these deployments.

"The 6500 PoE Midspan Family provides enterprises a new level of security with this advanced remote management solution," said Igal Rotem, PowerDsine's CEO. "Moreover, we have such confidence in the 6500 Midspans' reliability that we are offering customers a lifetime warranty."

http//www.powerdsine.com

Primal Solutions and TMNG Partner to Tackle the Challenges of IP Transaction Management

Primal Solutions, (news - alert) the leading provider of IP transaction platforms, and The Management Network Group (TMNG), (news - alert) a leading provider of management consulting services to the global communications industry, announced a new strategic partnership. This relationship combines Primal Solutions' IP Correlytics platform with TMNG's Digital Services Assurance program to deliver a revenue assurance solution for the cable industry as it moves to sustain market leadership in an aggressively competitive marketplace.

In this new environment, cable operators and other advanced communications and new media companies are offering new IP-based services and content. Increasingly, delivery of these new services necessitates complex relationships and delivery models that create revenue assurance challenges. The TMNG and Primal Solutions partnership addresses these challenges by offering an end-to-end program designed to maximize revenues, by linking all stages of product planning and deployment, to ensure successful launch, operations, and profitability.

"We are very excited about this relationship because TMNG and Primal Solutions bring together the technology, people, processes and professional services required to address these complex and growing business challenges," said Bob Richardson, executive vice president of operations at Primal Solutions, Inc.

"After listening to our MSO clients, TMNG chose Primal Solutions because we believe its IP Correlytics platform is best in its class, and unique in its abilities to enable our clients to maintain the integrity of their revenue streams as they launch new voice, video, high speed data and content services," said Bill Opet, principal and vice president of TMNG's cable practice.

http//www.primal.com
http//www.tmng.com

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PMC Telecom Prepares for Continued VoIP Market Expansion

PMC Telecom, (news - alert) already one of the leading suppliers of telephony equipment in the UK, is preparing for continued expansion of VoIP use in Britain, with new product ranges to cope with demand. VoIP — Voice over IP — is fast becoming a revolution in telephone use. With just a home computer and a phone that will connect to it, people can call anywhere in the world for less cost than using fixed landlines or mobile devices.

According to Paul Conway, the Managing Director of PMC Telecom, “2005 has shown that VoIP isn’t going to be a passing phase. People realize that VoIP’s promise of low-cost internet phone calls isn’t a gimmick, it’s a reality. It’s a real consumer market, and one that’s expanding rapidly.”

And VoIP isn’t simply just for the home user. There are real advantages for business with VoIP as well.

“Businesses are waking up to the idea that they can cut down their telephony costs simply and easily,” said Paul. “Heavy users, such as call centers, are moving to VoIP use.”

Whichever VoIP network supplier becomes dominant in the UK, PMC Telecom has moved itself into a prime position to supply all aspects of the domestic and business VoIP product market.

http//www.pmctelecom.co.uk

Tekelec, Convergin Enable MiRS' IMS Future

By Patrick Barnard

Tekelec, (news - alert) a developer of telecommunications products for next-generation fixed, mobile, and packet networks, and Convergin, (news - alert) a provider of core fixed mobile convergence (FMC) solutions, have provided Israeli wireless provider MiRS with the Tekelec next-generation gateway mobile switching center (GMSC) server solution, the companies announced today.

The solution will provide MiRS with next-generation capabilities, including FMC and hosted session initiation protocol (SIP) services, and will facilitate its transition to the Internet protocol multimedia subsystem (IMS) architecture.

A major part of the solution is Convergin's Accolade Wireless Convergence Server (WCS), which is designed to serve in the core of mobile and fixed networks.

Tekelec's IMS architecture drives switching for both MiRS' entire mobile device line and SIP-based clients, and it supports the execution of common network applications. The solution also includes Tekelec's next-generation gateway mobile switching center (GMSC) server solution and its 9000 Distributed Switching Solution (DSS). Tekelec will package the Accolade WCS as a product in the Tekelec wireless convergence gateway (WCG) family of products.

"By enabling the convergence of SIP and cellular, MiRS is well positioned to capture additional lucrative markets as well as to strengthen its position in its existing business and enterprise markets," said Dr. Ayal Itkovitz, CEO of Convergin.

http//www.tekelec.com
http//www.convergin.com

Avaya and Polycom Team to Offer Complete IP Video Telephony Solution

By Erik Linask

Placing a phone call from your office in New York to a colleague in San Diego is one thing. Doing the same via IP is another. But being able to see your colleague by simply placing a phone call takes business communication and productivity to an entirely new level and helps to speed up decision-making processes.

Avaya (quote - news - alert) and Polycom (news - alert) have teamed to offer a complete business communications solution that includes desktop video conferencing, group video conferencing, and multipoint video conferencing. Avaya IP Video Telephony Solution integrates Avaya Communication Manager software for IP telephony with desktop video conferencing, group video conferencing, and multipoint network solutions from Polycom.

"With single number dialing for video and voice, and call control functions that make video easier to use and content easier to share, the Avaya - Polycom solution makes video an integral part of a communications applications suite," said Micky Tsui, Avaya vice president and general manager, Communications Systems Division.

The Avaya/Polycom collaborative solution provides a single, easy-to-use, easy-to-deploy voice/video infrastructure. Enterprises will benefit from simplified integration of video across locations, lowers cost of ownership, and scalability — the solution is based on established industry standards for communications, including SIP, H.323, H.320 and 802.11, and is able to accommodate more than 750,000 IP video stations on a single network. Naturally, network administrators also will benefit from having a single, unified communications environment to manage.

"Avaya is a longstanding Polycom strategic partner in video telephony and this is a critical milestone in our joint development of IP-based, next generation, video telephony conferencing and collaboration solutions," said Ed Ellett, senior vice president and general manager of Video Communications at Polycom. "Now the best video experience is available to our joint customers through Avaya's familiar softphone interface. Video collaboration is now as simple as making a phone call."

http//www.avaya.com
http//www.polycom.com


Considering VoIP?
Know, know, know. A thousand times, know.
The move to VoIP can save substantially, or cost dearly, depending on how much you know. With NetIQ's end-to-end VoIP Management, you know for sure that your network is ready for VoIP and that your infrastructure is protected from the unique security risks it poses. You also know your voice quality is assured. As a leader in systems and security management, NetIQ offers you unrivaled effectiveness in managing the overall performance, availability and security of your VoIP environment. Now, isn't that good to know?
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Rangers and Stars Pick XO for Data and Voice

By Johanne Torres

Telecom service provider XO Communications Inc. (news - alert) announced it scored a three-year agreement to provide voice and data services to the Texas Rangers and Dallas Stars.

The agreement calls for XO to deploy communications systems for the Rangers at the Ameriquest Field in Arlington, Texas and for the Stars at the Frisco headquarters and at three Dr Pepper StarCenters located across the Metroplex. With the communications system in place, XO will be able to integrate all four locations.

The deployment will increase the Rangers' voice capacity by 25 percent at Ameriquest Field, while doubling available data bandwidth. Stars headquarters will be experience a similar increase in data bandwidth as well.

The agreement also asks for XO to install its XOptions Flex VoIP system at the Dr Pepper StarCenters. Because Flex works with the existing communications infrastructure, there will not be a need to invest in new VoIP equipment to benefit from Flex's dynamic bandwidth allocation across a shared voice and data connection, producing a minimum of three times the currently available bandwidth at those locations.

"Businesses throughout the Metroplex want four things from their communications provider, product, performance, predictability and price," said James McLendon, XO Communications' general manager for the Dallas market. "We look forward to delivering all four to the Texas Rangers and Dallas Stars for the next three years, and beyond."

http://www.xo.com

AT&T Launches IPTV Service in Texas

By Patrick Barnard

AT&T's (quote - news - alert) new TV service, which is delivered via IP technology, boasts 200 channels, including HBO, MTV, ESPN, Discovery Channel and A&E, along with all three major broadcast networks. It also offers several hours of on-demand programming.

According to company officials, the new service differs from the TV service already available from cable companies. One difference is that it offers fast channel changes, so that when viewers flip through channels, there is no lag. This is a common problem with many digital cable TV systems. The service also offers picture-in-picture channel surfing, so that viewers can continue watching their current show while getting a glimpse of what is available on other channels.

One of the primary differences between AT&T's offering and Verizon's FiOS TV — which was announced the same day — is that FiOS TV is delivered via fiber-to-the-home, whereas AT&T is extending fiber only to nodes which are close to homes (also known as "fiber-to-the-curb"). Because it has limited bandwidth to devote to its video service, AT&T is delivering it using IP technology.

Even though AT&T's service costs much less to deploy, its network is riskier from a technology perspective because it uses new and evolving technology. Many experts agree that eventually all TV networks will use IP to provide more interactive content, which means all eyes will be on AT&T as it expands its service later this year.

http://www.att.com

Skype 2.0 Lets Users Talk Face to Face

By Patrick Barnard

Skype Technologies (news - alert) announced that version 2.0 of its free VoIP software for Windows — including video capability — and version 1.4 for Macintosh are now available.

The updated version of Skype for Windows offers not only better call quality, but also free one-to-one video conversations. It also allows users to organize their contacts by creating groups and provides new buttons to display their Skype status on their blog or Web page. Currently, 2.0 is available only for Windows 2000 or XP.

Skype 1.4 for Mac OSX has more significant upgrades, including the addition of call forwarding and call auto-answer. Users also can set up calls more quickly and receive notification of new messages. As an added feature, Version 1.4 — which works with Mac OSX v10.3 or newer — can automatically pause iTunes when a call is answered.

Skype 2.0 has been in public beta since December and is the first Skype release to include full-screen video calling.

"Our goal is to make technology easy to use, and Skype is a simple Internet communications service that is changing the way people stay in touch," James Bilefield, vice president of business development for Skype, said in a statement.

http://www.skype.com

Penn State Selects Qovia for VoIP Management

By Susan J. Campbell

The education market — college campuses, in particular — have fully adopted IP telephony systems for their powerful, money-saving features that offer an alternative to traditional phone systems. The size of the systems, however, poses a challenge for providers to maintain both call quality and system security.

According to David Woodall, Qovia CEO, (news - alert) the institutions in the education market are among the most advanced in their acceptance of VoIP. However, because they are on the cutting edge, they are also the first to feel the full effects of the limitations of the systems. Deep insight into their systems is needed to ensure that voice quality remains high, while the systems remain secure.

Penn State has invested more than five years in research into VoIP as it started the process in 1998. In 2002, Cisco systems were deployed in earnest. Home now to approximately 6,000 VoIP phones, the University Park campus continues to increase its VoIP integration.

Security for Penn State has been a priority from the onset. The university chose to lock down handsets to specific ports instead of allowing the portability of the handset, negating one of the key features and benefits of VoIP.

Only predefined IP telephones are attached to the network, according to Phil Coolick, network services manager in the Information Technology Services department at Penn State. The MAC address of the telephone is locked to the switch port, protecting the core equipment from problems such as viruses by eliminating the possibility of a rogue device, such as a laptop, being used in a VoIP port.

The bottom line is that Penn State will realize not only benefits from the use of the VoIP, it will also realize significant cost savings; that, in and of itself, is a major selling point for a public university.

http://www.qovia.com

Vonage Selects Sonus Networks Architecture to Support Expansion

By Laura Stotler

Vonage (news - alert) is supporting its continuously growing subscriber base through a newly announced agreement with Sonus Networks. (news - alert) The Sonus architecture, including the GSX9000 Open Services Switch, PSX Call Routing Server, SGX Signaling Gateway and Sonus Insight Management system, is being rolled out in Los Angeles and New York. Vonage plans to expand their network to support additional markets in the United States, and eventually broaden to international locations.

The Sonus solution enables Vonage to easily add new services while reliably increasing their subscriber base. Vonage has already deployed more than one million lines throughout its network. The Sonus IMS-based Voice over Broadband suite of solutions, improves operators' abilities to easily make changes to their service offerings, and has already been deployed in broadband VoIP networks throughout the world.

"Voice over broadband is a technology that's redefining how service providers and consumers alike think about communications," said Hassan Ahmed, chairman and CEO, Sonus Networks. "From day one, Vonage has been one of the most successful service providers in the industry. By selecting Sonus Networks, Vonage will have the ability to build out a premium network infrastructure to support the rapidly increasing demand for its services."

"Vonage has been a VoIP pioneer in North America," said Michael Tribolet, executive vice president of operations for Vonage Network. "Throughout Vonage's history we have remained committed to providing high-quality telephony service. By selecting Sonus, another pioneer in the VoBB market, we are in a position to continue to do this, while rapidly and successfully scaling the size of our network."

http://www.vonage.com
http://www.sonusnet.com

Dash911 to Launch E911 Service for VoIP Providers

By Johanne Torres

"Dash911's (news - alert) architecture offers coverage to any telephone number, and with our SOAP API interface, we can quickly implement a VoIP provider within ten days," said Michael Giagnocavo, Dash911's CTO.

Dash911 is "telephone number provider agnostic," therefore providing E911 services to any USA telephone number, at any USA address.

"Our relationship with our backbone provider allows us to provide an incredibly rich service, because they are the largest and most experienced company in the emergency calling services field," noted sales engineer Denise Ferrara.

This news follows Dash911's request in October for a 21-day extension for all of its customers from the FCC. The FCC had required that all VoIP providers deploy an E911 solution for subscribers by November 28, 2005. Dash911 requested an extension for its customers "due to the unprecedented damage caused by hurricanes Katrina, Rita and Wilma which has caused administrative hardships and infrastructure disruptions for many smaller VoIP companies."

http://www.dash911.com

Level 3 Provides E-911 Solutions For Packet8 Subscribers

By Laura Stotler

8x8, Inc. (news - alert) has chosen Level 3 Communications (news - alert) to provide E-911 capabilities to 8x8's Packet8 fixed and mobile customers. Level 3 will provide the underlying network for the E-911 solution using their E-911 Direct service to reach Packet8's VoIP and videophone services customers.

Level 3's E-911 Direct is a portfolio of solutions including FCC compliance for nomadic VoIP providers and a fixed-line solution with network connections to Public Service Answering Points (PSAPs) that reach about 69 percent of U.S. households. The solutions deliver address-specific and callback information to PSAPs when VoIP users make a 911 call. This enables first responders to be dispatched to the scene even if callers are unable to speak or if the call is disconnected.

"We chose to broaden our work with Level 3 because they understand our commitment to consumer safety and have built a sophisticated network that has allowed us to quickly implement an E-911 solution that meets the FCC's requirements for nomadic and fixed-line VoIP," said Bryan Martin, chairman and CEO of 8x8.

"Although the FCC recently amended the requirements for VoIP providers to be E-911 compliant, it is compliance that continues to be an urgent hurdle for the industry to overcome," said Myrle McNeal, senior vice president of Local Voice Services for Level 3. "8x8 is ahead of the curve, and we are glad to be working closely with them to solve the need for E-911. It is crucial for the safety of consumers and the momentum of our industry that VoIP providers accelerate their efforts to comply with FCC regulations around E-911."

http://www.level3.com
http://www.8x8.com

Telecom New Zealand International Selects Veraz's VoIP Solution for U.S. and U.K. Deployments

Veraz Networks, Inc., (news - alert) a leading global provider of VoIP softswitch solutions, announced that Telecom New Zealand International selected Veraz's VoIP softswitch to be the preferred solution for TNZI's telecom switches.

Initially, TNZI (news - alert) will deploy Veraz's ControlSwitch softswitch, service delivery platform and I-Gate 4000 family of media gateways in key U.S. and U.K. cities. This initial deployment will be a phased rollout that will be completed in early 2006 and will include installations in Los Angeles, New York, Miami, and London.

"The strength of Veraz's offering is its open and flexible architecture and, in particular, the flexibility of their system's voice traffic routing that can adapt to TNZI's business agreements now and in the future," said Anthony Briscoe, General Manager of Telecom New Zealand International. "Veraz's open architecture paves the way for TNZI to build new business models using best of breed application servers where appropriate, and fits with TNZI's strategy of building its business on voice application wholesale rather than transactional minutes based revenue."

"TNZI wanted a new switching and operational model in order to scale its business and meet future demands," said Doug Sabella, CEO of Veraz Networks.

"This is a great win for Veraz in a major market and demonstrates the strength and flexibility of our solutions as well as our ability to deliver what is required in the market of today and tomorrow."

http://www.veraznetworks.com
http://www.tnzi.co.nz

BellSouth and 8x8 Team Up To Deliver Residential VoIP

By Patrick Barnard

BellSouth Corp. (news - alert) announced today that it will be partnering with 8x8 Inc. (news - alert) in launching a new residential VoIP service. The new service, BellSouth Digital Phone Service, will use 8x8's Packet8 service.

Martin Chandler, vice president of product management for BellSouth, said in a company press release that the new service "is another example of our commitment to providing customers with the greatest choice when it comes to their communications and entertainment services."

The BellSouth Digital Phone Service is based on 8x8's internally developed technology. This includes a suite of VoIP service components including a call switching platform, feature servers, customer portals and consumer premise equipment.

"BellSouth's selection of 8x8 as its VoIP partner is a tremendous credit to the technology and service expertise which has made 8x8 one of the industry's leading VoIP service providers," said 8x8 chairman and CEO Bryan Martin. "Our extensive residential, business and video technology portfolio was a key component to our winning BellSouth's business. We look forward to working closely with BellSouth to offer their customers an enhanced, robust Internet phone service alternative."

http://www.bellsouth.com
http://www.8x8.com

Vonage Now Provides E911 in Over 1700 Calling Centers

By Johanne Torres

New Jersey-based VoIP service provider Vonage Holdings Corp. (news - alert) now has E911 service in over 1,700 PSAPs across the country. Vonage claims that, in just one week, the company equipped an additional 112 calling centers in more than 50 new counties with 911 calling capabilities.

"Vonage has been continuing to quickly roll out Enhanced 911 service to new counties across the U.S.," said Jeffrey A. Citron, Vonage's chairman and CEO. "Our goal is to get every single Vonage customer help when they need it, and we will continue working with the FCC, regulators, Congress and public safety until there is equal access to E911 for Vonage customers."

The VoIP provider currently offers E911 access in the U.S. by sending the call, along with the customer's address and phone number, to the proper local emergency call center based on the caller's street address. The caller's info is then displayed on the dispatcher's screen whenever they dial the digits 9-1-1 from a Vonage phone. In the event local authorities cannot display the Vonage customer's phone number or address, Vonage offers basic 911.

http://www.vonage.com

Navizon Transforms Cell Phones and WiFi Devices into A GPS Device

Mexens Technology announced Navizon, (news - alert) the world's first software-based, peer-produced, wireless positioning network. Navizon is a software system that can be loaded onto a mobile device to provide its users with geographical positioning information plus many more advanced customized features. Navizon's innovation arises from the fact that its map is both created, and constantly enhanced, by the users themselves.

Navizon's software is currently compatible with Windows mobile devices and cell phones using the Symbian operating system, and is free for non-commercial use. Navizon members are allowed access to a massive storage and retrieval repository containing highly accurate positioning data which is contributed from members around the world.

"The soft launch of Navizon has been successfully tested, and now we are very excited to grow the community even more," said Houri. "Navizon utilizes WiFi and cellular signals to help you find your way around most cities and to provide you with pertinent information about your location. What makes Navizon unique is that the network will continue to expand as the community expands. This shared information is at the very center of what makes Navizon a powerful tool."

http://www.navizon.com

Ekahau Deploys WiFi-Based Location Technology for Emergency Management

In the event of an earthquake or train derailment that results in mass casualties, hospitals are confronted with simultaneously treating hundreds of patients with varying degrees of injuries. To meet the challenge of managing patients, checking the availability of doctors and locating open beds and medical equipment in wide-scale emergency, Nagoya Ekisaikai Hospital in Japan has deployed and successfully tested the Real Time Location System (RTLS) from Ekahau Inc.

The Ekahau RTLS is a mission-critical location tracking solution that easily integrates with WiFi networks that are already in use in many large public and private facilities, such as Nagoya Ekisaikai Hospital.

"The ability to incorporate a location tracking solution on top of existing WiFi networks provides hospitals and other facilities with a cost effective way to prepare for natural or man-made disasters and mitigate the chaos associated with these events. The Ekahau RTLS gives hospital staff the real-time information they need to ensure quality care and ultimately save lives," said Jarmo Ikonen, Ekahau's director of sales, EMEA/APAC. "Hospitals represent just one target segment for such a location-aware emergency management system. Enterprises and public agencies, such as those tasked with homeland security, also can benefit from this technology."

During a recent disaster simulation drill, Nagoya Ekisaikai Hospital put WiFi tags on each hospital staff member and triage patient. Once the exercise was complete, Nagoya Ekisaikai Hospital reported that with the T201 WiFi tags and the Ekahau Positioning Engine, it was able to locate patients and staff with an accuracy of three to five meters.

http://www.ekahau.com

Tele Atlas and Skyhook Wireless Announce Agreement

Tele Atlas, (news - alert) a leading global geographic content provider, and Skyhook Wireless, (news - alert) provider of the industry's first WiFi Positioning System (WPS), announced today an agreement aimed at delivering next generation location-aware solutions.

The companies will work together to provide application developers with a single source for location-aware geographic content. The agreement will not only expand the addressable market for location-aware applications to any WiFi enabled laptop, PDA or mobile phone, but will also enhance the location coverage available in environments where traditional location technologies fail.

"The ability to easily pinpoint a specific location on a map — whether indoors or out — is imperative for the fast developing location-based services (LBS) applications market," said Michael Shean, co-founder and vice president of business development, Skyhook Wireless. "Tele Atlas has a superior process for ensuring its maps and other geographic content are highly accurate and completely up-to-date, as well as a long heritage of serving the very complex emergency services market. They are also the ideal partner, with the best database, for the emerging location application market."

"Skyhook's WPS technology breaks traditional barriers in metropolitan markets, and will deliver a significant advancement to our partners," said Mike Gerling, chief operating officer, Tele Atlas, the Americas. "The combination of Skyhook's WPS solutions and Tele Atlas street navigation and geocoding data means mobile consumers of all stripes and emergency services personnel can, without interruption, locate any place, product, or person."

http://www.teleatlas.com
http://www.skyhookwireless.com

U.K. Cities To Get Blanket WiFi Coverage

The United Kingdom has unveiled plans for citywide WiFi networks that will give residents in nine cities high-speed wireless Internet access from laptops, PDAs, and mobile phones.

The first phase of the project, due to be completed by March, will see citywide WiFi hot spots rolled out in Birmingham, Cambridge, Edinburgh, Leeds, Liverpool, Manchester, Nottingham and Oxford, along with the London boroughs of Kensington and Chelsea, Camden and Islington.

The networks are being built by European wireless provider The Cloud. (news - alert) They will be open to any Internet service provider that wants to offer services. Blanket wireless coverage will be provided in the cities through WiFi equipment fitted on lampposts and street signs.

People who want to use the wireless network will pay one of the ISPs for access, and revenues will be split between The Cloud, the local council and the ISPs.

WiFi coverage for more cities is expected to be announced later this year. George Polk, CEO of The Cloud, said the aim is to provide wireless coverage across all U.K. cities and major centers of population.

"Providing ubiquitous wireless broadband access over a network that is available to millions of WiFi devices...will have a major impact on the way people communicate, work and play in city centers," Polk said in a statement.

http://www.thecloud.net

Broadcom Delivers Revolutionary Solution for Video IP Phones

Broadcom Corporation, (news - alert) a global leader in wired and wireless broadband communications semiconductors, today announced the world's first WiFi video phone chipset for wireless handsets and desktop video IP phones. The chipset and its associated software leverage several market-leading Broadcom(R) technologies including VoIP, mobile multimedia and wireless LAN (WLAN). The revolutionary video IP phone chipset further expands

Broadcom's leadership position in the IP phone chip market.

Video phones that are based on Broadcom's new WiFi video phone chipset will change voice communications into a completely new form that is practical and fun. Imagine business travelers saying good night to their children "face-to-face" by simply dialing home, or teenagers being able to compare outfits before they meet up for an evening out together, or families being able to celebrate special occasions together despite the distances that separate them.

"The age of video telephony is finally here. Our ability to integrate several of Broadcom's leading technologies will enable our customers to bring to market cost-effective, low-power video phones with world-class video quality," said Paul Shore, Director of Marketing of Broadcom's VoIP phone products. "Video phones on the market today lack the price point and video quality needed for mass market adoption — our new chipset and software will radically improve both of these key market drivers, establishing Broadcom as the market leader in video IP phones."

http://www.broadcom.com

Internet TV and Satellite Radio Come Together With Wi-Fi TV InterSat

Wi-Fi TV Inc., (news - alert) which streamed the first full-length movie on the Internet in 1995 and, in 2005, became a pioneer in Internet TV, announced it is offering global TV stations, radio stations, and content producers a new distribution platform.

The Wi-Fi TV distribution platform is called InterSat because it utilizes both satellite and Internet delivery technologies in tandem to offer a global delivery platform compatible and accessible to the widest possible number of TV stations, radio stations and content producers.

Wi-Fi TV InterSat, like satellite radio, offers stations the opportunity to reach a large geographic audience and to do so without the licensing associated with traditional TV and radio. Wi-Fi TV InterSat delivery also offers a number of interactive features not possible with traditional TV and radio.

The Wi-Fi TV InterSat system was successfully implemented by Wi-Fi TV in a Beta format for two live events: New Year's Eve from Times Square and the Tarver-Jones 3 pre-fight weigh in. In each case, live coverage from cameras on the scene were uploaded to a satellite, retrieved from the satellite at the location of Wi-Fi TV Internet servers, encoded on the fly, and streamed by Wi-Fi TV both live and archived to a global audience. In the case of the New Year's Eve Webcast, a live Wi-Fi TV chat feature included comments from viewers.

http://www.wi-fitv.com

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ICOA Powers WiFi at Stop & Shop Locations

ICOA, Inc., (news - alert) a national provider of wireless broadband Internet networks and managed services in high-traffic public locations, announced that it is powering Stop & Shop's WiFi initiative.

Stop & Shop tested the technology in their corporate headquarters waiting room this summer. Based on the success of that trial, the company, which has more than 360 stores throughout New England, New York, and New Jersey, has begun rolling out free amenity wireless Internet access for its customers. The first store to offer this service is Stop & Shop's South Bay Plaza outlet in Dorchester, Massachusetts.

"We are pleased to have been selected by a service oriented company such as Stop & Shop," said ICOA's CEO Rick Schiffmann. "Providing Internet access in their cafe seating area is another service that helps their customers accomplish more during their busy day."

"The ability to stop and surf, and maybe have a meal or a cup of coffee is yet another convenience for our customers," adds Stop & Shop's Mike Drumm. "We look forward to working closely with ICOA during the rollout to optimize our overall customer experience."

Installation of the service at additional Stop & Shop locations is expected to begin early in 2006.

http://www.icoacorp.com

TRENDnet's New Mobile Solution Combines Bluetooth and 802.11g

TRENDnet, (news - alert) a best-in-class networking manufacturer of wired and wireless networking solutions announced its latest mobile innovation, the TBW-103UB 802.11g WiFi & Bluetooth Combo USB Adapter. The device integrates WiFi and Bluetooth access into one USB 2.0 adapter to enable the seamless transfer of information between Bluetooth enabled devices, such as cell phones, and computers.

By integrating the two wireless access tools into one unit, a bridge between the Bluetooth-equipped cell phone and the PC has been created. While Bluetooth has become a common feature in newer cell phones, it is not always included as a standard feature in PC's. By connecting the TBW-103UB to a regular laptop USB port, mobile users can now sync contact files, share photos, listen to music or make VoIP calls via their Bluetooth-equipped cell phone or headset. Simultaneously, users can surf the Internet via their WiFi connection.

"This unit is the latest product in a line of solutions recently launched intended to improve the productivity and lifestyle of laptop users," stated Heath Gregory, Marketing Director for TRENDnet. "By combining these two standards into one solution we are helping users make the most of their mobile devices and enhance their ability to distribute their content and stay connected anywhere."

http://www.trendnet.com

Phihong's One-Port Midspan Supports High-Speed Wireless Access Points

Phihong USA (news - alert) has developed a one-port midspan for high-power devices that are capable of powering 802.11N access points. Typical applications for the 30W midspan include wireless/WiMax network access points, security cameras and IP telephones with streaming video.

"Along with the new standard, customers are looking for equipment that is interoperable across multiple vendor platforms," said Keith Hopwood, vice president of marketing for Phihong USA. "Our 30W one-port midspan provides Power-over-Ethernet technology that does just that."

Phihong is a member of the new IEEE 802.3at task force, which was formed to increase the power levels distributed via Ethernet to at least 45W. When implemented as a standard, the PoEPlus initiative will more than double the wattage available to powered devices.

The 30W midspan features diagnostic LEDs and is fully compliant with the IEEE 802.3af standard in detection, disconnect and voltage control. The device is also Gigabit-compatible.

http://www.phihongusa.com

Get Connected

Convert your PC into a communication center!

Nero SIPPS Connect offers simple and cost-effective VoIP solutions for all PC users connected globally.

Nero SIPPS Connect is an intuitive internet phone that offers you all the features you expect along with a high level of customization. Use the 3-way calling feature to conference in several parties at one time, customize interface and ring/knock tones, select On-Hold music, forward calls, personalize greetings, record your conversations and play back, manage address book, setup instant messaging, and utilize a webcam for the ultimate connection*. High quality headset included to optimize your SIPPS Connect experience.

With two SIPPS Connect licenses included in one package, you can share the VoIP experience with family or friends everywhere. Call your friends and family at home or on their mobile phones at extremely competitive rates!

2005 INTERNET TELEPHONY PRODUCT OF THE YEAR

DIAL

TALK

UNITE

*Webcam not included

www.nero.com

Available at participating retailers, including:

Buy.com

COMPUSA

amazon.com

Intel Launches Viiv Entertainment PC

Chip-maker Intel Corp. (quote - news - alert) launched its Viiv entertainment PC platform and announced a slew of deals to provide content for the new systems.

Viiv PC owners will be able to watch video that's stored in Google Inc.'s video service, high-definition highlights from NBC's coverage of the Winter Olympics and classic TV shows from America Online. In all, Intel has signed dozens of content deals.

Viiv computers will be capable of replacing the array of standalone boxes that surround the television — such as a digital video recorder, DVD player or cable box. Intel also says it's working to ensure a Viiv PC works seamlessly with other equipment.

By focusing on more than chips, Intel hopes its technical and marketing muscle will help make entertainment PCs easier to use — and more appealing.

"With our new platforms, we're not only boosting wireless computing, but also advancing digital entertainment a few steps closer to effortless," Intel CEO Paul Otellini said.

Some of the options include support for a technology that allows Viiv users to quickly turn their systems on and off after the initial boot. Machines also will ship with at least 5.1 surround sound and support for high-definition video. TV capabilities are optional.

http://www.intel.com

IPTV and HDTV Video Content Creation Sped Up with Tarari Encoder Accelerator

By Laura Stotler

Creating Windows Media and SMPTE VC-1 video content for consumer electronics devices is now quicker and easier, thanks to a new hardware video encoder accelerator from Tarari (news - alert) and its partners. The Tarari Encoder Accelerator for Windows Media speeds up the creation of video content for devices that play HD video over the Internet, through VOD systems, via IPTV systems, on set-top boxes, on HDTVs and on next-generation DVDs.

The new product features expanded support for SMPTE VC-1 encoding, in addition to Microsoft WMV-9. That Tarari accelerator reduces the time to encode both standard-definition and high-definition video content. It is integrated with Microsoft's encoding software stack, available via Windows Media Format SDK and the GUI encoder application.

The Encoder Accelerator directly offloads functions within the Microsoft encoder, offering up to a 10x boost in encoding speeds. The Microsoft codec can automatically detect if Tarari hardware is present and subsequently offloads complex encoding tasks, leaving existing workflows unchanged. The plug-and-play Tarari board may be installed in a server or workstation to enable the additional encoding performance.

"Tarari's new Encoder Accelerator makes a significant contribution toward enabling quick output of high-quality video content for network distribution through the new media channels now emerging in business, education, and entertainment," said Randy Smerik, Tarari CEO. "Tarari's partnership with Microsoft has produced an acceleration processor that is capable of seamlessly working with practically any application that can encode Windows Media and SMPTE VC-1 video."

http://www.tarari.com

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Tadiran's Coral IPx Office.™

You can survive VoIP with a smart solution.

For the last 40 years Tadiran Telecom has been quietly solving the telecommunication needs of the world's largest organizations. The Coral IPx Office™ is designed to provide flexible voice and data in a single box. Perfect for a 100-user corporate office or a 5-user branch office. Use your favorite SIP phone or Tadiran's FlexIP feature phone. Survive the migration to IP.

For a free "IP Survivability" T-shirt and more information on Tadiran Telecom solutions, call us at 866-595-4900 or visit us online at:

TadiranAmerica

www.tadiranamerica.com

Brix Offers Improved IPTV Service Assurance and Monitoring

By Laura Stotler

Brix Networks (news - alert) has expanded its IPTV service assurance and monitoring capabilities to offer the required visibility into channel change times and network performance related to multicast infrastructure. The company's solutions can now assure quality of bundled services, while providing network operators with service lifecycle support. The solutions provide extensive visibility into delivery infrastructure from super head ends to central offices to subscribers, helping to ensure the success of IPTV services.

The Brix System of integrated hardware and software products proactively monitors VoIP and IP video quality. The solutions also enable providers to measure user response times and monitor network performance and service degradation related to running dynamic IP video services over IP. Providers may also test simulated channels by emulating set-top boxes for zap and VOD function delays using the solutions.

"Service quality matters, and consumers have a higher expectation of quality when it comes to IPTV," said Robert Travis, director of product marketing at Brix Networks. "Working closely with our customers, we know that measuring channel change times and monitoring IP infrastructure performance is imperative to the success of rolling out quality IPTV services. These factors, combined with the growing demand for bundled services, have made service assurance a requirement to the successful deployment of revenue-generating services."

http://www.brixnetworks.com

New Cisco Products Pave Way for Free Muni WiFi

By Patrick Barnard

Cisco Systems (quote - news - alert) announced it has developed a set of new products that will make it easier and less expensive for municipalities to deploy free WiFi Internet access over large, outdoor areas. The company claims that by harnessing the power of a new technology known as "wireless mesh networking," it can set up cities and towns with free public WiFi — something that many city officials now view as essential in order to boost economic development and improve municipal operations.

The new technology was born out of Cisco's acquisition of Airespace, Inc., a provider of centralized wireless local area network equipment (including mesh architecture), earlier this year. By combining Airespace's technology with Cisco wireless products, the company developed its "wireless mesh" offering, which takes conventional WiFi "hot spots" and links them together, much the same way routers link the access points on a wired network.

In essence, mesh technology extends WiFi service from "spot" coverage to "blanket" coverage, making it possible for users to be continuously connected while traveling through a large area. Mesh access points can be installed on street lights, power poles, or other public infrastructure in a matter of minutes. The only requirement is a source of power.

One advantage of mesh technology is that it makes centralized management much simpler than with previous WiFi technology. For example, if one access point fails, the network can continue to operate just as it did before, only losing reception around the incapacitated access point.

http://www.cisco.com

NEC Develops Multicore Processor Technology Enabling Automatic Parallelization of Application Programs

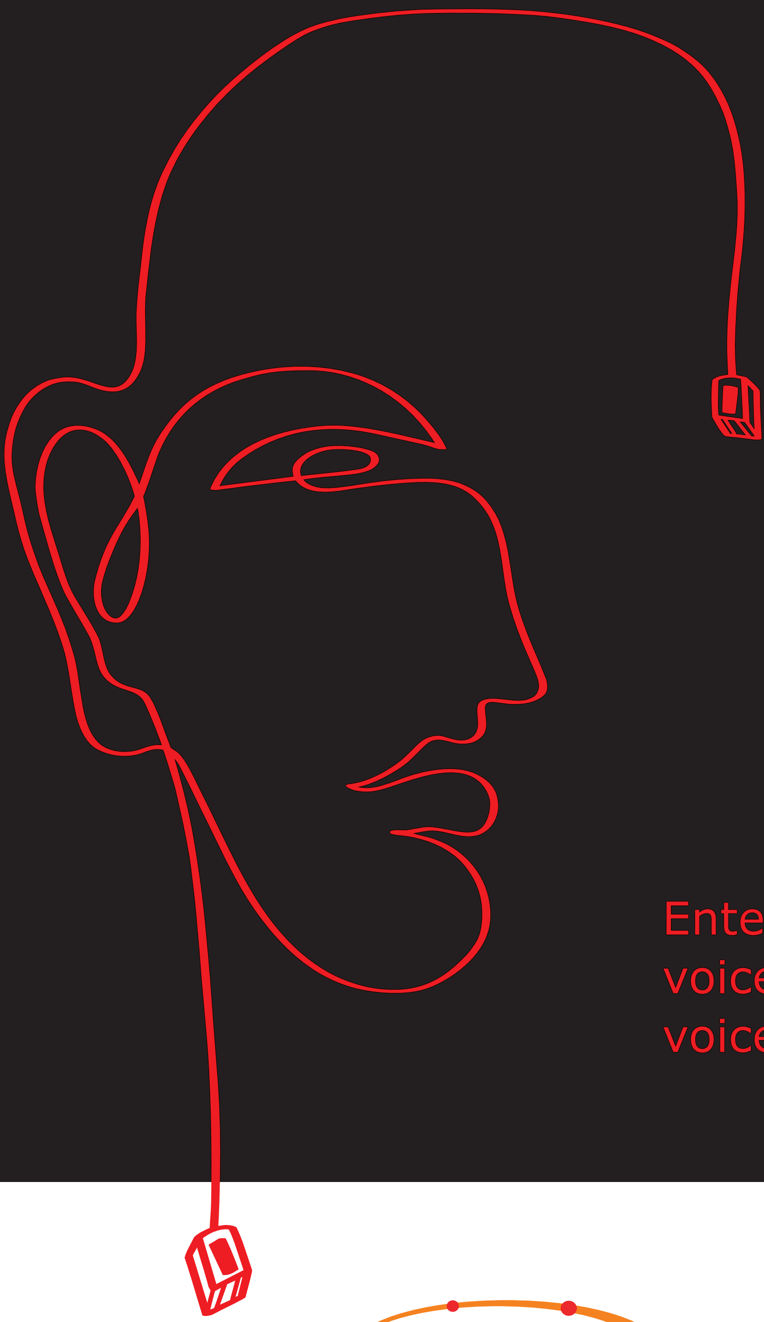
NEC Corporation (news - alert) announced that it has succeeded in the development of multicore processor technology capable of performing automatic parallelization of application programs, without modifying them.

The distinctive feature of this new technology is the ability of the automatic parallelizing compiler that utilizes profile information to aggressively exploit parallelization patterns, which are effective for accelerating the speed of application programs. In addition, although the parallelization is speculative, the speculation is almost always completely accurate.

The speculation hardware works as a safety net by handling any rare misses, guaranteeing the correctness of the execution. This ensures that the compiler is not conservative in decisions concerned with these cases, resulting in an increase in the amount of parallelism exploited. The parallelism exploitation is supported by the speculative execution hardware that realizes efficient handling of detection of incorrect execution orders caused by the parallel execution of the program parts, cancellation of the incorrectly executed part, and re-execution of it. Moreover, the parallelization process can be performed in a practical period of time.

NEC believes that its automatic parallelization technology is the first to be brought to a stage of practical use. This is supported by the fact that NEC has succeeded in operating this technology on a field-programmable gate array (FPGA). Moreover, its implementation has confirmed that only a marginal hardware extension is required and that application program speed is actually accelerated.

http://www.necus.com



Intel Bows Developer Tools Designed for Dual-Core Apples

By Robert Liu

One day after legendary Steven Jobs convinced Intel's President and CEO Paul Otellini to don its hallmark white suit and unveil the new Apple Computer (news - alert) models based on x86 processors, Intel bowed new software development tools and resources specifically for dual-core processors.

As part of the Intel Core Duo processor tools and resources, the company launched free, beta versions of the Intel Fortran Compiler, Intel C++ Compiler, Intel Math Kernel Library and Intel Integrated Performance Primitives for developers to try out as well as other resources to assist with software optimization, dual-core threading and migration information. And officials with its software products division hint that is only the tip of the iceberg.

"We look forward to developer feedback prior to introducing our products in the coming months," said William Savage, general manager of Intel's Software Products Division. "Our highly optimized compilers and libraries yield significant performance advantages for applications and take advantage of the opportunities made available through multi–core and multi–threaded environments."

These development tools are integrated into Apple's Xcode development environment and offer an alternative to existing tools and compilers. The Intel Fortran compiler enables the scientific and technical community to bring the fastest versions of their applications to Mac OS X and Intel-based Mac platforms using industry-standard math libraries and language. The Intel C++ Compiler provides the ability for Xcode users to apply targeted optimizations to performance-sensitive areas of their applications, allowing them to get the most out of the Intel Core Duo processor.

http://www.apple.com

Arista Launches its New Compact, Low-power Industrial 6.4-inch LCD Panel Computer

Arista Corporation, (news - alert) a leading industrial PC solutions provider, introduced its new industrial 6.4-inch LCD Panel Computer, the ARP-2606AP. Designed to meet the demands for mission critical, high-performance, and reliable operation, the unit is ideal for factory automation, facility monitoring, machine automation, and environmental monitoring.

The new ARP-2606AP is equipped with 6.4" LCD and 3.5" embedded board, which incorporates a VIA Eden 667MHz processor. System memory for the unit comprises one 144-pin SO-DIMM socket with up to 512MB of memory capacity. The Local Area Network (LAN) uses a Realtek 8139 C PCI PnP Base-T Ethernet controller. Video support features a built-in VGA controller with up to 32MB of shared memory for display.

The system comes with three RS-232 ports, a four-Wire resistive touch screen and an optional PCMCIA for wireless applications. In addition, the ARP-2606AP has an on-board CompactFlash Type-1 socket, an optional 2.5-inch hard disk drive (HDD) and one 16-bit PC/104 extension connector.

"The new compact ARP-2606AP is ideal for compact environments," said David Marich, sales manager for Arista Corporation. "The unit's rugged design and its panel-mount capability ensure easy installation for a variety of industrial applications. All of Arista's panel computers are backed by our optional next day replacement warranty, further ensuring our commitment to quality and excellence."

http://www.aristaipc.com

Phihong's Redundant Power Source Provides up to 180 PoE Ports for VoIP, IT Applications

Phihong USA, (news - alert) a global leader in Power-over-Ethernet solutions, has developed a redundant power source with up to 180 PoE ports to meet the needs of telecom applications, such as VoIP systems. The RPS accepts three 500-watt power supplies for powering VoIP phones, DC UPS, and lighting systems with single UPS. A typical VoIP phone draws less than 8 watts. With 1500 watts available, 180 PoE phones can be supported from this redundant source.

"Redundant power sources are essential to VoIP systems, particularly for businesses where downtime due to power failure can significantly compromise operations," said Keith Hopwood, vice president of marketing for Phihong USA. "If VoIP power is provided by PoE devices, an RPS is necessary to keep power generating during an outage."

Each output in the redundant power source is individually protected against overloads causing overheating of wiring from excessive current, thus fully protecting the system against overload, over-temperature and over-voltage. It features diagnostic capabilities and LEDs next to each connector. The RPS was designed to allow easy connection of a battery plant for full DC UPS capability for VoIP phone systems or network switches. A single battery plant can provide a UPS function for over 180 IP phones or a complete rack of network switches.

http://www.phihongusa.com

Octasic Expands its Voice Processing Product Line with New Devices

Octasic Semiconductor (news - alert) is enhancing its OCT6100 family with the introduction of lower power, lower cost devices, enabling higher density and the highest voice quality for customers designing down to four channels. The OCT6100L builds upon the proven success of the current OCT6100 devices, offering unprecedented Voice Quality Enhancement features through the use of 130nm technology.

As echo cancellation moves from a standalone box to an internal embedded feature of gateway and switching equipment, designers are faced with increasing demands for lower power to meet their density targets, while improving quality. Octasic has responded to this demand with the OCT6100L. The OCT6100L proves Octasic's commitment to compress product development time for their customers and provide cost efficient, power efficient, performance driven silicon, while ensuring the longevity of the product for the enterprise and carrier equipment market space.

"With over 15 million ports deployed, the OCT6100 has been a great success. We are pleased with the acceptance of this architecture. Now, with the OCT6100L, Octasic can respond to the most stringent customer requirements for density and cost," explains Doug Morrissey, Octasic CTO.

By retaining all the features and functionality of the existing OCT6100 devices the OCT6100L leverages the extensive field experience of the existing devices which have been qualified into several leading carrier networks.
http://www.octasic.com

Picolight Addresses Needs of Market With New 1310nm VCSEL

By Susan J. Campbell

Picolight, Inc., (news - alert) a designer and manufacturer of optical transceivers and components, is the first to ship 1310 nanometer (nm) vertical cavity surface-emitting laser (VCSEL) transceivers in a 4 gigabit per second (Gbps) triple-rate Small Form Factor Pluggable configuration to target one of the fastest growing segments of the data center market. The new transceivers offer extended reach and lower power consumption for short-to-medium distance applications.

VCSEL technology emits light vertically through the surface of water as opposed to Fabry-Perot and differential feedback lasers that emit light through the edges. VCSEL devices require very little electrical current to produce optical energy output of 850nm and above and are easy to use to transmit light into an optical fiber as they emit narrow, circular beams.

The 1310nm VCSELs from Picolight deliver lower power consumption, lower electromagnetic interference (EMI) and lower heat generation to produce increased performance and reliability over single-mode fiber according to Vidya Sharma, vice president of marketing for Picolight. Sharma's position is that next-generation high-density form factors will strongly favor the exclusive use of VCSEL technology due to its low heat and low EMI-generation characteristics compared to that of edge-emitting lasers.

Highlights of the 1310nm VCSEL include the use of 850nm transceiver architecture; single transceiver architecture addressing multiple specifications at 4km and 10km, reaching up to 40km; an un-cooled 1310nm oxide-confined, high-speed VCSEL coupled to an LC optical connector; extended temperature and voltage range options; internal AC coupling on both transmit and receive data signals; all-metal housing for increased EMI shielding.

http://www.picolight.com

Concurrent Rolls Out MediaHawk 4500 Server for VOD Applications

By Laura Stotler

A new high-density server ideal for video-on-demand (VOD) applications is now available from Concurrent. (news - alert) The company has announced its MediaHawk 4500 server, enabling video service providers to store more than 20,000 hours of on-demand content and enabling VOD ingest rates of more than 2,500 hours per day.

The new server offers improved stream density of up to 2,400 streams per 2RU enclosure, enabling small and large deployments with minimal power consumption and footprint. The MediaHawk offers a scalable RAM cache approach and a commercial hardware design, enabling service providers to employ a scalable, pay-as-you-grow deployment model.

"Since introducing video-on-demand to the US market in 1998, Concurrent's expanding portfolio of MediaHawk products have successfully met the growing needs and ambitions of video service providers the world over. The MediaHawk 4500 takes VOD server capabilities to the next level, delivering significantly improved performance with the utmost in platform reliability, while meeting the stringent economic demands of our customers," said Gary Trimm, CEO of Concurrent.

"Concurrent was the first VOD server company to offer independent scalability of ingest, streaming and storage," said Bob Chism, Concurrent's CTO. "The MediaHawk 4500 offers a fourth dimension of flexibility, enabling RAM cache to be scaled as needed. The result is an ideal combination of high performance and economic attractiveness." Chism added, "The MediaHawk 4500 platform is fully compatible with existing MediaHawk deployments, allowing Concurrent's existing customers to take advantage of this breakthrough technology without the need for forklift upgrades."

http://www.ccur.com

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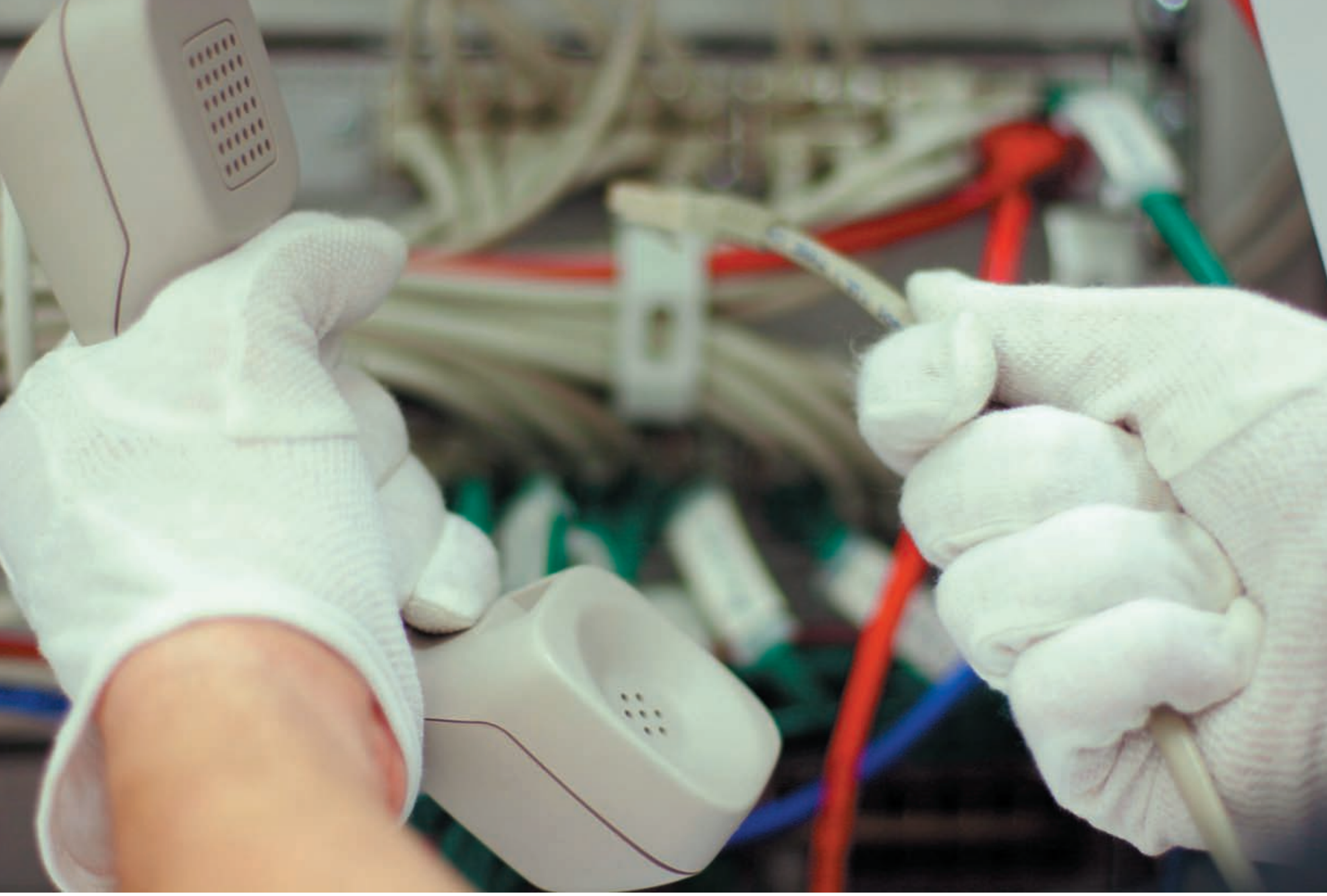


Pactolus Intros InService High Availability Software to Prevent IP Service Session Loss

Pactolus Communications Software Corporation (news - alert) introduced its InService multi-site high availability architecture, an innovative framework to provide its SIPware carrier service applications with unprecedented resiliency and the fastest achievable recovery speeds.

The first InService integration is for Pactolus' widely-deployed SIPware Audio Conferencing, making it the industry's first IP service application to help network operators and service providers ensure uninterrupted subscriber service delivery during both occasional IP network component failures and extraordinary disruption events,.

"Momentary IP service outages can be particularly apparent in hosted multi-user service sessions. Even a fraction of a second's interruption can force all audio conference participants to terminate and restart a call session, which is why many established carriers have balked at offering IP-based conferencing service, despite their obvious economic, flexibility, and innovation advantages," notes analyst Deb Mielke of Treilage. "Pactolus is the first IP services vendor that has addressed this need, which opens important doors for IP service delivery for both new and established service providers."

"The ability to encapsulate a SIP-based services architectures with 'Tier 1 Telco-Grade' reliability and call state protection hasn't been available, which has slowed migration to IP services for some Tier 1 service providers," noted Chris Brimhall of XO Communications.

"Pactolus has been uniquely innovative in identifying and responding to this issue, first with CallComplete for service sustainability, and now with the session resiliency that only a multi-site availability architecture can offer. The ability to support calls across multiple POPs simultaneously — and the intelligence to automate call state preservation during a disruption — are definitely capabilities that help redefine and 'market-harden' IP services."

http://www.pactolus.com

Linksys Announces SIP-Based IP PBX, Desktop Phones, and Gateway for Internet Telephony Service Providers

Linksys, a Division of Cisco Systems, Inc., (news - alert) announced a new line of SIP-based telephony products for Internet Telephony Service Providers (ITSPs) targeting large residential, SOHO, and very small business customers. The new line of IP communication solutions includes an IP PBX/Key system, a wide range of IP desktop phones, and an analog gateway for connection to the PSTN. Used together with an ITSP voice service, they provide a complete IP telephony system for up to 16 users.

"With the new Linksys SIP-based IP communication offerings, ITSPs can offer residential and small businesses a voice service with many of the features found in large business voice IP networks, such as multi-line service, music on hold, auto attendant, and more at a more affordable price," said Jan Fandrianto, vice president of voice engineering at Linksys. "The new IP PBX and IP phones bundled with a service provider offering will make the deployment of voice networks easy to install and simple to use at a price small businesses can afford."

This new solution will complement the recently announced Linksys One solution. Linksys One is an ideal solution for small business with 5 100 users needing a complete communications solution that addresses voice, video, and applications. The LVS series was developed to address the residential and very small business of 1-4 users that may grow to 16.

http://www.linksys.com

Vertical Introduces InstantOffice 7.0

Vertical Communications (news - alert) announced Vertical InstantOffice 7.0, the new release of its award-winning integrated communications platform for retailers and other large, distributed enterprises. InstantOffice 7.0 takes advantage of the Session Initiation Protocol (SIP) standard, which allows customers to create a true channel to manage voice across their entire networks.

"Vertical InstantOffice was designed as a unified platform, to make it simple for our customers to streamline the need for multiple devices," said Bill Tauscher, Vertical's Chairman and CEO. "In InstantOffice 7.0, we have made it even easier for our customers to use the latest voice, data and application services technologies. In leveraging SIP, we are providing our customers with a critical advantage for better serving their customers and thus creating the opportunity for more loyal and profitable customer relationships."

Designed for retailers with hundreds or thousands of stores, InstantOffice 7.0 is a integrated communications platform that consolidates voice, data networking, and voice applications, thus allowing customers to track, manage and optimize voice communications in a flexible, timely way to support their business goals. Because these organizations are continually looking for more cost-effective ways to provide better customer service, InstantOffice 7.0 now leverages SIP to provide the greater productivity benefits and lower costs delivered through Voice over IP (VoIP).

Importantly however, InstantOffice 7.0 does not force customers to commit to a pure IP environment. The open, standards-based platform offers investment protection for existing resources and allows customers to leverage new technologies as they emerge and as their business needs dictate. As part of the company's open IP approach, InstantOffice 7.0 allows customers to choose the best mix of phones for their businesses, including digital, analog, wireless and IP phones.

http://www.vertical.com

IMS/SIP-Based Services Delivery Made Easier Through Pactolus and Convedia Partnership

By Laura Stotler

Pactolus Communications Software Corporation (news - alert) and Convedia Corporation (news - alert) are teaming up to accelerate and simplify deployment of media-rich services and features for applications based on the Pactolus RapidFLEX Service Creation Environment and its SIPware Carrier Services Suite. Convedia's IP multimedia processing capabilities will be leveraged through the partnership.

The partnership initiative will both simplify media-rich feature deployment, as well as delivery of services across fixed and mobile networks. The IMS cross-network service architecture promotes adoption of advanced calling features, unified calling databases and customizations. These enhancements promote subscriber loyalty and extend the functionality of carriers' services. The integration of the two network services will enhance carrier support for integration of rich media applications like conferencing and collaboration, as well as entertainment applications.

"Convedia and Pactolus consistently rank among the most respected, innovative and broadly-deployed names in VoIP-enabled telecommunications," said Dave Horton, president, CTO and founder of Pactolus. "Our joint IMS development further enriches our ongoing relationship, and simplifies service provider IMS migration and expansion."

"Pactolus and Convedia were among the first companies to whole heartedly embrace SIP and the access agnostic, vertically layered approach to enhanced services that 3GPP's IP Multimedia System so well articulates," said Grant Henderson, co-founder and executive vice president of Convedia. "By embracing best-in-class products from companies like Pactolus and Convedia, fixed and mobile service providers are better able to realize the true promise of IMS."

http://www.pactolus.com
http://www.convedia.com

Mercom Releases Version 2.0 of Mercom Interaction Quality

Mercom Systems, Inc. (news - alert) announced the release of version 2.0 of Mercom Interaction Quality (MIQ). Mercom's MIQ software is a powerful browser-based call evaluation and quality monitoring system, which advances the correlation of quality and productivity metrics in the contact center.

New features in MIQ 2.0 include:

• Quality Key Performance Indicator Report — shows correlations between quality scores for behaviors and an outside metric the user would like to impact (e.g., revenue, customer complaints, etc.). This directly translates into improved coaching and training as well as enhanced return on investment.

• Performance Dashboard — allows users to get a quick graphical overview of quality score trends and key quality indicators in their center. Organizations can customize their performance dashboard so they can easily view information pertinent to them.

• Error Analysis Reporting — allows users to view and analyze specific areas in which groups or individuals are making errors during calls

• Enhanced Calibration Function and Reporting — makes comparing multiple evaluators' scoring to a mean or standard faster and easier. Standard deviations can be tracked against goals in easy-to-read graphical views and drill-down functionality enables root cause discovery when deviation is present.

"The enhancements we've made to MIQ version 2.0 demonstrate our continued commitment to providing the best tools for our customers," said Avi Margolin, president and CEO of Mercom. "MIQ has always been a strong, popular product for us, and by adding these new intuitive features for performance management, we feel it will bring more value to the marketplace and that our customers will be very pleased with the results."

http://www.mercom.com

Aspect Software Announces New Version of Aspect EnsemblePro 6.0

Aspect Software, (news - alert) the world's largest company solely focused on the contact center, announced the general availability of Aspect EnsemblePro 6.0. This latest release from Aspect Software's Unified Product Line offers substantial new features and functionality, including hosted services capabilities, enterprise quality monitoring, increased security functionality and expanded enterprise administration. The new product capabilities allow companies to more easily achieve their customer service, sales and telemarketing and collections business objectives.

Aspect EnsemblePro is a complete contact center solution that unites inbound, outbound, and blended multi-channel contact, while delivering robust queuing, routing, reporting and agent empowerment capabilities.

The product also provides, for the first time, application service provider (ASP) capabilities. The enhanced architecture enables service providers to host multiple clients segmented as individual tenants to prevent them from accessing or exhausting each other's resources while leveraging a common platform.

"The newest version of Aspect EnsemblePro not only enters Aspect Software into the hosted services solution arena for the first time, it also provides enterprise users with an even greater level of flexibility and security in solving their diverse contact center challenges," said Steve Herlocher, vice president of product management at Aspect Software. "Aspect EnsemblePro bridges the gap between inbound, outbound, self-service, and quality monitoring functionality, enabling organizations to easily provide blended contact center services. In addition to offering three times the scalability of previous versions, it offers the most comprehensive features and capabilities available today from any unified product."

http://www.aspect.com

TI helps Net Telephony Providers Manage Devices

Texas Instruments (quote - news - alert) has developed software that aims to help Internet telephony operators better manage devices sitting in their customers' homes and offices.

The company unveiled Piqua, a software that sits in voice over Internet Protocol (VoIP) gateways, such as those sold by Linksys or D-Link, and VoIP phones. The software can detect and automatically make adjustments to ensure better calling quality. For example, if there is an echo on the call, the software can detect it and automatically fix the problem. Problems can now be fixed even before customers notice there is an issue, TI said.

The Piqua software also can help technicians in the VoIP provider's call center diagnose problems more efficiently and effectively by giving them more visibility into the device, the company said.

Troubleshooting problems with a technician on the phone can turn into marathon ordeals where the problem is still unresolved at the end of the day. And all too often, customers blame the VoIP devices for their troubles, which they often send back to the service provider. But most times, the device is fine — it just wasn't configured properly, said Lindsay Schroth, a senior analyst at The Yankee Group.

"In an IP service, like VoIP, the intelligence is in the phone and not at the central office, like it is in the traditional phone network," said William Simmelink, general manager of the packet voice and video business unit for TI. "So it's important for technicians in a call center to be able to control the devices sitting at the customer site."

http://www.ti.com

KANA Response Selected for Integration into Gem Contact Center Services

By Susan J. Campbell

KANA Software (news - alert) has been selected to partner with Gem in its contact center services. Gem, a Northern-Ireland based provider of multi-channel, multi-lingual contact center services to global organizations, is implementing KANA Response from KANA Software, a provider of Service Resolution Management (SRM) solutions.

KANA Response is an email response management application that will assist gem in handling its ever increasing volumes of email, up to 500,000 per day from 22 countries. Since the implementation of KANA's technology, response capabilities have increased by 40 percent.

Gavin McGoldrick, Technology Director at gem noted that the company bombarded KANA Response with emails during the testing phase and it quickly became apparent that the system could easily manage any seasonal volume spikes, such as the Christmas period. With the level of performance and extensive response automation, McGoldrick felt that KANA Response delivered the wow factor that gem was looking for.

Through KANA Response's extensive automation functionality, agents are free to handle more complicated calls into the contact center instead of referring these calls directly to clients.

According to Marchai Bruchey, SVP of Marketing and Alliances at KANA, the KANA Response enables gem to tailor its operations to meet the best practices of each client it supports.

http://www.kana.com

Verizon Completes $8.5 Billion Merger with MCI

By Robert Liu

Verizon Communications (quote - news -alert) and MCI (news -alert) announced they have finally closed their $8.5 billion merger, creating a regional Bell operating company with a national footprint and approximately 250,000 employees serving customers in 150 countries.

As part of the integration of MCI, Verizon said it has created a business unit encompassing business and government customers and related functions of the former MCI. Michael Capellas, former president and CEO of MCI, also announced that he is leaving the business, now that the merger has been completed.

"This milestone for Verizon creates a new competitive force with the power of the global MCI network and the reach of Verizon's broadband and wireless networks in the U.S.," said Verizon Chairman and CEO Ivan Seidenberg. "The combination of our world-class wireless and broadband access networks with the leading global IP (Internet protocol) backbone will allow us to deliver the highest quality end-to-end experience for our customers."

As a result of the merger, Verizon will now operate three network-based businesses: the newly created Verizon Business, Verizon Wireless, and Verizon's landline segment, which is deploying wireline broadband and video networks. Verizon Business will also include part of Domestic Telecom, including the former Verizon Enterprise Solutions Group. Verizon Business will target medium and large businesses and government customers. New products and services are expected later this month.

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

8x8, Brightpoint Ink Deal

By Johanne Torres

VoIP and videophone service provider 8x8 Inc. (news -alert) announced an agreement with Wireless Fulfillment Services LLC, a subsidiary of mobile phone distributor Brightpoint Inc. (news -alert) in order to distribute Packet8 Internet phone services and related equipment into its Value Added Reseller (VAR) and System Integrator (SI) channel.

Through its online Wireless Enterprise Solutions (WES) program, Brightpoint's nationwide network of VARs and System Integrators will serve as subagents for Packet8 residential and business service plans. Brightpoint will warehouse, activate, provision, program and fulfill Packet8 devices, eliminating the need for subagents to maintain device inventory while at the same time allowing them to receive commissions for the activation of Packet8 VoIP-based services. Additionally, the WES Web site will provide Brightpoint's customers with access to its catalog of wireless accessories, mobile applications and content to provide customers with a bundled wireless solution.

"By focusing on the wireless and mobility needs of the enterprise market, Brightpoint is creating a valuable opportunity for its dealers to augment their business with targeted solutions like Packet8 Virtual Office," said 8x8's vice president of sales & marketing Huw Rees. "We are very pleased to partner with Brightpoint in this effort to introduce small businesses to a phone solution that will dramatically cut their telecommunications costs while enhancing productivity and growth."

http://www.8x8.com
http://www.brightpoint.com

Two Leading Infrastructure VARs To Resell Visual Networks' New Select Bandwidth Manager for Citrix Presentation Server

Visual Networks, Inc., (news - alert) a leading provider of network and application performance management solutions, today announced that it has entered into reseller agreements with two leading Citrix resellers. Convergence Technology Consulting and IntelliSuite Technologies, Inc. will resell Visual Networks' new software solution, Select Bandwidth Manager for Citrix Presentation Server, which provides in-depth, cost-effective visibility at local and remote sites into published applications under Citrix's standard ICA protocol.

"Our Citrix clients want to know what information is traversing the network, particularly given that enterprises are increasingly relying on the transfer of data from one location to another," said Larry Letow, COO of Convergence Technology Consulting. "Select Bandwidth Manager for Citrix Presentation Server is an important tool in our arsenal, helping our clients successfully deliver data from one site to another. Its price point and robust functionality make this the best offering for us and our clients."

Select Bandwidth Manager for Citrix Presentation Server enables IT managers to identify bandwidth utilization on a per-user, per-application basis while also having end-to-end visibility into all Citrix-published applications and non-Citrix applications traversing the network — at both local and remote sites.

"We are very excited to have two experienced, highly regarded Citrix resellers in Convergence and IntelliSuite helping us take our new Select Bandwidth Manager for Citrix Presentation Server to market," said Mark Skurla, executive vice president of worldwide sales at Visual Networks.

http://www.visualnetworks.com

Tech Data U.S. to Distribute Latest Communications Solutions from Mitel

Tech Data Corporation (news - alert) announced it has established a distribution agreement with Mitel, (news - alert) a leading provider of IP communications solutions. Tech Data, through its Telephony Specialized Business Unit (SBU), is the only U.S. broadline distributor to offer Mitel hardware, software and accessories.

"Mitel is a significant addition to Tech Data's Telephony SBU product offering, which already includes a broad array of converged voice, video and data communications solutions from industry-leading manufacturers," said John Ruel, Tech Data's director, Networking Product Marketing. "The Mitel agreement provides our customers even greater flexibility when developing state-of-the-art communications solutions that offer end users compelling cost savings, improved operational efficiencies and enhanced workforce productivity compared to legacy analog systems."

Tech Data will distribute Mitel's broad range of IP communications platforms that support from 10 to as many as 65,000 users in a single network configuration; applications and services gateways that provide connectivity to Microsoft's Live Communications Server; and numerous desktop devices. The Telephony SBU also will offer resellers innovative Mitel solutions for call centers, mobility, speech-enabled unified communications, messaging, video conferencing and wireless communications for deployment in a wide array of vertical markets including hospitality, healthcare, retail and government.

http://www.techdata.com
http://www.mitel.com

Metaphor Solutions and XO Interactive Deliver Automated Phone-Based Customer Service Solutions

Metaphor Solutions (news -alert) and XO Interactive (XOI) (news -alert) announced a reseller agreement for XOI to deploy Metaphor's speech IVR applications on XOI's hosting infrastructure. This partnership enables XOI to offer all 30 packaged and configurable Metaphor applications and use its life-cycle management tools as part of XOI's Teleservices offering.

Metaphor's Support Entitlement application has now been deployed at the leading provider of network security and availability software with four more ordered in the current customer agreement. This joint customer now enjoys deflection of calls from human agents to the automated speech IVR system while providing callers with a high quality of service without putting them on hold.

"Metaphor provides us with a competitive advantage in offering cost-effective speech IVR solutions that can be quickly implemented for our customers. Packaged applications designed for and targeted at specific vertical industries is clearly what our customers require today," said Andrea Jadwin, Director of Sales and Business Development at XO's Interactive Division.

"With its highly reliable, flexible and scalable hosting infrastructure, XO Interactive provides a world class platform for our packaged speech applications. And as one of the largest providers of hosted speech IVR solutions, XOI will be contributing to our mission of enabling adoption of speech applications at every enterprise," said Michael Kuperstein, CEO of Metaphor Solutions, Inc.

http://www.metaphorsol.com
http://www.xo.com

Avaya, Tallard Connect Florida Hispanic Business Market

By Johanne Torres

Business communications provider Avaya (news -alert) and Tallard Technologies, (news -alert) a communications systems reseller that provides solutions to companies in Latin America, the Caribbean, and Florida, have joined forces to bring Avaya's communications systems to Hispanic small and medium businesses in Florida.

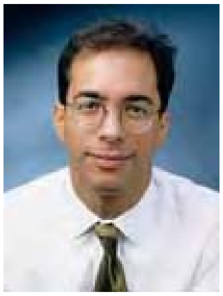
Through the new partnership, Tallard will distribute a suite of Avaya systems for small and medium companies. The suite comprises Avaya IP Office, a converged voice and data system for small and medium businesses; the PARTNER Advanced Communications System, a telecommunications system that supports advanced telephony capabilities, including messaging and wireless applications; and PARTNER Small Office Edition, a bundle that connects up to eight extensions and includes the PARTNER system, four analog phones and a voice mail package priced at $1,199. Tallard is already an Avaya distributor in the Caribbean and Latin America region.

"There are more than 55,000 small and medium Hispanic-owned businesses in Florida, and most of them are in the Southern part of the state," said Manny Rodriguez, director of convergence, Tallard. "Every company, no matter what size they are or industry they're in, wants to compete and flourish, so resellers must meet the needs of a diverse market. Many companies want the latest applications made possible by IP telephony, and others, the small retail stores, delis, pizza shops, dentists, doctors and small distributors, are looking for systems like the Partner bundle for basic voice telephony functions and value for their money."

"We are an application service provider, but it was clear to us that our customers also needed the systems that could support sophisticated applications," noted Jose Matto, vice president, business development, ANEW Broadband Communication.

http://www.avaya.com
http://www.tallard.com

Mind Share 2.0



By Marc Robins

CES Skinny: Seeing Scads of Skype-Certified Solutions

Anyone who has been reading my columns lately knows I'm all agog over the latest trends surrounding Web Telephony. After so many years and so much hype, the marketplace is finally catching up with the grand vision of easy-to-use peer-to-peer VoIP services. I, for one, couldn't be happier.

You'd have to be hiding under a rock these days not to notice all the announcements of new phones, headsets, and WiFi routers — most of them Skype-certified and designed to make it easier and more enjoyable to make and receive peer-to-peer VoIP (define - news - alert) calls.

To date, there are about 100 products that have been Skype-certified, and it seems that new ones are being announced on a weekly basis, which is not that surprising given the rapidly expanding size of the Skype marketplace (231,731,694 downloads and counting). Many of these products can be found at Radio Shack and online at Skype's online store at http//www.skype.com/store.

The latest and more drool-inducing product of this ilk — Netgear's (news - alert) Skype WiFi Phone — was announced just last month at the Consumer Electronics Show. As we all know, Skype is great for making free phone calls anywhere around the world, but the downside has been that you needed to be chained to your PC in order to use the service. Netgear's soon to be released Skype WiFi phone is sort of a "Holy Grail" for Web telephony users in that it offers unparalleled FREEDOM for the user. With this WiFi phone, all you need is to get in range of a WiFi network, turn on the phone, and log into your Skype account. No PC needed! The phone ships with Skype software pre-installed. Pricing hasn't been announced yet, but if it's along the lines of other relatively inexpensive cordless/wireless Skype setups (i.e., ~$100), I expect sales to be brisk. In conjunction with this new phone, Netgear is also offering their WPN824 wireless router — supposedly designed to optimize throughput and reduce interference for problem-free Skype calling.

Several other handsets, headsets, and other products join the flurry of Skype-related releases. A new USB handset from IPEVO (news - alert) – the Free-1 — recently won a design award and is designed for both Macs and Windows PCs. During the holiday season, IPEVO was offering a special discount off the suggested retail price of $30 for the Free-1 — go to http//www.ipevo.com to see if it's been extended.

Another new product — Icemat's (news - alert) Siberia Multi Headset — retails for $79 and comes with exceptionally comfortable large ear-cups and a unidirectional microphone that can be attached to anything, and volume can be adjusted with a separate control for convenience. The Siberia Multi Headset also includes a mini jack and converter for professional audio equipment.

Venerable Sennheiser Communications (news - alert) — the granddaddy of professional-grade headsets — offers nine Skype certified headsets, including the Professional PC 120 headset. Retailing for $30, the PC 120 is an ultra-light, one-sided headset with an integrated volume control and microphone mute switch.

Philips Electronics (news - alert) introduced a new cordless Internet phone, the VoIP321. The new phone has dual functionality so consumers can make free Skype calls, as well as ordinary landline calls. The Internet phone operates on Digitally Enhanced Cordless Telephone (DECT) technology, the cordless standard typically used in Europe.

Aliph's (news - alert) new Jawbone PC Edition VoIP communications headset integrates Aliph's military-grade audio processing and product design to help eliminate background noise. The device has a metal outer edge that protects the internal electronics, while the inside surface is made of medical-grade plastic to provide a comfortable, soft, and smooth feel on the skin. The new Jawbone is PC-compatible and uses a USB 1.1- or 2.0-enabled host device and co-located standard three-conductor, 3.5mm mini-microphone and headphone plug ports, and is available for limited time for $79.95.

Other notable Skype-certified products that have been available for a while include:

The marketplace is finally catching up with the grand vision of easy-to-use peer-to-peer VoIP.

Linksys (news - alert) CIT200 Internet Telephony Kit — This is a cordless phone that plugs into your PC's USB port and allows you to take Skype calls from anyplace in your home or office. It's bundled with a voucher for 60 free SkypeOut minutes, and has a suggested retail price of $129.

Motorola (quote - news - alert) Wireless Internet Calling Kit — This is a wireless headset and PC adapter bundle, which allows you to make Skype calls while wirelessly connected to your PC. The same headset will also work with your Bluetooth cell phone. The headset is bundled with a voucher for 30 free SkypeOut minutes, and has a suggested retail price of $99.

VoIPvoice (news - alert) Cyberphone-K — This is a USB

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handset that plugs into your computer's USB port and indicates the phone's connection and off-hook condition. It also features an invisible hook switch with proximity sensing for an intuitive start and end to calls. A PC sound card is not required, and calls can be made while listening to music on the computer. The Cyberphone-K has a suggested retail price of $50.

RTX (news - alert) DUALPhone — This is a cordless phone that plugs into your computer and landline at the same time, which allows you to make and take Skype and landline calls from the same phone. It has a suggested retail price of $139.

To date, there are about 100 products that have been Skype-certified.

Logitech (news - alert) QuickCam Fusion and QuickCam for Notebooks — Designed for desktop or notebook computers, these 1.3 megapixel Web cameras work with Skype and offer a free directory of characters and avatars that track your face and follow your movements. These Webcams have a suggested retail price of $99.

Skype (news - alert) Starter Pack — Available at the Skype online store and at RadioShack for $9.99, the Skype Starter Pack includes a Skype-enabled headset with microphone and 30 SkypeOut minutes to call any number anywhere in the world.

With all the above and more on the way, consumers now have a wealth of cool devices to make Web telephony services like Skype not just smart because of the low-cost factor, but also far more appealing and attractive than the plain vanilla POTS service they supplant. As if the incumbent telcos didn't have enough reasons for a migraine. IT

Marc is Chief Evangelism Officer of RCG (Robins Consulting Group), a marketing intelligence and communications company dedicated to the needs of the IP communications industry. Marc has been involved in the telecommunications industry as a reporter and analyst, trade show producer and publisher, and marketing executive and consultant for more than 25 years. Contact RCG at 718-548-7245 or e-mail info@robinsconsult.com for more information.

By Tony Rybczynski

A Unified Security Framework for Regulatory Compliance

Security breaches — and the resultant loss of productivity and access to confidential data — costs enterprises millions, but the security imperative goes beyond these financial incentives.

When effects of security breaches extend beyond individual enterprises to entire industries or further, the government is forced to step in: The Sarbanes Oxley Act; the Patriot Act; industry-specific regulations, such as the Gramm-Leach-Bliley Act for the financial industry and the Health Insurance Portability and Accountability Act (HIPAA); and security-related regulations outside of the U.S., like the Data Protection Directive in the EU.

Indeed, each of these government directives is based on historical events and precendent, and they are designed to benefit the communities to which they apply. Nonetheless, each places an additional burden on the enterprises that must abide by them. They have wide-ranging impact on security, including requirements for encryption, disaster recovery and business continuity, archiving, and consumer privacy. Failure to comply with these regulations can bring civil and criminal penalties.

Policy, Processes, and People

A properly designed and implemented security policy must clearly identify the resources in the enterprise that are at risk, as well as any resulting threat mitigation methodologies, whether procedural or electronic. People and processes refer to the enforcement of security policies that address not only technical considerations, but also the business and human aspects of security. The objectives are clear: keep the bad guys out, make it easier for the good guys, and provide clear situational awareness for proactive defense.

Baseline Security Establishes a Sound Foundation

Individual network and user devices that provide (switches, routers, and communication servers) or use (desktop or mobile devices) network services all need to be secured, at least to a basic level. For example, the integrity of the device operating software and configuration data needs to be protected. End devices should support personal firewalls and anti-virus software, while administrator authentication and authorization and securing remote network operations are critical for network devices.

The objectives are clear: Keep the bad guys out, make it easier for the good guys, and provide clear situational awareness for proactive defense.

Layered Defense Approach to Network-Level Security

A layered defense approach to security is designed to ensure there are no single points of security failure in a network. This is accomplished by using multiple approaches to security enforcement in different parts of the network. Layered defense approach is also bolstered by leveraging systems that utilize security capabilities and products from best-of-breed security vendors. Based on open, standards-based technologies, this approach enables easy integration and simplified operations that reduce the overall network security total cost of ownership. Intrinsic to layered defense is closed loop policy management, which includes configuration management of network devices, enforcement of policies in the network, and verification of network functionality via audit trails. These functions are implemented in a closed feedback loop to ensure that polices are continually refined for maximum security.

As employees, business partners, and customers make more use of the enterprise network to meet their business objectives, enterprises need more control of the endpoints that are used to access the network. The goal of endpoint security is to ensure valid user identity, and device security policy compliance (e.g., most recent anti-virus software). Because so many threats are now coming from internal network users, this must necessarily include wired and wireless endpoints within the network, as well as those at remote sites, where there is less control over users' devices.

Perimeter security, the second key element of layered defense, can be applied at internal perimeters, at the external edge of the network (the DMZ), around data centers, around secure multimedia zones to protect multimedia and IP Telephony call servers, and even around a single critical user. Perimeter security ensures effective and efficient secure network zone boundaries, enabling businesses to ensure information assets are protected without losing business agility. The tools of the trade include sophisticated, state-aware, packet filtering and application firewalls, which perform deep packet inspection to detect and block attacks that directly target applications. They also include VPN routers and gateways, which provide firewalls

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as well as SSL and IPSec VPN support for remote offices and users. First attack protection systems provide perimeter security protection for network killer attacks (e.g., high volume DoS, virus, and worm attacks), in addition to application delivery capabilities, such as load balancing and bandwidth management.

Keeping watch for malicious software and traffic anomalies, enforcing network policy, and enabling survivability is the role of core network security in a layered defense approach to security. It also is a key function in enabling situational awareness and evolving a security architecture to increased autonomic operation. Continually monitoring the network for malicious activity is key to ensuring that if an attack slips through other layers of security, a network will detect it and take appropriate action to block the attack and ensure survivability.

Application Security: Protecting Information in Transit

Protecting corporate and customer information from unauthorized discovery, eavesdropping, or misappropriation while stored or in transit across networks is an important element of the layered defense approach to security. While IPSec provides cryptographic protection at the network layer (OSI Layer 3), Web traffic uses Secure Sockets Layer (SSL) to secure communications at the transport layer (Layer 4) and offers the added benefit of not requiring the support of client software. Given that unified networks can carry voice, data, and video, when and where to protect this traffic is a major consideration. Real-time traffic can be protected using these or can leverage TLS (Transport Layer Security) and SRTP (Secure Real Time Protocol) to encrypt the signaling traffic to the network, and voice or video traffic on an end-to-end basis, respectively.

It all boils down to finding the true value of a unified security framework. A unified security framework allows enterprises to develop and enforce risk-optimized security policies across increasingly converged environments, and address process and technical considerations as well as regulatory mandates to protect data integrity and confidentiality. IT

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

By William B. Wilhelm, Jr., Esq. and Paul O. Gagnier, Esq.

A New Year's Resolution

2005 saw a flurry of regulatory activity on VoIP services in a number of countries. In many cases, however, regulators failed to account for the fundamental differences between VoIP and traditional telephone service. Too often the result was bad rules and bad public policy.

So what is in store for 2006? More of the same — unless the VoIP (define - news - alert) industry grabs the reins and starts to drive the regulatory debate. Our collective New Year's resolution should be to take control of the direction of regulation of VoIP.

As consumer uptake of VoIP increases, VoIP providers can be sure that regulators will take action to ensure that consumer interests are protected. Regulators also have made clear their intent to ensure that certain social goals that have long been met primarily by traditional telephone carriers, such as universal service, access to emergency services, law enforcement, and national security access to communications, and access by persons with disabilities, continue to be available in a VoIP world.

Most of us recognize the importance of these objectives. At the same time, the VoIP industry has been largely passive in the debate over how VoIP will affect them. To a large extent the industry has allowed those with other agendas to set the direction and tone of the discussion. Instead of showing how VoIP can advance these goals — and even improve their execution — the industry has reacted to proposals put forth by others. This must change in 2006.

The common theme of the many regulatory decisions issued in 2005 was the recognition that the growth of VoIP presents a challenge to the assumptions that underlie much of communications regulation. Rather than try to develop a new paradigm, however, most regulators chose to shoehorn VoIP into existing regulatory structures without proper recognition of the important differences in the technology.

• In the U.S., the FCC's E911 and CALEA orders imposed onerous and, in some cases, unreasonable obligations. In its E911 order, the FCC promulgated rules that ignored the unique practical and technological challenges faced by VoIP providers in providing access to emergency services. The result has been widespread acknowledgement that the 120-day compliance deadline is unachievable. For example, one RBOC recently reported a 30 percent compliance rate while another reported 67 percent compliance several months after the deadline had passed. Likewise, the FCC's CALEA order disregarded important differences between broadband, VoIP, and traditional telephony. While both of these Orders are being challenged in court, these challenges and the uncertainty that accompanies them generally weigh more heavily upon innovators than it does incumbent operators.

• Canada's emergency access requirements for VoIP also went into effect. The CRTC also imposed extensive obligations on VoIP providers and a short timeframe for compliance. However, unlike the FCC, Canada's regulator recognized the technical difficulties associated with providing 911 service for "nomadic" VoIP and allowed providers the flexibility to adopt interim solutions for providing access to emergency services.

• In the EU, the story was largely one of inaction. The European Commission's stalled VoIP consultation continues to languish. As a result, the EU, despite its forward-looking communications legislation, is on the sidelines of the VoIP debate. This has required VoIP providers to navigate 15 sets of national regulations (25 if you count the EU's new members) with widely differing approaches to VoIP.

Certainly there were positive developments in 2005. Regulators in a number of countries issued decisions that recognized the differences between VoIP and traditional telephony and recognized the benefits that VoIP confers on consumers. For instance, Australia, Hong Kong, Malaysia, and Singapore issued new regulatory frameworks for VoIP that regulate VoIP services with a "light touch" and exempt many VoIP services from the full panoply of obligations imposed on legacy networks. Another important ruling was Finland's December 2005 decision with respect to Skype. In its decision, the Finnish regulator undertook a detailed analysis of Skype's VoIP services that recognized the substantial differences between those services and traditional telephony. More important, the decision took pains to minimize the regulation of Skype's services while also ensuring that Finland's social goals were met.

Our collective New Year's resolution should be to take control of the direction of regulation of VoIP.

Looking to 2006, it is clear is that regulators are going to continue to look at VoIP and the regulatory obligations that

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should be placed on it. Pending proceedings in the U.S. and a number of European countries will examine various regulatory issues related to VoIP.

What is less clear is what the VoIP industry is going to do about these proceedings. For too long, many in the VoIP industry have laid low, apparently in the hope of being overlooked by regulators who were focused on bigger issues. This is no longer a viable approach (if, indeed, it ever was). What is undeniable is that the popularity of VoIP has attracted regulatory attention and the industry must respond.

So what is required? To be more proactive — with consumers and consumer groups, business, law enforcement agencies, and regulators — about the ability of VoIP to promote universal service, provide access to emergency services, to allow wiretapping, and to serve the disabled. We need to demonstrate how our technology can in many cases meet these goals even more effectively than traditional telephony. Finally, we need to demonstrate why cramming VoIP into ill-fitting legacy regulation is likely to undermine, rather than further, the goals that regulators are trying to advance.

The industry has allowed those with other agendas to set the direction and tone of the discussion.

The regulatory genie is out of the bottle. Our choice is to continue to allow others to frame the debate or to take the lead in defining the future of VoIP. The outcome is up to you. IT

William B. Wilhelm and Paul O. Gagnier are Partners at the law firm of Swidler Berlin LLP. For more information, please visit http://www.swidlaw.com. (news - alert) The preceding article is not to be considered to legal advice and it represents solely the personal views of the authors.

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By Hunter Newby

CableLabs' VoIP Peering RFI

CableLabs has taken one small step for VoIP and one giant leap for VoIP peering by issuing their VoIP (define - news - alert) Peering RFI. The action suggests how serious the MSOs are about providing on-Net voice communication between their networks. It also lends credibility to the concept.

For now, this represents merely a request for information and not a request with a defined path to any action on the part of the cable companies. CableLabs will be providing to its constituents essentially a concise picture of the VoIP peering landscape as it exists as of this moment. The decisions of whether or not to peer with the other MSOs and how that may be accomplished will be left to the MSOs themselves.

Still, it is decidedly a collective and open acknowledgment of the existence of this new interconnection method and a new breed of service provider breed within. How this ultimately shakes out will be left to the respondents and their explanations, and then the determination of the group and its individual members. But regardless of the untlimate outcome, it has raised the awareness of VoIP peering, and that is the key element to furthering the cause.

So much of what has been done to date with VoIP peering has been cast in the shadows of semantics, with many claiming to be in the space. Taking away all of the "new service offering" announcements that were merely publicity with no real customer implementations, there has actually been considerable progress made by a few who have services that perform real, valuable functions in the interconnection of disparate VoIP networks. Some providers, under the VoIP peering umbrella, have carved out niches in just protocol conversion from TDM to SIP; others even help resolve SIP-SIP interoperability issues. Mainly, though, the benefit to MSOs is that it provides information about past mistakes and how to avoid them.

The bottom line is that there are ways to connect VoIP networks so that calls can originate as IP on a private network and terminate on another private network as IP, such that they never have to touch the legacy voice network. This is real and it is happening. There are many parts of the equation still unsolved, such as what codecs to use or which version of SIP to employ, but the market will help rsolve those issues and, over time, a commonly understood and agreed upon standard will arise. It will likely be modified over time, but it will exist and this RFI will help set things in motion.

CableLabs will be providing to its constituents essentially a concise picture of the VoIP Peering landscape as it exists as of this moment.

One of the biggest battles brewing in the VoIP peering world realtes to the numbering plans and, more importantly, who gets to run, control, administer, and manage the number database. Even though a number database may not seem like a traditional telecom "service," that data — and being the master of its domain — is seen as a position of power. Interestingly, this is a struggle that is more political than it is tehnical.

A year ago, you could still find some people who argued that ENUM does not work and won't work. All the while, the ITU recommended to its members that they each gain control over the ENUM administration within their respective countries or risk losing control of the phone system. It would seem the industry is past that stage now, and it is equally obvious that ENUM is on the minds of interested participants, as evidenced by repeated references in the CableLabs RFI. Again, the RFI is not an indication of a chosen path, but the well thought out questions were very heavily geared towards ENUM. Let us not forget SIP either, but notice the conspicuous absence of H.323.

Now that the road map of "right questions" has been prepared by CableLabs, others can take advantage of this free step-by-step guide. The process helps map out the steps for any vertical to create a private, all-IP environment for voice. As it relates to ENUM, this is what I refer to as "Private ENUM," or, taken a step further, within a specific group, "Enterprise ENUM." this is the essence of Enterprise Peering — the private VoIP WANs of multiple enterprises each connecting to a common platform with a set of standards.

As the RFI states, "The main advantage for IP network providers to peer is to reduce the amount of traffic that transits between the two networks via third-party backbone providers. Such an arrangement has the effect of reducing network costs and increasing quality of service." Enterprise IP networks can take advantage of this as well, and those that are equipped with the right information and capabilities have a great incentive to do so. IT

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

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SuperShuttle®
Inter-Tel® Customer for Over 10 Years

"We're in 18 major airports.
Our communication with each one of those locations is vital to handling our customers properly.

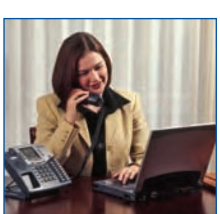
(Inter-Tel) provided me something that I couldn't do in any other system. They became part of our infrastructure. Even as a third party, they were always innovative. And when I went to Inter-Tel last year as my primary vendor, they gave me a great deal of confidence in their capabilities. Their NetSolutions® services, along with their phone systems, gave me the single point of accountability I was looking for to help us manage our business.

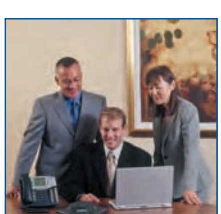
I have very little downtime with any of these systems which is extremely important to our operations since we're 24 hours a day/365 days a year. I can't afford downtime.

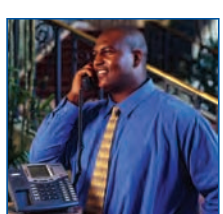
It's easy for me to compliment Inter-Tel: One, for their reliability, both in their products and their network. And, also, for their customer service – their employees are very, very customer service oriented. Their team being able to say "yes" to me whenever I have a crisis is extremely important. I find that refreshing."

Linda Paquin
V.P. of Reservations & Telecommunications
SuperShuttle Transportation Systems

To learn more about Inter-Tel's value-driven communications systems and solutions, visit www.inter-tel.com







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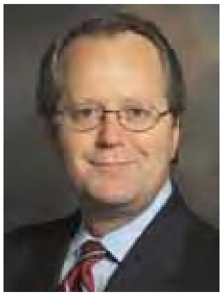

By Richard N. McLeod

VoIP Helps Branch Offices Get The Respect They Deserve

I tell ya… Branch offices can be like the Rodney Dangerfields of the corporate world. They just don't get enough respect.

Branch office workers often feel exiled, cut off from the business tools and the easy flow of communication enjoyed by employees at central locations. Branch office requirements are often overshadowed by the focus on headquarters' needs, even though branch offices often drive the business for many industries, such as banking, real estate, and retail. Without local branches, these industries simply could not sell products or deliver services.

It does not have to be this way. With IP communications, applications can be extended to all employees, be they at headquarters, a regional office, or a branch ofice. Data, voice, and video can flow easily between all locations, bossting productivity, efficiency, and customer service. Perhaps as importantly, instant, unfettered communication and common applications can integrate branch offices with headquarters and with one another, allowing an entire organization to operate as a seamless whole.

Integrating the Branch

Today, many organizations function with two incompatible networks: a voice network and a data network. While never efficient or cost-effective, this dual network structure is especially cumbersome and wasteful for branch offices, where approximately 40 percent of all employees work.

Remote office workers are disadvantaged in a wide variety of ways because of non-integrated voice and data networks. For instance, they can't easily access the company's central voice mail system. They can't take advantage of critical applications and tools available to other employees. They're cut off from the daily flow of knowledge and communications. In some cases, working in a branch office is almost like working for another company entirely.

This problematic situation certainly does not help the organization either. Costs for operating, administering, and maintaining separate networks for the branches can be high, as can long distance telephone fees. Moreover, companies pay a substantial price in lost productivity and dissatisfied customers when calls cannot be transferred and when information cannot be shared transparently between locations. The threat of isolation also is very real. Branch offices can become cut off from the larger organization and begin to regard headquarters — or other branch offices — as peripheral or even as an adversary. That, of course, is dangerous for any organization, especially for those that depends on remote workers for critical functions, like sales and customer support.

Branch offices are the Rodney Dangerfields of the corporate world.

IP communications goes a long way toward solving these challenges. It integrates voice and data into a single network, allowing branch office staff to readily access the same applications and data as central office workers. Productivity and customer service then can increase significantly, because branch employees are able to respond immediately to customer inquires and offer new services to them. In addition, the company's total networking costs decrease, while availability, resilience, and security all improve.

Perhaps most importantly, remote offices are no longer quite so remote. Convergence means that all employees can work as if located in the same office. Branch workers are privileged to the same real-time communications — data, voice, and video — as other staff and are able to participate as full team members in projects that require document management, workflow, and even face-to-face meetings, which can be held through videoconferences. Certainly, these improvements boost efficiency and overall quality, but they also imbue a sense of camaraderie and team spirit that is vital for any company to be fully competitive.

A Migration Path

Today, many organizations with a branch office system understand the substantial value inherent in a converged data and voice network. They may, however, balk at changing an existing system to which they are already fully accustomed. What many of them fail to realize is that most companies already have the foundation for upgrading to a fully converged network.

Nearly every organization has an existing data network upon which to build. In fact, adding voice communications is a logical next step in the natural evolution of the network. IT professionals also already understand the technology, since they know how to administer, secure, and support IP. Therefore, the migration can be done incrementally and affordably.

For instance, a company may begin its migration by installing IP-based voice applications at its headquarters location. These applications may include unified messaging, integrated voice mail and rich media collaboration, among others. Connections then can be made from headquarters to branch offices through existing circuitry. This way, branch employees are able to receive many of the benefits of an integrated network quickly.

Eventually, the organization will want to replace its headquarters PBX with a more comprehensive IP communications system, which will deliver the full benefits of convergence companywide. In the meantime, the company can make step-by-step improvements that will enable workers to become accustomed to the technology over time. The company's IT professionals also will have the opportunity to receive additional training, if necessary, and adjust to a new way of working.

IP communications allows remote offices to be firmly integrated into the enterprise.

Historically, branch office workers have been cut off from the daily pulse of the organization. They've often had difficulty communicating and collaborating with other offices, which might have diminished their productivity and the service quality they could provide customers. IP communications evens both sides of the equation, allows remote offices to be firmly integrated into the enterprise and the organization. As a result, these offices are no longer "out there," but can operate as if they're located on the very same floor. IT

Richard N. McLeod is director, IP Communications Solutions, Worldwide Channels at Cisco Systems, Inc. For more information, please visit the company online at http://www.cisco.com. (quote - news - alert)

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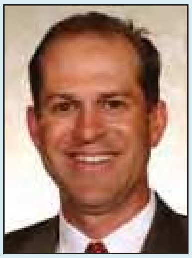

Covad's Jeff Ahlquist

I recently had a chance to interview Jeff Ahlquist, who serves as Vice President of Corporate Development and Strategy at Covad. As you no doubt know, the company has become a big player in VoIP and has done a good job branding itself these past years. I wanted to delve into the company more and learn how they are doing. I was wondering how the VoIP market is treating them.

Here is my very candid interview.

What is biggest barrier to your success?

Jeff thinks the biggest barrier is market education and the validation of VoIP (define - news - alert) as a technology ready for prime time. He believes demand generation is another barrier.

From here, our conversation turned to pure hosted VoIP providers. Covad doesn't want to be considered a pure hosted VoIP provider; they understand there is more to the market than just hosted solutions and not every company wants to go this route.

This is why the company has focused on a portfolio of options consisting of traditional hosted services as well as a product called PBXi that is similar to the offering from Cbeyond Communications, which also plugs into an existing PBX.

I asked about the competitive landscape, to which Jeff replied that they have their own network and are well positioned in the market.

What about 911 support?

Regarding to E-911 compliance, Covad (news - alert) has two types of customers: those who get their broadband service from Covad ("managed circuits") and those who get it from another provider ("unmanaged circuits"). Covad is at 100% compliance with its managed customers while not yet being quite there on the unmanaged side. Covad is working diligently on the issue and will be providing full 911 functionality to nearly 99.5% of its stations in Q1.

What is your pricing?

The price boils down to three parts for a hosted solution. First, there is access (for SDSL, T1, etc.). Second, there is the equipment, which us supplied by a dealer/VAR. Finally, we have the monthly service fee per seat/all your long distance and features, which ranges from $30 to $50, depending on the total number of seats.

Will prices in the hosting market go lower over time?

Jeff feels the price will compete well with PBXs. He tells me the up front costs are lower for their hosted solution. He also feels the price is lower on an ongoing basis. For example, he mentions you don't need a technician for moves or adds or deletes when you use a hosted provider.

What about the cost of service for truly large enterprises?

Ahlquist says they don't do many extremely large installs. Their sweet spot is 20-150 seats. (Jeff gets points for honesty, for rarely does anyone tell me they don't do something.)

How do you compete with on-premise equipment?

There are many interesting ways Covad attacks this issue. Covad can complement on-premise equipment with PBXi service or offer a hybrid of on-premise and hosted solutions.

Covad can also add phones that are hosted and even add a dashboard for voicemail.

How is your reseller market?

There are approximately 400 dealers selling Covad's service. These dealers can also sell the PBXi product with hardware.

What is the reason resellers would rather sell a service instead of a PBX?

It boils down to customer need, primarily. That said, there are compensation differences, as resellers get a recurring fee for service and also get to sell equipment along with the service. In the end, Covad takes on the brunt of operating costs of ongoing service.

Is pure hosted better than hybrid?

This depends on the deployment. There is room for anything, based on customer need. With a hosted solution, upgrades are easy and customers aren't locked in. You can't easily upgrade a PBX, Jeff points out. A PBX is a piece of hardware. Covad upgrades service all the time, providing a constant service refresh, which is cheaper and provides a considerably lower TCO.

How is your solution better than Centrex?

Centrex was static and used centralized equipment. It was not a software platform. With IP, you can easily integrate apps into service, according to Ahlquist. Telephony itself becomes an IP application. You take all the value of IP applications and apply it to communications.

Then Jeff pointed out the ease of distribution and consistency. With Centrex, you were limited to the LEC in your region. When you moved to a different area, the service changed. A solution like Covad's does not have these drawbacks and stays consistent internationally as well.

What about competition from other 'hosters'?

There is a lot of competition out there, but this validates the category, according to Jeff. It doesn't take a lot to become a hosted provider. He is concerned that some players may taint the industry by providing inferior service. "This can hurt the VoIP reputation," he concluded.

What is the feedback from customers?

It has been good. The customers have been direct and poignant. In addition, Covad has done some custom voice work, which companies love, says Jeff. Other customers are ecstatic that their VoIP service came back within 48 hours after the Florida hurricanes; the LECs were down for weeks.

Will hosting gain share? When?

It is, in fact, gaining share. Analysts are pushing out expectations a bit. Some businesses are waiting to see how it plays out but, as ILECs get into game, it will validate the space. Jeff feels 2006 will be good year, which will be one year behind analysts' predictions

Do enterprise customers need education on the hosted market?

Yes.

Who should be educators?

"You guys," meaning TMC, *INTERNET TELEPHONY*, and analysts. Market awareness must be built through advertising. The LECS won't push hard to validate VoIP, since they have a core TDM network. Jeff added that equipment providers also may not push the market in the hosted direction — but they do sell phones to hosted providers. He feels that Covad and others are the ones that need to drive market forward.

Are you concerned about a Skype-like hosted model?

This will cost money to produce, but if it is offered as a loss leader, yes, there is potential danger. Jeff went on to say that VCs won't give money to acquire customers at a loss. He thinks this model is easier to pull off in the consumer market than business.

Covad has been a major marketer in making businesses aware of VoIP and has been a great educator of the market. With a number of products and services in its portfolio, the company is in a position to be a supplier to all but the largest of enterprises. As a carrier providing voice and broadband, the company is in a great position to not only provide VoIP, but to provide it at guaranteed service levels.

In my view, the company's decision to sell a hosted solution as well as one that allows companies to work with a standard PBX is very smart. In the end, companies will go hosted when they are ready. Covad is well positioned to supply customers' VoIP needs, regardless of who or what they are. IT

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IMS and AdvancedTCA®: Building A New Foundation for Enhanced Services

"IMS is a strategic anchor point for next generation services and integrated service platforms. Standardization, integration and modularity are key to success."
Source: Yankee Group

A Brave New Network

As service providers around the world continue their migration to a converged network infrastructure, they are increasingly focused on offering a host of new enhanced services in addition to voice and data transport as the means to retain subscribers, attract new customers, and more importantly generate significant new revenue streams. Examples of such enhanced offerings relate to triple play and fixed mobile convergence, and include video and music, network-based gaming, multimedia ring back tones, and Enterprise applications such as presence management and unified messaging.

Uncharted Waters

Indeed, the relentless pricing pressure on standard voice and data services has created a critical scenario for today's service providers: either they introduce new innovative "must-have" services, or face gradual extinction. However, there's no certainty what the acceptance level of such new services will be, and there's no guarantee that what succeeds in one market will succeed in another. The challenge for today's service providers is to employ a dynamic network environment that allows them to develop and deploy promising new services quickly, add resources to successful services on demand, and easily reduce or remove resources for unsuccessful services.

The Promise of IMS

IMS, or IP Multimedia Subsystems, provides the answer to this challenge. Defined by the 3rd Generation Partnership Project, or 3GPP, IMS is an IP and SIP standards-based framework designed to allow the rapid deployment of new IP-based services across wireline, wireless and cable networks at low cost, and with minimal network interruption. The spotlight is on IMS today as providers look to IMS as a primary means to deliver fixed-mobile convergence and to accelerate new rich multimedia service deployments in a highly cost-effective manner.

IMS allows providers to deliver real-time IP-based multimedia communications; integrate real-time communications with non-real-time communications; enable multiple services and applications to interact; and escalate communications sessions easily, by turning an IM session into a voice call with a single click, for example. IMS also delivers significant improvements in mobility management, service quality, service control and standards interfaces for developers of new applications and services.

AdvancedTCA: The Ideal Platform Architecture for IMS

In order to deliver on the promise, the IMS framework requires modular, standards-based network elements designed with the specific considerations of communications networks in mind, including "five-nines" availability and superior performance. AdvancedTCA, for Advanced Telecom Computing Architecture, delivers the reusable, modular platform architecture that can provide the scalability, flexibility, and performance that IMS demands.

Indeed, AdvancedTCA-based solutions incorporating Intel® Architecture-based building blocks, including processors, and compute and packet processing blades, provide an ideal fit for IMS. AdvancedTCA addresses all key technical requirements for IMS network elements, including:

- High availability
- High compute density and SIP performance
- Rich media content and transcoding
- Video encoding and RTP acceleration
- SS7 and other PSTN signaling
- Bladed storage
- Large in-memory database support
- Control and user data separation with an intrinsic flexible GbE fabric interface architecture, and,
- Support for TLS and SSL-based security, and session border control functions.

Providing Key Tools for IMS Success

In addition to its leading-edge architectures in compute and packet processing silicon and industry-leading AdvancedTCA solutions, Intel has a comprehensive portfolio of educational activities, products and initiatives designed to accelerate time to market for IMS-based services. The Intel® Communications Alliance, with nearly 200 telecommunication vendor and solution provider members, works to ensure that products are successfully integrated into comprehensive IMS solutions.

For IMS system developers, the "Interoperability Guide for Modular Communications Platforms" is a design guide for standardizing hardware designs and enabling hardware reuse. Another useful guide is the "IMS Telco Server Proof Of Concept", where the best available products have been integrated, tested and validated to assure that they all work together as a fully functioning IMS.

Intel has also built a worldwide network of IMS solution competency centers in North America, Europe and Asia with the charter to assemble all the pieces of the IMS solution. These labs provide IMS solutions providers the opportunity to utilize a fully functional IMS core that can be used to integrate and validate interoperable solutions.

Delivering on the Promise of IMS

Combined with the proven time-to-market benefits of AdvancedTCA solutions utilizing Intel building blocks, IMS stands ready to deliver the enhanced services that today's and tomorrow's service providers require in order to succeed and prosper.

For more information about how a modular platform architecture based on AdvancedTCA and Intel building blocks can meet IMS requirements, contact your local authorized distributor, or visit us online at www.intel.com/go/ims.

Session Border Controllers

VoIP service providers, network operators, developers, and really anyone with a role in the VoIP industry, has come to understand that, in order to successfully deploy a VoIP solution, a session border controller (SBC) must be part of the equation.

In very basic terms, an SBC acts somewhat like a doorman at a high-end hotel, who allows hotel guests to enter, while keeping unwanted and uninvited visitors out. The SBC performs the same tasks at the edge of the network, allowing authorized traffic to pass and rejecting the rest. It is a session-aware device that usually sits at the edge (border) of a network controlling real-time traffic as it crosses from network to network.

But SBCs are capable of much more than interconnecting networks or simple security functions; there are a variety of deployment options, which is why, in this month's product round-up, we have attempted to describe the differences in products from the various players in the SBC market. What follows are brief overviews of SBC products from some 20 companies. Much more information can be gleaned by contacting the companies in person.

Acme Packet

http://www.acmepacket.com

Acme Packet (news - alert) Net-Net SD is a high-performance, high-capacity, signaling and media control engine that steers signaled media flows over the best route and performs required security, service assurance, and law enforcement functions at network borders. It supports service provider SIP, MGCP and H.323 networks; complements IP edge and border routers and offers security, SLA assurance, L2 & L3 QoS marking, revenue and profit assurance, law enforcement, high performance, high capacity. Net-Net SD is also available in a configuration that satisfies security and SLA assurance requirements at enterprise network borders.

Acme Packet Net-Net 9000, the next-generation session border control (SBC) platform, supports next-gen, converged fixed-mobile architectures being defined by 3GPP IMS, ETSI TISPAN, the Multi-Service Forum, PacketCable and the DSL Forum. It supports both subscriber access and network interconnect border requirements in wireline, cable and wireless networks. As a next-generation platform the Net-Net 9000 offers the industry's highest levels of performance, capacity and availability in a single SBC platform.

Acme Packet Net-Net PAC is the industry's most scalable, full-featured session border control solution that optimally delivers premium interactive communications across wireline, wireless and cable IP network borders. Designed for large-scale, Tier 1 service provider deployments. Net-Net PAC sets new levels for SIP performance, availability and capacity that are three times to sixteen times better than the nearest competing session border controllers.

Cisco

http://www.cisco.com

The Cisco (quote - news - alert)XR 12000 Packet Services Card (PSC-1) provides the session border control (SBC) functions on the Cisco XR 12000 Series Router product line, making it the first carrier router with integrated SBC. The SBC application builds on the secure virtualization, continuous system operation, and multiservice scale provided by the Cisco XR 12000 Series. With the seamless integration of SBC functions into Layer 2/Layer 3 services provided by the Cisco XR 12000, the Cisco XR 12000 SBC eliminates the need for overlay networks and standalone appliances.

The Cisco XR 12000 SBC provides an open and flexible architecture for all service provider deployments, whether for peering or for customer access. With its ability to handle unified and distributed signaling deployments, the Cisco XR 12000 SBC provides superior deployment flexibility to cable, wireline, and wireless service providers. An element of the Cisco Systems IP Multimedia Subsystem portfolio, the Cisco XR12000 SBC further accelerates network convergence while providing investment protection.

With its comprehensive suite of features and high-availability hardware and software, the Cisco PSC-1 helps enable a broad range of SBC applications on Cisco XR 12000 Series routers for cable, wireline, and wireless service providers. The SBC application incorporates the security, QoS, and secure virtualization capabilities of the Cisco XR 12000 Series to enable support for service provider-to-service provider interconnect and service provider-to-access interconnect.

Data Connection
http://www.dataconnection.com

Data Connection's (news - alert) SBC is a portable software solution designed especially for equipment vendors that is deployable immediately, delivering dramatic cost and time-to-market savings. The solution also includes many of the essential control functions required in an IMS environment, and the Session Border Controller architecture is designed for unparalleled modularity, flexibility, scale, and reliability.

DC-SBC is available to OEMs as a complete solution or can be modularized to incorporate individual components. Both solutions are highly portable and can be integrated seamlessly with any hardware and operating system as well as with existing, field-deployed systems including softswitches and routers.

Data Connection's SBC software can also be used to provide many of the essential control functions required in an IP Multimedia Subsystem (IMS).

DC-SBC is built on a foundation of proven and field-hardened VoIP signaling technology. These products have been shipping for the past five years, and have been field-hardened and deployed in a wide variety of VoIP devices.

Data Connection's SBC architecture is highly modular. The logical areas of functionality are broken down into separate subcomponents that do not share resources, and communicate using open, asynchronous, message-passing interfaces. This allows components to be distributed and even completely replaced or removed from the system.

Ditech Communications
http://www.ditechcom.com

(news - alert) PeerPoint's extensive feature set is aimed at making VoIP a reality in the service provider environment. PeerPoint C100's flexible modes and broad interoperability allow deployment in IP networks without requiring changes to the network or to the VoIP subscribers' equipment configuration. Deployed at the IP network edge, PeerPoint helps reduce the cost of delivering broadband VoIP services.

The most common problem service providers encounter when deploying hosted IP services is the inability of subscribers' voice equipment to work seamlessly behind NAT equipment and firewalls. PeerPoint ensures that subscribers can connect anywhere, anytime, without having to reconfigure their equipment.

PeerPoint C100 is the first SBC to solve the NAT issue in a Microsoft Live Communications Server (LCS) deployment. PeerPoint serves as an SBC interconnecting the LCS clients to the Microsoft Access Proxy, which, in turn, communicates with back end servers. PeerPoint enables service providers to deploy compelling integrated voice and video applications in parallel with secure instant messaging and presence.

PeerPoint also helps service providers reduce costs and reach new markets by enabling seamless peering with other VoIP service providers. PeerPoint manages both media and signaling, which allows it to collect forensic data, such as QoS metrics, remote device information, and voice quality information to help isolate telephony issues or enforce Service Level Agreements (SLAs) with peering partners.

Emergent Networks
http://www.emergent-netsolutions.com

(news - alert) When signaling control is not enough, the ENTICE Session Controller allows full access to all the communications flows (signaling, media, and media control) required to establish a media session in a packet network. This solution is designed to bridge the gap between two networks, offering protocol translation between H.323 and SIP, media access and control to solve security and vendor compatibility issues as well as providing call detail records for billing.

Because of the E-SC's unique architecture, custom solutions can be easily created by modifying features via the built-in APIs to fulfill the exact requirements for a specific network without

the need to release new software generics. This frees carriers from the gridlock caused by relying only on their vendors for new application and features and, enables them to control their own destiny while at the same time providing stable and reliable software in the form of well tested and field proven software generic releases.

The ENTICE Session Controller also frees carriers from these problems and allows them to easily connect with networks from around the world, while providing the security features that they need to protect their current revenue and a path to add services to their network to enhance those revenues.

Ingate Systems

http://www.ingate.com

Ingate (news - alert)Firewalls are the world's first SIP-capable firewalls, making Ingate the only choice for enterprises that want access to SIP-based communications such as presence, instant messaging, audio/video conferencing, and VoIP. Ingate products include a SIP proxy and a SIP registrar, support NAT and PAT, have TLS support for encrypted SIP signalling — which means that instant messages are automatically encrypted — and have been cited by users and media for ease of use.

Ingate Firewalls are cost effective and prevent unauthorized access to and from enterprise networks, while allowing SIP-based communications. All messages entering and leaving the network are routed through the Ingate Firewall, which examines each packet and blocks those not explicitly authorized to pass. Ingate's VPN and SIP modules make it possible for enterprises to adjust the number of users with minimum investment.

Compatible with all existing networks and operating systems, Ingate Firewalls come in a range of models to meet the needs of the entire enterprise market.

However, for enterprises that want to preserve their previous firewall investment, there is the Ingate SIParator. The Ingate SIParator is a device that connects easily to an existing network firewall to seamlessly enable the traversal of realtime SIP-based communications. The Ingate SIParator controls SIP traffic without affecting the security provided by your firewall. Compatible with all existing firewalls, networks and operating systems, Ingate SIParators can support the needs of enterprises of all sizes. SIParators are cost efficient, easy to use, and have the flexibility and scalability required to meet the dynamic needs of today's enterprises.

Juniper Networks

http://www.junipet.net

The highly scalable Juniper Networks (quote - news - alert) VF-series session border controllers ensure service providers and enterprises a fast, expert delivery of IP communications services (voice, video, multimedia) and a cost-effective, competitive edge. VF-series session border controllers enable seamless and secure VoIP networking by resolving security, service assurance (QoS), address translation, and regulatory compliance issues at the edge of VoIP networks.

Purpose-built to enable real-time IP services, VF-series session border controllers are deep-packet processing devices that handle all common VoIP protocols (SIP, H.323, MGCP). They support both signaling and media streams to classify, measure, and manipulate each packet for management and reporting of all VoIP traffic.

Juniper has three SBCs with progressively larger capacities, each designed specifically for different classes of users. The 1000E is designed for service provider networks, managed services, large enterprises; the 3000 should satisfy service provider and carrier edge or core applications; and the 4000 is large enough to handle large service provider and carrier edge or core applications.

Marconi Corp.

http://www.marconi.com

The Marconi (news - alert) Impact IMS Session Controller XCD6000 Release 4.2.0 enables rapid deployment of new multimedia services to generate increased revenues while lowering the costs and risks of deployment.

The Marconi Impact IMS Session Controller removes the constraints of vertically integrated networks: it provides an open, standardscompliant, carrier-grade IP Multimedia Subsystem (IMS) Call Session Control Function (CSCF) that allows voice, video and data services to be converged on a single, horizontally integrated, IP network. You can quickly deploy enhanced multimedia services to customers, via multiple access networks — without replicating core network components.

Key benefits include: Rapid deployment of innovative services with low cost and risk; seamless integration of advanced voice, video, and data services over a converged IP network; increased revenue potential; ubiquitous communications services for customers from a range of terminal devices; integration of SIP terminals with existing voice networks; cost savings through improved operational efficiency; open standard interfaces to other IMS components; seamless integration with existing voice networks.

MERA Systems

http://www.mera-voip.com

MERA's (news - alert) MVTS is a carrier-grade session controller that provides a single interface between multiple IP networks for delivery of end-to-end VoIP communications. It enables safe and secure interconnection with customers and peering partners, protects your network from unauthorized access and DoS attacks, and provides for reliable and cost-effective control of VoIP calls that go through your network. In brief, MERA Session Controller is a complete network management solution for VoIP carriers that combines smart routing, network protection and border control mechanisms all on a single platform.

MERA's MVTS Session Controller is a full-featured solution for carriers and service providers of any size. It provides client networks with a single point-of-entry into the VoIP infrastructure to enable CALEA service, provide centralized authentication, authorization and accounting, as well as NAT Traversal. It supports interoperability with all major equipment vendors and provides enhanced network security through proxy functionality and DoS attack prevention.

Mera's MVTS can increase profit margins by 30% and improve ASR (Average Success Rate) by 50%. It will ensure PSTN reliability and quality, accelerate speed to market from weeks to minutes, reduce CAPEX by 50% and OPEX by 30%. With Mera, you will reach investment payback in less than six months, while being able to grow your network up to 40,000 concurrent sessions.

Netcentrex

http://www.netcentrex.net

NeoXBC (news - alert) is a carrier-class SBC, a new breed of networking equipment specifically built to handle VoIP traffic in real time. NeoXBC makes possible for Telecom Operators and Service Providers to switch their voice services from PSTN to full IP networks and achieve superior profitability and customer satisfaction without compromising security and performances.

NeoXBC enables Telecom Operators and Service Providers to handle SIP/H.323 interworking protocol issues, solve NAT traversal and security issues, manage their bandwidth, collect statistics for billing and safely interconnect with carrier partners via VoIP peering.

NeoGate is a session border controller that answers the needs of private companies and government agencies (CPE). NeoGate enables companies to switch from TDM to VoIP and make substantial savings without putting the whole organization at risk.

NeoGate is located transparently in the company's or end-user's demilitarized zone, by avoiding the use of VPNs which are both expensive and difficult to configure. It adapts to a variety of configurations such as Centrex and PABX trunking, as well as video-telephony via specialized terminals (Polycom, Tandberg, Leadtek, etc.).

Net.com

http://www.net.com

NET's SHOUT (news - alert) platform provides superior session border controller functions within a scalable, voice over IP solution. SHOUT provides the control functions you need to route VoIP call signaling and media; converge separate VoIP protocols, including SIP & H.323; ensure quality and secure calling; and much more.

A standout in the VoIP solutions industry, SHOUT offers advanced functionality that is uncommon to many session border controllers. For example, using SHOUT's session initiation protocol registration feature, an enterprise or vendor can enable SIP phones in a private network to work with an external proxy in the public network. Also, limitations normally associated with NAT traversal and multiprotocol VoIP environments are completely avoided with SHOUT. In addition to advanced session border controller functionality, SHOUT features include:

- SIP to H.323 Conversion
- SS7/C7 Signaling
- Simplified Provisioning & Maintenance
- Billing Record Generation
- Integration of Communication Networks onto One Platform.

If your enterprise is making the switch to voice over IP network communications, NET can help integrate session border controller functionality into your current network. SHOUT is easily deployed into existing networks,

which minimizes setup and equipment upgrade costs.

Netrake

http://www.netrake.com

Netrake's (news - alert) session border controllers resolve the peering, latency, quality of service, capacity and control issues preventing widespread commercial deployment of VoIP.

The nCite session border controllers deliver security, ease of management, media anchoring, session admission control solution, session detail record solution, and interconnection.

Netrake's SBS offer unmatched scalability and are designed to more than satisfy the requirements generated by any economic model, from small to very large. nCite call capacity ranges from 1,000 simultaneous sessions up to 42,000 simultaneous sessions, all in a single compact system. Netrake provides unequaled support, thereby reducing infrastructure requirements by replacing multiple point solutions with a single, integrated (signaling and media) solution and reducing VoIP network complexities.

The ultimate corporate benefit is a reduction in operating costs — nCite SBCs leverage existing investments in networks and back office systems, offer high availability and reliability via fully redundant hardware and software architecture for system processor, switching fabric, and I/O modules, which ensures no loss of calls and the highest quality of reliability.

Deployment of nCite session controllers does not require modifications and/or special configurations to existing network infrastructure including; routers, gateways, gatekeeper capabilities, or security systems.

Newport Networks

http://www.newport-networks.com

(news - alert) The 1460 session border controller enables peering and interconnect between operators. It allows managed IP-based voice and multimedia services to be securely delivered to consumers and businesses.

Key capabilities provided by the 1460 include: The ability to traverse corporate, consumer, and core network NAPT and Firewall devices for SIP services; Quality of Service enforcement via session admission control and policing; Security protection for the core network, for customers, and for service revenue; Regulatory compliance providing Lawful Intercept and Emergency Call Handling.

Other feature include managing traffic volumes; prevention of fraudulent or faulty sessions from exceeding agreed bandwidths, protecting the QoS of other clients; limiting the effects of DoS attacks; enabling media flow differentiation based on a quality policy; enabling media streams to be routed directly between User Agents (UAs) in the same corporate network without transiting the 1460 and consuming access network bandwidth.

The 1460 has no single point of failure, providing in excess of 99.999% availability. It has 1+1 resilience on all system modules, including power distribution units, fans and disks, and physical link aggregation (802.3-2002) providing link resilience and load balancing. The power is distributed into six power zones with front panel indicators displaying the health of each power zone. Management is via dual, independent management networks.

NexTone

http://www.nextone.com

The NexTone SBC (news - alert) is a flexible and intelligent networking device that can be deployed on the enterprise premise or in the carrier's network. Once deployed, it acts as an integrated real-time services firewall and network address translation (NAT) device providing secure connectivity between the carrier and the enterprise. The signaling intelligence of the NexTone SBC ensures protocol interoperability between IP PBXs with support for advanced calling features. The NexTone SBC also provides support for emerging IP PBX features, such as desktop video. With NexTone's SBC, carriers are able to deliver secure real-time services with a level of device connectivity not possible with competing products.

The NexTone SBC utilizes FlexControl technologies, NexTone's unique suite of advanced capabilities which include FlexPeer, FlexRoute and FlexPolicy, to address the complexities of connecting carriers with the VoIP infrastructure of their enterprise customers. FlexPeer is NexTone's signaling software that resolves incompatibilities between VoIP devices by adapting session signaling while also providing control to route, process and manipulate session media. FlexRoute enables carriers to leverage routing facilities such as ENUM to gov-

ern the routing of real time traffic between IP networks while FlexPolicy ensures carriers can manage and control real-time traffic.

This carrier-class device can seamlessly scale from 500 to 30,000 sessions and comes with a hot standby configuration for redundancy which provides stateful call migration. The NexTone SBC is equipped to optionally utilize third party media firewalls to provide advanced features such as media processing.

Quintum

http://www.quintum.com

Quintum (news - alert) offers two SBC product — one for enterprise clients and another for service providers. The two feature the same technology and functionality, with the difference being in capacity (the Call Relay SP for the service provider market must have a higher call handling capacity).

Quintum's Tenor Call Relay provides a single VoIP portal between IP networks allowing for end-to-end VoIP communications across multiple IP networks. All calls are switched through multiple IP networks with just one single compression and decompression of the voice. The result is less latency and higher voice quality as the call passes from network to network.

The Tenor Call Relay allows VoIP endpoints such as VoIP gateways, IP phones and IP soft phones which are behind a NAT firewall, to communicate with VoIP endpoints on external IP networks. This allows both enterprises and service providers to expand their VoIP networks to home offices, branch offices, customers, partners, and across the public Internet.

Tenor Call Relay also provides a single point for call management, administration and security at the edge of your VoIP network. With this unique, intelligent VoIP network switching, the Tenor Call Relay makes expanding VoIP calling both easy and risk free.

With its PacketSaver technology, the Tenor Call Relay reduces bandwidth consumption of VoIP calls by multiplexing multiple calls into the same packet, reducing the overall bandwidth utilization by up to 57%, beyond voice compression and silence suppression.

Sonus Networks

http://www.sonusnet.com

(news - alert) Media gateway functionality is critical in any next-generation architecture, providing voice over packet capabilities and serving as the key transitional element between legacy circuit-switched and packet networks. Media gateways are the primary hardware in a packet design, setting the foundation for voice quality, reliability, scalability and performance.

The GSX9000 Open Services Switch is a revolutionary voice transport component so advanced that it defies definition as a typical media gateway. It provides advanced any-to-any voice transport capabilities and hosts the GSX-GC softswitch module, which provides the call control and signaling for the GSX9000. By supporting this functionality on the gateway, the need for additional computing platforms is reduced and performance is increased by an order of magnitude over alternative packet voice architectures.

The award winning GSX9000 Open Services Switch operates in two key capacities:

As a high capacity media gateway, the GSX9000 delivers toll-quality voice over high-density hardware that seamlessly scales in an efficient manner, maximizing cost effectiveness. The GSX9000 provides carrier-class reliability with a unique internal distributed processing design that optimizes system performance, providing advanced capabilities.

As an integrated component in the OSA, the GSX9000 enables even greater advancements in packet voice solutions. Within this framework, the GSX9000 becomes an intelligent voice switch. This distributed approach that streamlines processing and data flows within the network to maximize performance and scalability, meeting the needs of even the largest global carriers.

Tandberg

http://www.tandberg.com

The TANDBERG (news - alert) network infrastructure products make it easier to connect with people outside of the company, enabling enterprises to seamlessly extend the benefits of video to suppliers, partners, and customers

The TANDBERG Border Controller is a H.323 specific session border controller designed to simplify dialing and firewall traversal for all H.323 devices. This solution solves two of the industries biggest issues, traversing firewalls and global dial plans.

This SBC utilizes an appliance-based architecture that allows for easy deployment and high reliability. It is designed to work with any H.323 device and any firewall and offers full multi-vendor support.

Application features include: Embedded setup wizard; automatic registration of Expressway enabled H.323 devices; flexible zone configuration; support for inter- and intrazone bandwidth control; traversal of any number of firewalls; URI Dialing; device authentication using H.235; and a policy engine for processing calls.

The SBC uses only solid state memory for high reliability, and supports up to 1000 registered devices and 100 concurrent traversal calls, as well as up to 100 neighboring zones and up to 1000 devices and 100 services.

Tekelec

http://www.tekelec.com
http://www.vocaldata.com

The VocalData (news - alert) Session Border Controller serves two primary purposes: It provides normal firewall functions for the devices on the service provider's private LAN, and it provides firewall traversal for IP phones and IAD (integrated access device for connecting analog devices to IP) devices located behind a business or residential customer's existing firewall so their devices can function with the VocalData solution without any modification to the customer's NAT (network address translation) firewall or PAT (port address translation) firewall.

The V-SBC supports multiple control and signaling protocols between the IP endpoints and the VocalData application server. Supported protocols include SIP (define - news - alert), MGCP, SCCP and MiNet.

The V-SBC resides in the service provider network. No additional firewall components or special firewalls need be deployed on each customer network, nor do the customers' data firewalls need any modification. Customers use private IP addresses for all their network-connected devices, including IP phones and IADs.

Redundancy support is provided by deploying the V-SBC in pairs (one active and one standby). Since call state information is continually updated to the standby V-SBC, calls are not lost in the event of failure of the active V-SBC.

Versatel Networks

http://www.versatelnetworks.com

With either of Versatel's (news - alert) SBC solutions, as a service provider, you can deliver a complete spectrum of revenue-generating features including prepaid calling card, tandem, conferencing, voice mail, personal agent, call center, and IVR services, to both circuit-based and packet-based networks from a single EdgeIQ platform. You can save on network bandwidth by delivering full services on the network of origin without backhauling traffic for media processing.

The IQ1500 is the ideal solution for businesses seeking high port density in a small footprint. The IQ4000 is the ideal solution for businesses seeking a fault tolerant carrier-grade solution with 99.999% reliability and no single point of failure.

Either solution can be deployed in existing and new network infrastructures — IP and PSTN. Versatel's IQ-series session border controllers are the most advanced integrated voice services platforms in the industry, integrating softswitch connection control, media gateway, media server, and SS7 signaling gateway functionality into a single compact platform. IT

AMIDEAST Improves Global Workgroup Collaboration, Cuts Travel-Related Communications Costs, With Siemens HiPath OpenScape

Effective communication on a global scale is critical to America-Mideast Educational and Training Services Inc. (AMIDEAST), a private, nonprofit organization that works to establish mutual understanding and cooperation between Americans and people of the Middle East and North Africa.

Each year, the Washington, D.C.-based organization provides English language skills training, educational advising, and testing services to hundreds of thousands of students and professionals in the Middle East and North Africa. AMIDEAST also supports many institutional development projects in those regions and administers educational exchange programs.

The group operates a network of field offices scattered among a dozen countries. Managers and staff in those offices and the Washington headquarters frequently need to be in touch with each other.

Challenge:

Provide Efficient Communications and Collaboration Tools to Geographically Dispersed Facilities and Employees, While Keeping Costs Down

Many of AMIDEAST's facilities and employees are located overseas. As a result, the organization has had to cope with heavy long distance calling expenses. To provide services to customers and conduct other operations, employees in the U.S. frequently need to communicate with counterparts in other countries. Managers and staff spend many hours on audio and Web conferences every month to collaborate on training projects and discuss issues related to their Web-based training model.

But the high cost of international calling had become a burden to the organization. "Most of the time, people avoided making international calls entirely because of the cost," says Ugur Usumi, Director of IT at AMIDEAST. "When people did try to call each other, because of the time zone differences they often wasted time — and money — in failed attempts to contact team members overseas when they were not available."

Obstacles related to time-zone differences are especially challenging for AMIDEAST because many of its field offices are in countries that begin weekends on Thursdays and Fridays. Consequently, Saturdays and Sundays are work days in those countries, but not so for U.S.-based staff.

The problem of high long-distance costs was especially apparent when AMIDEAST employees made calls from hotels in other countries. Typically, these calls cost a minimum of $4 to $5 per minute and, in some cases, as much as $9 per minute, Usumi says. AMIDEAST needed to find a cost-effective way to keep its employees connected.

Solution:

HiPath OpenScape Delivers Presence-Based Conferencing and Collaboration Tool that Helps Users Easily Determine the Most Effective Way to Reach Colleagues

The group explored solutions from a variety of vendors, and selected Siemens HiPath OpenScape, a presence-aware, real-time communications software suite. With OpenScape's single presence-powered view of all users and the devices available to reach them, users know instantly the best way to reach a colleague in real time.

AMIDEAST had been using Siemens communications systems with success for more than 10 years, Usumi says, and that was one of the factors in the decision to deploy OpenScape. Another was the fact that OpenScape adds value to Microsoft's Live Communication Server, which AMIDEAST had recently installed.

AMIDEAST first began using OpenScape in its IT department, with about 10 staff members trying out the system to get a sense of its capabilities. The organization expanded the rollout to users in other departments, including staffers in field offices in the Middle East.

Usumi says the plan is to have about 100 employees, including field operations directors, new business development managers, country directors, and others using the system by the end of this year. Siemens has provided training for the new system.

The implementation of OpenScape is part of an overall upgrade in AMIDEAST's telecommunications system that also includes a move to voice over IP (VoIP) networking. The remote country managers have optiPoint 410 IP telephones and mobile workers will be using the Siemens softclient, the optiClient 130 which allows for easy access to telephony features from the workers PC.

VoIP (define - news - alert) devices are already in place in most of the field offices, and the use of VoIP will help the organization significantly cut down on calling costs, Usumi says.

The upgrades represent a significant change in AMIDEAST's communications, and Usumi was initially concerned about how some employees might adapt to OpenScape and its presence-aware features. But he says the changeover has gone smoothly. The system has been well received by employees, who view the technology as a better way to communicate successfully and more frequently with managers and co-workers.

AMIDEAST is also using the latest version of Siemens' unified messaging product, HiPath Xpressions, in its Washington office. The system offers a single view of all email, voice messages, and faxes in one speech-enabled inbox, a telephony-accessible rules-based filtering folder for easy text-to-speech phone retrieval sessions and extensive multilingual support.

These messaging features, coupled with OpenScape's real-time model, give mobile users around the world the best of both real-time presence awareness and unified messaging for those times when real-time contact isn't practical. HiPath Xpressions supports Microsoft Exchange, Lotus Domino, Novell Groupwise (Groupware) and other IMAP 4/POP3 email environments.

Result:

Upgraded Communications Environment Improves Collaboration While Reducing Costs, and Boosts Team Productivity Worldwide

With OpenScape, AMIDEAST can now more easily set up and conduct audio and Web conferences among its offices in Washington, the Middle East and North Africa. Collaboration is much more effective and users have easier access to meeting documents, an important feature for all the training sessions that AMIDEAST holds for students. AMIDEAST is able to avoid service provider conferencing costs, since all conferences are routed via the existing IP/PBX infrastructure.

Because AMIDEAST can now more efficiently conduct scheduled and ad hoc audio conferences and Web-based collaboration, it will rely much less on costly overseas travel for meetings.

AMIDEAST has also eliminated the risk of security breaches, now that it's using managed, enterprise-based instant messaging with Microsoft Live Communication Server. The organization has increased the value of its Live Communication Server investment by adding OpenScape to the server and extending presence-based communications to voice, video, instant messaging, and e-mail.

The presence-aware view of users and devices enabled by OpenScape allows employees to have immediate access to global team members. Employees no longer waste time and money playing "phone tag" via long-distance calls, Usumi says.

"It's easier to get hold of each other because of features like 'preferred device,'" says Usumi. "No matter where someone is, if you're trying to reach him, you just click on the person's name and it will find him on the preferred device. You don't have to remember seven or eight phone numbers and are accessible when you are traveling."

Most hotels where AMIDEAST workers stay have wireless access and high speed Internet service, so people on the road can set up their preferred device — whether it's a phone or a laptop — and be reached when they're needed.

Recently an AMIDEAST worker in Cairo was having a technical problem and needed help on a Sunday (a work day in Egypt). Using OpenScape, the employee was able to quickly make contact with a systems engineer who was visiting a relative in New Jersey for the weekend. The engineer had programmed the preferred device setting to his relative's phone, so was able to get the call and help his co-worker during his work day in Egypt.

Without the concern of high long distance costs, "people are now more likely to make calls when they need to," Usumi says. "This ability to communicate more easily has increased our efficiency and enabled us to exchange information and make decisions more quickly. Our teams work better as a result." IT

Students, Faculty, and Staff at SJSU Need Up-to-Date Network

Like many universities, San Jose State University had many challenges that caused them to look for new network services to support the needs of their students, faculty and staff. SJSU had mostly 1960s era residence halls that no longer met the needs of students, especially with regards to communications. Student housing infrastructure provided legacy telephony and Internet access but the Cable TV was provided from a third-party. This resulted in high maintenance costs and lost revenues.

Since most students came to campus with their own cell phones and were not subscribing to the dorm room phone service, the university again was saddled with support and maintenance costs without the supporting revenue.

Project Goals

To address these and other challenges, SJSU established a project with the following goals:

- Provide a more integrated communications solution including
  - VoIP, Internet, Edutainment
  - E-learning, distance learning, outreach education
  - Student access portal
- Convert the existing static relationship between SJSU and its students to a dynamic lifelong relationship
- Turn cost centers into revenue centers
- Become an ASP to the other 23 campuses in the California State University system

Project Description

SJSU is the first California higher education institution to implement IP-based triple play to dorms and offices on campus.

The initial pilot project, called “Campus Village” opened in the Fall of 2005. Campus Village is a seven-story, suite-style residence hall designed for freshmen students with world-class amenities. The goal is to provide a more integrated communications and content management solution for VoIP, E-learning, and distance learning. SJSU is turning a cost center into a revenue center.

Campus Village is designed to house and support more than 4,500 students and staff. The dorm rooms are provided with triple play voice, video, and data via “fiber to the pillow.” The triple play services include VoIP (define - news - alert) telephony, high speed Internet access, and video on-demand in the dorm room. Campus Village is also equipped with a computer lab and interactive gaming center, along with on-site restaurants, such as Starbucks.

VoIP and IP Services Deployed

SJSU deployed the Netcentrex MyCall Residential Services Suite and Business Services Suite component of the U-Play3 solution. Netcentrex AllPlay3 Partners also played a crucial role. A converged IP triple play network was installed based on Allied Telesyn’s Integrated Multiservice Access Platforms and Intelligent Multiservice Gateways.

To migrate from and extend the capabilities of previously installed legacy PBXs, a PRI Interconnection Gateway from Audio Codes was used to trunk traffic onto the IP network managed by the Netcentrex softswitch. Finally, SJSU selected IP phones from Swissvoice and video head-end and middleware equipment from Minerva Networks. All of these Netcentrex (news - alert) AllPlay3 partners worked together to provide an integrated turnkey solution for San Jose State University.

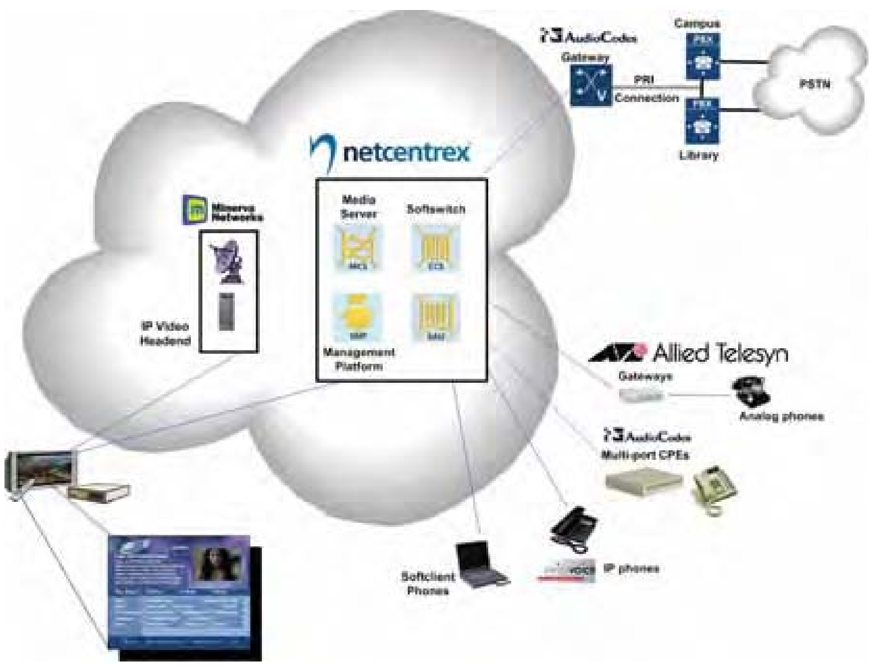
What is U-Play3?

The IPlay3™ Consortium has developed a tailored solution for Universities called U-Play3™ that promotes:

- An unparalleled technology and learning experience for students
- Reduced costs through converged networks and support resources
- New sources of revenue for universities
- Strong campus community and affinity programs for faculty, staff, students, their families, and alumni

Why Netcentrex?

- Tried and tested solution with 41% market share and over 1.8M VoIP lines deployed
- We are the experts in integrated communications including VoIP, IP Video, and wireless technologies
- AllPlay3 partners deliver pre-integrated solutions for gateways, phones, video middleware, etc.
- Superior integration and support
- We are the most reliable, scalable, integrated and future-proof solution on the market



SJSU Campus Village Bundles available to residents

Basic Service :
- Included in monthly rent
- DSL-type Internet access
- Basic telephone and television service
- (100 local anytime minutes, 30 IP-TV channels)

Silver Service:
- Additional per month
- Cable type Internet access
- Basic telephone service package
- Basic telephone service package
- Upgraded television package to 50 channels
- One video on-demand per month
- WiFi access
- Online video game membership

Gold Service:
- Additional per month
- Ethernet Internet access (10 MB)
- Basic telephone service package (plus unlimited long distance)
- Expanded TV up to 100 channels
- Two video on-demand per month
- WiFi access
- Online video game availability
- Premium game center membership

2005 Internet Telephony Products Of The Year

In our January issue, we announced the winners of the 2005 Product of the Year award. This month we've expanded upon the simple list with brief write-ups detailing the companies' offerings. For more in-depth information, please be sure to visit the vendors' Web sites. Keep in mind, this list is intended as a starting point for those interested parties looking to research and purchase a VoIP (define - news - alert) solution. Remember to do your homework, and always ask for customer references!

–GG

3Com
http://www.3com.com/voip
3Com V6000 Integrated Branch Communications

The 3Com V6000 is a cost effective solution for extending the 3Com Convergence Applications Suite to branch offices integrated up to 100 users. It supports advanced SIP-based call control and applications with PSTN and analog gateways into a compact and cost-effective form factor.

AccessLine Communications
http://www.accessline.com
AccessLine's SmartVoice Service for Business

This VoIP-based hosted and managed telecommunications service is designed for medium to large enterprises, even if they are geographically dispersed. SmartVoice allows the enterprise to create and manage a common and consistent enterprise-wide "virtual dial plan," voice/fax messaging services, and other enhanced services such as find-me/follow-me, virtual receptionist, unified messaging, conference calling, disaster recovery and more.

ACE*COMM
http://www.acecomm.com
NetPlus for VoIP

NetPlus is an advanced telemanagement system for enterprise-level voice, data, and IP networks. The solution provisions a VoIP network for accurate fault, configuration, accounting, and performance management. Using a platform-independent architecture, NetPlus reduces telecom costs while enabling smooth, integrated operation.

Aculab
http://www.aculab.com
Prosody X

Prosody X is a card designed for media processing in IP-based networks with TDM connectivity as an option. Designed around an IP core, Prosody X is a highly configurable card that combines media processing resources, IP telephony, and TDM digital network access functions. With channel counts for fully configured cards around 600 Prosody X is an attractive option when looking to create large scale telco or enterprise VoIP applications.

ADC
http://www.adc.com
TrueNet Midspan Power over Ethernet Controller
The solution allows users to upgrade their network to power devices such as VoIP telephones, WiFi access points, and IP security cameras by injecting IEEE 802.3af compliant power onto any standard Ethernet cable. The solution also features dual compliance to IEEE 802.3af and pre-standard Cisco devices as well as N+I reliability using a redundant 400 Watt power supply.

Adomo Inc.
http://www.adomo.com
Adomo Voice Messaging
Adomo provides universal, PBX-independent voice messaging that leverages the existing corporate infrastructure for consolidated administration and lowest total cost of ownership. The solution separates the voicemail application from the PBX, working seamlessly across any mix of legacy and VoIP PBX technologies on a quick-to-deploy and easy-to-manage appliance platform.

ADTRAN, Inc.
http://www.adtran.com
NetVanta 1224STR PoE
Ideal for branch office connectivity, the NetVanta 1224R/STR power over Ethernet (PoE) network access solution combines a 24-port managed Layer 2 Ethernet switch with 24 PoE ports (802.3af), full-featured IP access router (with modular WAN interface), stateful inspection firewall, and optional VPN, in a single 1U platform at a price that is less than 50% of comparable multi-platform solutions.

AltiGen Communications, Inc.
http://www.altigen.com
IP710
AltiGen's IP 710 is a fully-featured IP telephone, offering single-button access to an array of features like, voicemail, intercom, speaker, mute, speed-dial, transfer, conference, activity/presence selection, greeting selections, call recording and even placing calls to employees in other countries. The IP 710 offers a 4-line, backlit liquid crystal display.

Amdocs
http://www.amdocs.com
Amdocs IP Convergence Solution for IPTV
This solution leverages the Amdocs 6 portfolio of billing, content revenue management, CRM, self service, ordering, and mediation products. Carriers can seamlessly integrate critical, customer-facing business processes to gain insight into customer behavior. This helps carriers to develop and support better IPTV offerings and bundles that address specific customer needs and preferences.

Aspect Software
http://www.aspect.com
Aspect Uniphi Suite v6.1
Aspect Uniphi Suite is an operating environment that gives businesses the flexibility and functionality needed for the convergence of contact center applications including ACD, CTI, and IVR on a single, switch agnostic platform that supports both PSTN and IP switches. Uniphi offers a single administration tool and a single point of control for developing and administering business rules across all contact center applications.

AT&T
http://www.att.com
AT&T Voice DNA
AT&T Dynamic Network Applications (DNA) Portfolio is a network-based communications solution that leverages one of the world's most advanced and powerful IP backbone networks and its standards-based, services over IP architecture to deliver a complete IP communications solution delivering greater business productivity through advanced features such as unified messaging, VIP routing, collaboration, conferencing, click to call and remote worker/mobility support.

ATCOM technology co. Ltd.
http://www.atcom.cn
AU-600
The AU-600 enables users to make and receive Skype calls using a standard telephone; forward Skype calls to your mobile phone; make Skype calls from a mobile phone even when away from your computer; and switch between a Skype call and a regular phone call, all the while enabling users to continue to make and receive regular calls as they normally would.

Atreus Systems, Inc.
http://www.atreus-systems.com
Atreus Multi-Service Provisioning Software
Atreus' Multi-Service Provisioning Software empowers providers to improve the profitability of their consumer and business VoIP offerings, as well as a variety of complementary IP services. Through automated provisioning and customer self-management, Atreus enables providers to quickly deploy feature-rich bundles, while reducing the time, cost and complexity of rolling out new services, modifying features, and managing customer growth.

AudioCodes
http://www.audiocodes.com
Mediant 1000
The Mediant 1000 is AudioCodes' cost-effective, converged wireline and wireless VoIP media gateway utilizing cutting edge technology. Neatly packaged in a stackable 1U chassis, it is designed to interface between TDM and IP networks in enterprises or small-scale carrier locations. Incorporating AudioCodes' innovative Voice over Packet technology, the Mediant 1000 enables rapid time-to-market and reliable cost-effective deployment of next-generation networks.

Avaya
http://www.avaya.com
Avaya IP Office with Microsoft CRM
This secure converged voice and data system is designed for small and medium businesses. It has a complete IP telephony customer management portfolio — the Avaya IP Office Customer Management suite, which features Microsoft CRM and includes all the hardware, software and support that businesses need to transform their sales and service operations. The solution combines IP Office with the Sales and Customer Service modules of Microsoft CRM Professional edition, coupling telephony call routing with CRM technology.

BayPackets
http://www.baypackets.com
Agility Multimedia Ringback Tone
BayPackets' Agility Multimedia Ringback Tone is an IMS-compliant streaming media solution enabling wireless carriers and cable MSOs to replace the typical "ring" or busy sound callers hear when they dial that user's number with recorded music, sound, or video clips. The application is as an enhancement to BayPackets' Agility Personal Communications Manager, a network-based solution that enables service providers to deliver real-time call routing and call management functions.

BEA Systems, Inc.
http://www.bea.com
BEA WebLogic SIP Server
The BEA WebLogic SIP Server is a combined J2EE and SIP application server, delivering an integrated service creation/delivery platform for SIP, HTTP/Web-tier, and EJB-tier applications. The SIP Servlet container is

completely integrated with the HTTP and EJB containers of the BEA WebLogic Server. As a result, SIP Servlets have access to all the J2EE APIs, such as JNDI for naming and directory services, JDBC for database access, JMF for media handling (codec, RTP), JavaMail APIs, and many others.

BlueNote Networks
http://www.bluenotenetworks.com
SessionSuite

BlueNote Networks' SessionSuite IP Telephony is an open, modular, standards-based IP software platform enabling communication services to be embedded into business applications within and beyond traditional enterprise boundaries. The solution leverages enterprise infrastructures, the PSTN, and the Internet to seamlessly extend voice services beyond desktop phones while reducing costs and preserving current investments.

Brekeke Software, Inc.
http://www.brekeke.com
OnDO PBX

OnDO PBX is a SIP compliant software-based telephony system, including many features of traditional PBX systems, such as; call transfer, call conference, call forwarding, automatic route selection, voicemail, and more. OnDO PBX comes bundled with OnDO SIP Server, a SIP Registrar and Proxy Server, registers and authenticates users, and routes calls between user agents. The product is ideal for small offices or branch offices already on a SIP VoIP network; the SmallOffice Edition supports 40 PBX users and 12 concurrent call sessions.

Brix Networks
http://www.brixnet.com
BrixCare Self-Service

BrixCare Self-Service is a VoIP quality measurement and reporting application that helps service providers reduce subscriber acquisition and support costs. The BrixCare Self-Service application measures VoIP quality from the customer premise, and enables providers to profitably launch, operate, and assure their residential and business broadband voice services while increasing customer satisfaction and loyalty.

Brooktrout Technology
http://www.brooktrout.com
SnowShore IP Media Server — Software only

Release 1.4 of the SnowShore IP Media Server provides carriers and application partners with unprecedented capacity expansion for large scale audio conferencing sessions, high-performance processing of large bandwidth video streams for multimedia and messaging services, and advanced collaboration feature support for contact center applications. The product's flexible architecture provides a cost-effective, scalable media server platform supported by a growing list of applications, speeding deployment of media-rich, high-value enhanced services.

Centillium Communications
http://www.centillium.com
Atlanta Family

Centillium's newly expanded Atlanta series — which includes four unique devices and easy-to-use, intuitive application software — is a complete, turnkey System on a chip (SoC) solution designed to allow equipment manufacturers to reduce development time and preserve software investments across multiple platforms. Each Atlanta device features a complete software suite, including an application program interface, drivers, and documentation for faster production time.

Check Point Software Technologies
http://www.checkpoint.com
Check Point VPN-1 Pro NGX

Check Point VPN-1 Pro NGX is a tightly integrated combination of firewall, VPN, and intrusion prevention that provides organizations with a platform for integrating VoIP into a comprehensive security policy without needing to sacrifice their deployment goals. With Check Point's Stateful Inspection and Application Intelligence technologies, VPN-1 Pro delivers an intelligent perimeter security solution available for VoIP.

Cicero Networks
http://www.ciceronetworks.com
CiceroPhone

CiceroPhone is a converged VoIP and Cellular softphone, optimized to run on mobile devices, including smartphones and PDAs. It enables users to make all of their calls — VoIP, Fixed, and Cellular Calls — using a single device. CiceroPhone runs on mobiles phones and delivers seamless mobility across IP and Cellular networks. It supports hand-off between Wi-FiI and Cellular networks so that users can seamlessly roam between networks without dropping a call.

Citrix Systems
http://www.citrix.com
Smart Agent for Citrix Application Gateway

Citrix Application Gateway is an appliance that delivers converged voice and data applications to the screens and speakers of IP telephones. Smart Agent technology provides productivity applications to users' personal computers in addition to IP telephones. The first Citrix application to use Smart Agent technology is Click-to-Call which enables telephone users to dial their desk telephone by simply clicking on linked telephone numbers in e-mail or Web applications.

Coaxsys
http://www.coaxsys.com
TVnet/C

TVnet/C transforms a building's coaxial cable into a high-speed IP network that coexists with the cable television signal, delivering a minimum sustained throughput of 100Mbps. Plug-and-play TVnet/C adapters work over a building's pre-existing coax. TVnet/C is a 100Mbps Cat-5 Ethernet replacement that enables an operator or service provider to network any Ethernet device without needing to run new wires. Users can connect IP set-top boxes, modems, media servers, PCs, video game consoles, etc.

Cognio
www.cognio.com
ISMS Mobile

Cognio's Intelligent Spectrum Management System (ISMS) Mobile is a laptop-based RF spectrum analysis solution designed to provide full spectrum analysis capabilities for identifying and locating specific sources of interference that undermine the security and performance of wireless networks.

CommuniGate Systems
http://www.communigate.com
CommuniGate Pro version 5.0

CommuniGate Pro Internet Communications server, provides VoIP applications and standards based messaging on a massively scalable applications server platform. Industry first SIP Farm clustering technology and the Active Dynamic Cluster allow large scale subscriber deployments in the 100M range, with geographic separation of cluster members for global deployments. Version 5.0, expands the software's voice applications to include: TeleConference Server, Auto-Attendant/IVR, Call Queuing/Automatic Call Distribution, Voicemail and Self-Service.

Convedia
http://www.convedia.com
Convedia Media Servers

Convedia's family of media servers allow service providers to reduce service delivery costs and increase revenues through new approaches to network based services. Convedia's media server product family includes the CMS-6000 Media Server, which is capable of scaling up to 18,000 media processing ports for large next genera-

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2005 INTERNET TELEPHONY PRODUCT OF THE YEAR

tion network deployments. Convedia also produces the entry-level CMS-1000 Media Server, which provides up to 300 media processing ports for smaller service providers, large enterprise, and network edge deployments.

Converged Access, Inc.
http://www.convergedaccess.com
Converged Access Point (CAP)

The Converged Access Point (CAP) is an integrated access platform purpose-built to deliver the application performance, WAN efficiency, security and low TCO required by Enterprise Remote Office and SMB/SME users as they converge business-critical voice, data, and video applications. The CAP addresses the challenges that have stalled VoIP deployments by combining a VoIP gateway, precise traffic management, firewall and VPN security, wired and wireless access, and an intuitive policy management solution.

Convergin
http://www.convergin.com
Accolade Platform

The Convergin Accolade platform is a commercially deployed solution for Fixed Mobile Convergence. Convergin uses SIP to tie the network together (in either the wireline or wireless side) and extend application services in a single blade server at the network core. This allows a VoIP service provider to add wireless services with just a Convergin wireless convergence server (WCS) and either a standard softswitch or application server.

CopperCom
http://www.coppercom.com
The CopperCommander Management System

The CopperCommander system is a comprehensive management system designed to enable carriers to simplify the deployment of sophisticated multi-nodal CSX switch networks, allowing them to operate at a significantly lower cost. The CopperCommander meets the needs of IOCs and CLECs whose networks serve large and often dispersed geographic areas.

CosmoCom
http://www.cosmocom.com
CosmoCall Universe 4.5

CosmoCall Universe (CCU) is an all-IP contact center platform with field-proven, carrier class reliability designed to empower organizations to achieve higher levels of customer service while reducing costs, by supporting all of their enterprise-wide contact center needs from one virtual contact center platform, self-hosted or hosted by a network service provider.

CounterPath Solutions, Inc.
http://www.counterpath.com
X-Lite for Linux

CounterPath's softphones for Windows, MAC OS X, Pocket PC and now Linux are software applications, which run on devices such as personal computers (PCs) or personal digital assistants (PDAs) or can be integrated into set-top boxes, embedded devices and applications like instant messaging. X-Lite users can make and receive calls to and from other on-net callers (IP-to-IP) and/or off-net callers (IP-to-PSTN/Cellular) with the same quality of service they would expect from an IP handset.

Covad Communications
http://www.covad.com
Line-Powered Voice Access

Line-Powered Voice Access (LPVA) is a VoIP-based replacement for traditional phone service. It offers enhanced alternative service at a competitive price. It does not require a broadband connection or additional consumer premise equipment. Customers use their existing analog phone equipment and in-house jack. As with traditional voice service, the line is powered from the central office (CO), so a power failure will not disrupt service. To the end user, LPVA works just like a regular phone line.

CTI²
http://www.cti2.com
InTouch

InTouch is an open, IP-based and standards-based next-generation messaging and conferencing platform, designed to provide a comprehensive suite of advanced communications and conferencing services in a single platform. InTouch services include IP-Voicemail, enhanced Visual Voicemail, Voice Conferencing, Voice to MMS, Who Called, Video Messaging, Mobile Mail, Unified Communications and more.

Data Connection Limited (DCL)
http://www.dataconnection.com
DC-SBC

DC-SBC is available as a complete source code solution for vendors of softswitches, routers, gateways, firewalls, and IP PBXs as well as standalone session border controllers. By incorporating DC-SBC into their existing products, OEMs will be able to offer service providers complete Session Border Controller (SBC) functionality as a software upgrade to platforms already installed in their networks. System vendors benefit by providing platforms with increased service and security capabilities but at greatly reduced development costs and time-to-market.

DecisionOne Corporation
http://www.DesktoptoDialtone.com
Desktop to Dialtone

Desktop to Dialtone is a portfolio of support services that provides an immediate national infrastructure for VoIP OEMs and Service Providers (VSPs) spanning the VoIP lifecycle including network assessment, logistics, field delivery, and service desk. For their end customers, these services translate convergence into a simple, cost effective, full-service solution for reliable carrier class VoIP and data communications, and allay concerns regarding reliability and support.

deltathree
http://www.deltathree.com
iConnectHere

iConnectHere is the consumer group of deltathree, whose products include Broadband Phone, PC to Phone and Virtual Calling Cards, to end users worldwide. iConnectHere recently expanded availability of its direct inbound dialing (DID) numbers to ten European counties including, Austria, Belgium, France Germany, Ireland, Italy, Spain, Sweden, Switzerland and the U.K. Additionally, more than 2,000 rate centers in 44 U.S. states with a total coverage of more than 185 million people (70 million households) are now available in 90% of the U.S.

Ditech Communications
http://www.ditechcom.com
Packet Voice Processor

The Packet voice Processor (PVP) voice processing platform enables carriers to overcome numerous challenges including addressing voice quality, transcoding of a broad array of compressed voice codecs, and interoperability problems of VoIP networks. The PVP is based on a high-capacity packet system incorporating Ditech's high-density media processing subsystems. It can be configured up to 16 gigabit Ethernet ports and scale to 50,000 transcoding and/or voice processing calls per rack. PVP supports current VoIP standards including SIP.

Dovado
http://www.dovado.com
WRG — Wireless Residential Gateway

Using a PC-card as backhaul (PCMCIA-WWAN), the WRG is able to provide subscribers with a secured form of Internet and WiFi access along with telephony services. The WRG is a very suitable device for the home or office where one can connect computers, phones, and fax machines upon a quick installation. The WRG can be upgraded to communicate with any new wireless broadband standard, thus offering the ability to protect the investment in the future.

Embedded Communications Computing, Motorola
http://www.motorola.com/computers
VoIP Open Application-Enabling Platforms from Moto

Motorola's VoIP Open Application-Enabling Platforms integrate Session Initiation Protocol (SIP) software with CompactPCI packet voice resource boards, allowing VoIP equipment manufacturers to voice-enable SIP applications without the need to generate low-level code or directly control hardware.

Empirix Inc.
http://www.empirix.com
Hammer DEX

Hammer DEX is a VoIP device emulation solution that provides 'out of the box' VoIP device emulations combined with a flexible platform that allows emulation of custom VoIP network devices. It enables equipment manufacturers and service providers to emulate a wide variety of VoIP devices on a single platform in their test labs, thus enabling them to perform testing without incurring the substantial costs of building out a full network infrastructure in their labs with 'real' vendor devices.

Envox Worldwide
http://www.envox.com
Envox 6 Communications Development Platform

Envox 6 is an open, standards-based communications development platform with integrated application development and management components designed to reduce the time, cost, and complexity of creating voice solutions. The platform provides a bridge between legacy systems and new innovative voice solutions, allowing customers to leverage prior investments in hardware, software, and solution development, while providing a smooth migration path to emerging standards and enabling technologies.

Esnatech
http://www.esnatech.com
Telephony Office-LinX version 6.5

The solution is a real-time instant communications solution for enterprise customers. Whether it's a large campus, distributed offices or a mobile workforce, this all-in-one unified communications platform offers a number of features, including Unified Messaging, Wireless connectivity, CTI call control, one number Find Me functionality, Web access, Instant Messaging, Speech Recognition, Text-to-Speech for e-mail.

Excel Switching Corp.
http://www.excelswitching.com
Integrated Media Gateway (IMG) 1010

The IMG 1010, generally available since June 2005 offers a compact 1U package providing carriers with a cost-effective entry point into VoIP, with the ability to scale and gracefully upgrade their networks as subscriber base increases. The solution also offers interoperability between SS7, SIP, and H.323 across multiple gateways so carriers can roll out enhanced telecommunication offerings without buying, integrating, or managing third-party softswitches, signaling gateways, application servers, media servers, proxy servers, or registration servers.

FacetCorp
http://www.facetcorp.com/facetphone
FacetPhone

FacetPhone is designed and built from the ground up as an IP-based phone system. FacetPhone's multi-location integration, call center features, presence management, visual voice mail, unified messaging, integrated instant messaging, IVR flexibility, graphical customer administration, built-in multi-party conferencing and telecommuter support, provides customers with significantly increased productivity and a rapid return on investment.

Flarion Technologies
http://www.flarion.com
NETGEAR Mobile Broadband Router 814 (MBR814)

This 802.11g-standard router will deliver a secure and reliable wireless Internet connection without having to plug into a DSL or cable connection. The MBR814 supplies complete network access with firewall protection so consumers can enjoy high-speed Web access, file sharing, video streaming, head-to-head Internet gaming, and MP3 downloads at home. The 54 Mbps wireless router provides state-of-the-art filtering and controls that allow parents and administrators to limit URL access and monitor Internet activities.

Flextronics Software Systems
http://www.flextronicssoftware.com
SIP Network Server (SIP NS)

SIP Network Server (SIP NS) is a carrier-grade, feature rich, core infrastructure server, based on an architecture that separates routing from application servers. SIP NS provides carriers the flexibility to quickly deploy innovative revenue generating services, such as prepaid, video, IM, P2P etc., from a amongst the best-in-class application server vendors. It has been guaranteed to interoperate with leading products and vendors in the marketplace. It can support as many as 600,000 subscribers.

Fonality
http://www.fonality.com
PBXtra

PBXtra is an IP-PBX that delivers enterprise-class capabilities to small businesses. Combining the powerful call handling, telecommuting, branch-office support, VoIP/PSTN capacity, and unified messaging features that companies expect from an enterprise-class PBX system, PBXtra delivers low cost of ownership — often saving companies up to 80 percent of what they are used to paying for PBX functionality.

GigaBeam Corporation
http://www.gigabeam.com
WiFiber Wireless Fiber

GigaBeam WiFiber is a true fiber substitute for the entire last mile. Customers need to increase network capacity while they lower access and maintenance costs. GigaBeam links are designed for the highest performance and availability and the lowest total cost of ownership. They are designed to improve network availability and resiliency to failure while reducing the costs of installation, network integration, and maintenance.

Go2Call.com, Inc.
http://www.go2call.com
Go2Call Softphone

Go2Call's SIP Dialer V9 softphone is designed for optimization in a wide variety of Internet environments. The dialer provides accelerated performance and a built-in diagnostic and feedback channel. In addition to these technical improvements, Go2Call has also empowered this latest version of the softphone with easy to employ branding and skinning options. Other features include multilingual capabilities, call hold, call waiting, call forwarding, and short-digit extension; voicemail integration; and more.

Grandstream Networks, Inc.
http://www.grandstream.com
GXP-2000

GXP-2000 is a next generation IP phone for the enterprise market. It has dual routed or switched (user configurable) 10M/100M auto-sensing Ethernet ports, integrated Power-over-Ethernet (802.3af) along with a universal power switching adapter (standard offering), large (130x64 dot matrix) graphic LCD that supports multiple languages, 11 line appearances with seven programmable keys, up to four independent SIP accounts and much, much more.

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I.S. Associates, Inc.
http://www.isassoc.com
TeleCount Billing

TeleCount Billing is a modular yet integrated billing and customer care solution, enabling carriers and other service providers to handle billing for both retail and wholesale customers. TeleCount offers a powerful and scalable real-time platform supporting authentication, authorization, accounting, mediation, rating, billing, inventory, and CRM for a variety of pre- and post-paid services including voice, data, and content.

IBASE Technology, Inc.
http://www.technoland.com
PICMG 1.3 Intel Pentium M SHB Express IB868

The IB868 PICMG 1.3 CPU card features the mobile Intel GM Express chipset with an integrated 32-bit graphics engine, supporting both CRT and LVDS LCD displays. Accommodating up to 2GB of DDR2 533MHz system memory with improved bandwidth and power savings over DDR memory, the low-power SBC is validated with the Intel Pentium and Intel Celeron M processors on 90nm process.

iBasis
http://www.ibasis.com
DirectVoIP Broadband

Through its DirectVoIP Broadband service, iBasis provides a complete solution that enables VoIP providers to offer consumers seamless connections to traditional PSTN phones and mobile phones worldwide. DirectVoIP Broadband includes fast and secure IP interconnection to the iBasis global VoIP network, network-to-network SIP authentication, and support for multiple protocols and the growing array of IP phones and softphone applications.

Incognito Software Inc.
http://www.incognito.com
Broadband Command Center Appliance

Incognito's Broadband Command Center Appliance is a turn-key device provisioning system for small- to- medium-sized broadband providers who need to automatically configure customer premise equipment (CPE) for high-speed data, VoIP, and video services. Supported CPE includes: DOCSIS modems and PacketCable and SIP multimedia terminal adapters and VoIP phones.

InfoVista
http://www.infovista.com
VistaInsight for IPT Telephony

VistaInsight for IPT addresses the complexity of IP telephony and assures the delivery of services across the enterprise through a service-centric approach to performance management. The solution focuses on monitoring the quality of the end-user experience as the objective measurements of service quality. Automated discovery and asset relationships across the IT infrastructure facilitate both top-down service management and bottom-up impact analysis.

Ingate Systems
http://www.ingate.com
Ingate Remote SIP Connectivity with SIParator 45/45+

Ingate Remote SIP Connectivity is a far-end NAT traversal solution that enables remote users to leverage the benefits of SIP communications already integrated into the company's network. Remote SIP Connectivity allows users behind many remote (non SIP-aware) NATs and residential firewalls to communicate via SIP without the need for adding additional hardware or software at the remote site, simply by installing the Ingate SIParator at the enterprise data center.

InSciTek Microsystems Inc.
http://www.allworx.com
Allworx 10x

The Allworx 10x, the cornerstone device of a networked phone system for small-to-medium sized businesses (under 100 employees per site), integrates into a single server all the various communications platforms — VoIP, phone, PC network, messaging software, group collaboration capabilities — necessary for businesses to succeed today.

Intel
http://www.intel.com/go/iptoday
Intel NetStructure Host Media Processing Software

Intel NetStructure Host Media Processing (HMP) Software combines the flexibility of software-based IP media processing with powerful high-density multimedia capabilities that run on a server's host processor. HMP is available in a Linux and Windows version. The version for Linux scales up to 240 concurrent user sessions, with media processing, per system and supports video.

InterEdge Technologies, LLC
http://www.inter-edge.com
Dial-Up VoIP Intelligent Telephone Adapter ITA-100

The ITA-100 allows VoIP service providers to expand their market by providing their offering to customers that use Dial-Up connection to access the Internet. The ITA-100 has an integrated modem that can connect up to 56 kbps and can be switched between the PSTN and the VoIP service provider by just pressing a pre-programmed code in the keypad of the user's telephone set. The ITA-100 supports traditional voice codec such us G.729.A and G.723.1. It also support SIP, DHCP Server, V.90, etc.

Interoute Communications Ltd
http://www.interoute.com
ARENA

Arena is a commission-free voice trading exchange that makes voice trading easier and erodes barriers to market entry. It is the latest service on Interoute's Virtual Voice Network (VVN), based on softswitch technology that enables operators to enter the voice market quickly without costly infrastructure investment. The service also provides for full route management and reporting via the desktop.

Interstar Technologies
http://www.faxserver.com
Interstar XMediusFAX Service Provider (SP) Edition

Interstar's XMediusFAX SP is a carrier-class, fax server software for service providers and large enterprises. Service providers can increase cash flow by purchasing their own Fax over IP (FoIP) software, instead of re-branding third-party fax services. XMediusFAX SP enables easy packaging, marketing and expansion of fax services per evolving business needs.

Inter-Tel, Inc.
http://www.inter-tel.com
Inter-Tel 5000 Network Communications Solutions

The Inter-Tel 5000 Network Communications Solutions is a family of feature-rich, IP-centric communications systems developed specifically for small- and mid-size businesses. Included is the Inter-Tel CS-5200, which is geared for small businesses by supporting up to 25 IP endpoints, while the Inter-Tel CS-5400 can scale up to 110 IP devices. In addition, both systems can support up to 96 digital endpoints via optional digital expansion units.

Interwise Inc.
http://www.interwise.com
ECP Connect 6.0

Interwise ECP Connect 6.0 is a complete, secure Web and voice conferencing capabilities for enterprises. Launched in 2005, ECP Connect revolutionizes how enterprises purchase and use conferencing — moving them from expensive, usage-based external services to a rich IP-based application that is available to every desktop, like e-mail.

2005 INTERNET TELEPHONY PRODUCT OF THE YEAR

Iotum Corporation
http://www.iotum.com
Iotum Relevance Engine

Based on a user's context information from existing information management tools like IM, calendar, contacts, and personal user preferences, Iotum's Relevance Engine intelligently filters, ranks and prioritizes digital communications. Iotum's Relevance Engine automatically knows which calls to take and where to send them: landline, office, VoIP, voice mail, colleague or simply ignore.

Iwatsu Voice Networks
http://www.iwatsu.com
Enterprise-CS powered by QuadFusion

Enterprise-CS is a converged voice processing system — powered by QuadFusion Technology — designed to provide small to medium sized businesses with advanced communication features. Enterprise-CS scales from 10 to 1000 telephones and provides advanced features like 100% transparent IP networking, in-building wireless, voicemail, and ACD integrated into one platform. Additionally, unified communications can be added to provide productivity enhancing features like text-to-speech, automatic speech recognition, fax server and IVR.

Juniper Networks
http://www.juniper.net
E320 Broadband Service Router

The E320 Broadband Services Router is a high-capacity broadband router scaling from 100 to 320 Gbps, and supporting up to 128,000 subscribers on a single chassis. This combination of capacity and subscriber density greatly reduces the average cost per subscriber. The E320 also provides a complete set of high-availably features, support for carrier-class Ethernet aggregation, and fine-grain QoS treatment for multimedia services — all of which enable service providers to deliver advanced services with industry-leading levels of quality, security and availability.

Kayote Networks, Inc.
http://www.kayote.com
FronTier — Advanced VoIP Traffic Management System

FronTier is a hosted VoIP solution that uniquely combines the features of a session border controller (SBC) and a quality-based least cost routing (LCR) engine with advanced, multi-level security features, specifically adapted to the VoIP call flow. The fully customizable Web interface enables Frontier users to take advantage of billing management tools to measure vital data in real-time, analyze network statistics, and generate carrier grade reports.

Kentrox
http://www.kentrox.com
Q1300 QoS Appliance

The Q1300 makes prioritizing traffic for critical applications (VoIP, video, and other business systems) and managing bandwidth practical. A graphical Web interface provides instrumentation to monitor, diagnose, and add policies to ensure proper network performance. The Q1300 combines the features of a QoS appliance and Ethernet switch into one easy-to-use network access device. Graphic reporting also helps ensure that QoS is performing in the network.

Level 3 Communications
http://www.level3.com
Level(3) E-911 Direct

Level(3) E-911 Direct is an FCC-compliant Enhanced 911 (E-911) solution for interconnected VoIP providers. It includes a nomadic VoIP solution and a fixed-line solution with network connections to PSAPs that serve 68 percent of all U.S. households. The service enables the routing and terminating of 911 calls with physical location information to the appropriate selective router and Public Safety Answering Point (PSAP) within the Level 3 footprint.

Lucent Technologies
http://www.lucent.com
Lucent IMS Service Enhancement Layer

Lucent's Service Enhancement Layer (SEL) is a library of six unique Bell Labs-developed software technologies that give Lucent's IMS solution distinct advantages over competing IMS implementations. The SEL helps enhance the end-user experience by enabling wireless, wireline, and converged network operators to create and deliver simple, seamless, secure, portable, and personal multimedia services to their subscribers. Components of the SEL work hand in hand with elements of Lucent's IMS-based solution.

LumenVox
http://www.lumenvox.com
Speech Driven Assistant

LumenVox's Speech Driven Assistant for Vertical Communication's TeleVantage IP-PBX is a complete turnkey program to speech-enable name directories, contact lists, voicemail, IVRs, and emails, providing all callers hands-free phone interactions. The solution integrates directly on the TeleVantage server and imports all user information and continuously synchronizes with TeleVantage's database.

MCI
http://www.mci.com
Next-Generation VoIP Portfolio for Business

Building upon MCI's advanced Hosted IP Centrex and IP Integrated Access offerings, MCI's Managed IP PBX Services lets companies offload the management of planning, designing, and implementation of their premise-based IP telephony systems. The MCI Advantage IP Trunking Service enables customers to carry local, long distance, and data traffic on a single converged network. This eliminates the need for customers to purchase costly IP to PSTN gateways to terminate and convert VoIP calls within the network.

Mediatrix Telecom Inc.
http://www.mediatrix.com
Dial IPCS

Based on Session Initiation Protocol (SIP) standards, DiaL IPCS is a cost effective, all-in-one Communications Server allowing the deployment of VoIP communications solutions that preserve your investment in legacy telephones and faxes. Used in combination with Mediatrix gateways, DiaL IPCS is designed to enable enterprises to deploy VoIP solutions that connect distant locations, while maintaining all major PBX features available in their head office.

Meru Neworks
http://www.merunetworks.com
Radio Switch family

The Meru Radio Switch family, includes the four-radio RS-4000, the eight-radio RS-8000, the twelve-radio RS-12000 and a patent-pending, built-in omni-directional antenna. The system delivers up to 648 Mbps of WLAN bandwidth utilizing 802.11 standard radios, and is capable of scaling up to 1.2 Gbps of bandwidth across its coverage zone. The Radio Switch accomplishes this performance by layering up to 12 channels in a single coverage area.

MetaSwitch
http://www.metaswitch.com
UC9000 Unified Communication Systems

The MetaSwitch UC9000 Unified Communications System is an extensible, standards-based platform targeted at service providers seeking to provide new services that attract new subscribers and increase revenues from existing customers. The initial release of the UC9000 included voicemail and Unified Messaging, with support for Web, PDA, telephone and Outlook clients.

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2005 INTERNET TELEPHONY PRODUCT OF THE YEAR

Minacom Labs, Inc.
http://www.minacom.com
PowerProbe 30 Service Level Responder & PocketDQ

The responder is the world's smallest comprehensive test solution for VoIP/VoDSL/VoCable acceptance testing and troubleshooting. When connected to any standard two-wire phone jack, a technician can initiate a complete assessment of over 60 service level measurements. Integration into Minacom's DirectQuality R7 Service Level Assurance platform provides standards-based testing, centralized data logging and reporting, third-party OSS integration, and Service Level classification.

MIND CTI Ltd.
http://www.mindcti.com
iPhonEX

The MIND-iPhonEX solution for IP services supports both prepaid and postpaid subscribers. The real-time interaction with the network elements enables MIND-iPhonEX to control the call and cut it off as the customer's balance bottoms out. MIND-iPhonEX balance manager enables simultaneous usage of multiple services on a single, centralized balance and provides up-to-the-minute account information for both, subscribers and CSRs.

Mitel
http://www.mitel.com
Mitel Navigator

Mitel Navigator is a communications device that enhances the meaning of voice, video and data convergence on the desktop. It is designed to fit neatly under a flat screen monitor and can be personally tailored for specific horizontal and vertical market applications. The end result is a phone with an innovative form factor that can deliver powerful functionality and applications via a tightly knit interface to the PC world. Navigator will work in conjunction with Mitel Your Assistant and Microsoft's Live Communications Server.

NEC Unified Solutions
http://www.necunified.com
SV7000 Multiple Purpose System (MPS)

The MPS supports 1,000 licenses divided between 500 IP Station licenses and 500 Analog stations IP trunks and PSTN trunks and is easily scaled to its maximum size. A robust set of applications is bundled with the SV7000 MPS, including desktop productivity tools, unified messaging and an intuitive web-based management system designed to reduces OPEX costs. The Operating System for MPS server is Linux-based, provided on a single CPU blade, which provides traditional telephony feature support as well as NEC Extended SIP and Standard SIP support.

Nero Inc.
http://www.sippstar.com
Sipps Connect

Nero SIPPS Connect is an intuitive Internet phone that offers users all the features they expect along with a high level of customization. Users can choose from different colors to customize the UI; use the three-way calling feature to conference in several parties at one time; select from different ring/knock tones; have calls forwarded to another phone or cell phone; and more.

Net2Phone Inc.
http://www.net2phone.com
VoiceDirector

VoiceDirector is a robust enterprise VoIP solution, combining IP-PBX functionality along with local, long distance and international calling plans. VoiceDirector is a complete service solution that allows corporate customers to obtain all voice communications using Net2Phone's platform, leveraging Net2Phone's technology and termination services. Built on open source software, VoiceDirector utilizes the SIP standard.

Netcentrex, Inc.
http://www.netcentrex.net
MyCall v3

MyCall v3 is a Class 5 Application Server delivering a complete set of services ranging from voice and video mail to voice and video portal service activation and Web self-care for true innovation and rapid return on investment. The solution comprises a fully integrated turnkey provisioning system, which ensures the endpoint inventory and provisioning functions, as well as comprehensive support for complex access network configurations including NAT traversal and media relay functions.

Newport Networks
http://www.newport-networks.com
Newport Networks 1460 Session Border Controller

The 1460 can handle over 100,000 concurrent calls. Newport physically separated its SignallingProxy and MediaProxy allowing MediaProxies to be controlled from third-party Softswitches and SignallingProxies to be deployed standalone as 'SIP signalling firewalls'. This flexibility enables service providers to deploy the solution either as a standard session border controller or as a separated Signalling Proxy and Media Proxy that can evolve to IMS and TISPAN architectures without major upgrades.

NewStep Networks
http://www.newstep.com
Mobile Call Handoff

With Mobile Call Handoff, dual-mode wireless phones hand off automatically from WiFi access to cellular access and back again as users move between coverage areas. For maximum reliability, a unique Call Retention feature even restores calls automatically if a WiFi connection is lost. With Mobile Call Handoff, enterprises enjoy single-number service, expanded wireless coverage, more reliable mobile communication, and the economic advantages of VoIP.

Nortel
http://www.nortel.com
Business Communications Manager 50

The Business Communications Manager 50 (BCM 50) is an all-in-one platform for converged voice and data communications for small to mid-sized business sites. Ideal for organizations with three to 20 users, yet scalable to serve 40+ users, the BCM 50 comes pre-loaded with a large portfolio of telephony features, more than 200 in all, enabling enterprises to process calls with exceptional reliability, efficiency and flexibility.

Occam Networks
http://www.occamnetworks.com
BLC 6312

The BLC (Broadband Loop Carrier) 6312 is an Active GigE Fiber to the Premises product for supporting FTTH and FTTB applications. It is designed to carry traffic at rates up to 1Gbps to each endpoint including multiple VoIP voice lines, high speed internet, large number of HD TV or standard definition IP TV channels. The BLC 6312 enables carriers to deploy more bandwidth than a OC192 at a fraction of the cost.

Orative
http://www.orative.com
Orative Enterprise Software

Orative Enterprise Software is a secure, client-sever solution that works across different networks and a variety of mobile handsets. In addition to presence information, the solution also provides the ability to quickly coordinate conversations, collaborate with colleagues, screen unwanted calls and interruptions, and access enterprise phone directories.

Pactolus Communications Software
http://www.pactolus.com
RapidFLEX Call Complete

RapidFLEX CallComplete software insures that subscriber calls continue interrupted in the event of a RapidFLEX Application Server hardware or software failure. This call protection capability works using 1:1 server redundancy and patent pending RapidFLEX software to ensure that SIPware Services — such as primary line broadband telephony (VoIP), audio conferencing, calling card, and voice messaging — stay running without subscriber intervention and maintain maximum system capacity.

Pandora Networks
http://www.pandoranetworks.com
Worksmart

Pandora Networks' Worksmart is an On Demand IP Communications solution designed for small and medium sized businesses (SMB). Worksmart enables SMB IT managers to deliver voice and data services that include virtual IP-PBX and ACD functionality; private and public instant messaging; Web-based presentation and collaboration; audio/video/IM/Web conferencing; Web-based contact center; and many other services from a browser and thin client to any office or employee worldwide.

Pangean Technologies
http://www.pangeantech.com
insta-REACT!

insta-REACT! is a SIP based converged solution for instant voice communications. It provides an all-in-one communication solution for the enterprise market. The Solution provides secure, standard-based multimedia communications including presence, voice broadcast, instant messaging, paging, ad hoc conferencing, and more. All these features are delivered on a single software user interface on the user's PC using the corporate network.

Pannaway Technologies, Inc.
http://www.pannaway.com
Broadband Access Manager (BAM)

Pannaway's Broadband Access Manager (BAM) is a sophisticated broadband element management system (EMS) that enables telcos to reduce the installation complexity and operational expenses associated with rolling out and maintaining voice, video, and data network devices and services.

PARADIAL
http://www.paradial.com
RealTunnel

RealTunnel is a comprehensive FW/NAT (firewall/network address translation) product and works with virtually any SIP client and any FW/NAT device — including strict corporate firewalls — without requiring FW/NAT or network reconfiguration. RealTunnel consists of RealTunnel clients, RealTunnel Signaling Servers and RealTunnel Media Servers.

Patton-Inalp Networks
http://www.inalp.com
SmartNode SN4630
SmartNode 4600 Series from Patton-Inalp Networks

The SmartNode SN4600 VoIP Gateway/Router Series features three products in one: a business class IP access router; a full feature set including fallback, least cost routing, call distribution, AOC; and VoIP signaling protocols, T.38 fax, and many commonly used voice codecs (G.711, G.723, G.729, G.726). The SN4600 is tailored to address the enterprise and service provider markets by allowing SME customers to connect to an IP/VoIP network.

pbxnsip Inc.
http://www.pbxnsip.com
pbxnsip pbx

The pbxnsip PBX is a feature-rich, SIP-based IP-PBX for Microsoft Windows or Linux operating systems. Features like call recording and paging demonstrate that traditional PBX functionality is available in SIP today. The PBX includes conference rooms, voicemail, auto attendants, hunt groups and agent groups. IVR interaction can be programmed via an external application server.

Performance Technologies
http://www.pt.com
NexusWare Linux-based Software Suite

NexusWare is a family of Linux-based development and software packages enabling users to rapidly develop and deploy value-added capabilities with their solutions. NexusWare Core is the foundation environment for PT's value-add NexusWare software packages. It is a comprehensive, Linux-based development, integration and management environment intended for system engineers using PT's embedded products to build packet-based systems including next-generation wireless and IP telephony systems.

Pingtel Corporation
http://www.pingtel.com
SIP IP PBX Appliance

Pingtel's certified SIP IP PBX line of turnkey appliances for small to medium enterprise, includes utilize Intel servers, AudioCodes gateways, and Red Hat Linux 4.0. Pingtel's SIPxchange, the SIP PBX for Linux, is at the heart of each certified solution, providing enterprise customers with a complete enterprise communications application, including hardware and software support, for about 50 percent less than competing solutions.

Polycom, Inc.
http://www.polycom.com
Polycom ReadiConvene

ReadiConvene is an all-in-one IP-based video and/or voice conferencing bridge that includes integrated conference scheduling, deployment and management applications. ReadiConvene provides an easy-to-order, cost-effective, single-platform solution that is optimized for IP, and includes the features of the Polycom MGC platform, including both on-demand and scheduled conferencing.

PowerDsine
http://www.powerdsine.com
48-port PoE Midspan

PowerDsine's 6548 Power over Ethernet (PoE) 48-port Midspan provides safe power over existing Ethernet cabling to 48 terminals simultaneously, without replacing the existing Ethernet switch. The 6548 PoE Midspan is compliant with the IEEE 802.3af PoE standard, which was ratified in June 2003.

Pronexus
http://www.pronexus.com
VBSALT 1.2

VBSALT is a Rapid Application Development Environment that complements the Microsoft Speech Server (MSS). It allows IT Managers, software programmers and call center developers to rapidly create speech-enabled IVRs, call center solutions, VoIP and high density applications. Developers leverage in-house programming skills, and use familiar debugging tools — all integrated in the Visual Studio environment. In addition, VBSALT handles SALT tags and Jscript transparently to the user, reducing the learning curve for developers.

Psytechnics
http://www.psytechnics.com
PSI

Psytechnics Speech IP Monitor (PSI) monitors the IP bearer of live customer calls and produces a highly accurate quality score based on the ITU Mean Opinion Score (MOS) scale, representing customers' quality perceptions. PSI is integrated into a variety of devices, such as network test equipment, VoIP enabled devices, and network infrastructure, to monitor traffic and provide real-time quality scores.

Pulse Voice Inc.
http://www.pulsevoice.com
pulsescp

pulsescp brings intelligence to a telephone network. Separating database resources and servicing logic from the telephone switch and into *pulsescp* allows delivery of new and

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innovative services to customers by eliminating the need for switch programming. The service creation environment uses a GUI for application development and SQL for business rules implementation. Web-based management permits complete control of service data; media resources offer advanced features such as speech recognition, text-to-speech, voicemail and conferencing.

RADVISION
http://www.radvision.com
Click to Meet for Microsoft Office LCS

Click to Meet enhances Microsoft Office Communicator providing for easier, quicker communications and faster decisions. It enables remote team collaboration with group communications and multi-device reach. The solution extends the communications capability of Communicator enabling users to escalate sessions to multi-party voice, video, and data conferences simply by clicking Communicator's Start Conference button.

RingCentral
http://www.ringcentral.com
RingCentral

RingCentral offers comprehensive voice and fax communication services with Virtual PBX capabilities — at affordable prices. Every plan includes RingCentral's exclusive tools such as RingCentral InternetFax with FaxEditor, Call Controller call management tool, and RingOut click-to-call dialing. Small businesses and busy professionals get capabilities designed to give them greater control, privacy, and freedom.

Sansay
http://www.sansay.com
Sansay SPX

The SPX is an Access Side Border Controller capable of transparently solving the NAT/Firewall issues with SIP. The SPX can act as a full Registrar or as a Registrar Proxy for a SIP Feature Servers. It provides an efficient means to offload the Feature Servers from repetitive registrations, and intelligently switches media for endpoints behind firewalls. The highly available SPX system is manageable via XML and HTTPS with system capacities up to 40,000 subscribers in a 2U rack space.

Seawolf Technologies Inc.
http://www.seawolftech.com
XRainbow Softswitch

XRainbow is a Web-based communication service platform which offers class 5 switch full functionalities for next generation network service providers. Based on SIP and Linux/Apache, XRainbow enables service providers to capitalize on revenue opportunities in the convergence market of voice and data services by accelerating time-to-market and reducing their development and operational cost.

SecureLogix
http://www.securelogix.com
ETM (Enterprise Telephony Management) System

The solution enables secure, optimized, and efficiently managed enterprise voice networks. It hosts a suite of integrated telecom applications that protect network resources from telephony-based attack and abuse, and simplify voice network management. The System application suite includes a voice firewall and intrusion prevention system. These security solutions are integrated with powerful management capabilities to monitor voice network performance and audit service use and utilization.

ShoreTel
http://www.shoretel.com
ShoreTel6

The ShoreTel 6 features Office Anywhere, which supports mobile users irrespective of their location and the device they are using at the moment. ShoreTel's management and integration capabilities get even better with integrated software distribution, media encryption, on-net dialing, and increased support of international operations. ShoreTel has also delivered two new telephone devices, a low-end IP phone and a 24-button programmable button box for operators and assistants.

Shunra Software
http://www.shunra.com
Shunra Virtual Enterprise 3.5

Shunra Virtual Enterprise is a powerful, flexible hardware and software solution that creates an exact model of any production network environment. With Shunra Virtual Enterprise network managers can test the functionality, performance, scalability, and robustness of their VoIP platform, equipment, and network infrastructure under current and future real-world network conditions before rollout.

SIPquest
http://www.sipquest.com
SIPquest Mobile Console

The SIPquest Mobile Console is a dual mode soft client for mobile phones and PDAs that allows a user to place and receive a call using a single number, over the best available network — WiFi or cellular. The Mobile Console offers a uniquely seamless end-user experience with a simple, easy to use unified GUI for both voice and data services. The Mobile Console provides multimedia-rich applications like call control, personalization and presence and it provides service providers a clear evolution path to IMS.

SJ Labs, Inc.
http://www.sjlabs.com
SJphone

"Zero Touch" version of SJphone is intended for Web-based "zero-touch" transactions and can be downloaded and launched with minimal support required. The Premium Version of SJphone is intended for OEMs and ITSPs who require significant support and integration, prefer customized branding and demand an advanced differentiated feature set. Features include video and video conferencing, business telephony feature set, voice and video recording, MS Outlook integration, IM and Presence, neighborhood discovery, and more.

snom technology AG
http://www.snom.com
snom 360 VoIP Business Telephone

The snom 360 is a full-featured, yet surprisingly affordable, multi-purpose VoIP (SIP) executive deskset with 128x64-pixel tilt-able graphic LCD screen, speakerphone and headset jack. Powered from AC or PowerDsign-certified power-over-Ethernet, the snom 360 is easy to install and manage; supporting all advanced NAT/firewall interoperation, self-configuration, and secure remote-management standards.

softroute corporation
http://www.vbuzzer.com
vbuzzer

Vbuzzer is a suite of instant messaging and internet telephony software that allows telephone conversations to travel across Internet and traditional phone networks seamlessly. A simple download installs a softphone and optional messenger. Users can then communicate with each other anytime, any where for as long as they wish. Vbuzzer offers competitive rates for its optional fee-based calling plans.

Spanlink Communications
http://www.spanlink.com
CentralControl

Spanlink CentralControl is a Web-based management framework that unifies and simplifies the disparate resources of a VoIP-based network. CentralControl provides virtual management for multiple administrators that resolves this issue and offers IT/IS staff a new perspective by hiding management complexity, providing audit trails for diagnosis of problems, generating multi-level reports for greater cost management, and enforcing best practices system wide, while maximizing solution up-time.

Sphere Communications Inc.
http://www.spherecom.com
Sphericall IP PBX

Sphere Communications delivers IP PBX technology for Service Oriented Architectures, representing a significant breakthrough in enterprise IP telephony. Sphericall IP PBX is an enterprise-class business software solution designed as an open, mission-critical communications infrastructure. Sphere has opened up its IP PBX to other enterprise-class business applications via Communications Web Services (CWS) whereby the rich functionality of the IP PBX is abstracted away from its underlying complexity.

Spirent Communications
http://www.spirentcom.com
Spirent Converged Network Impairment Emulator

The Spirent Converged Network Impairment Emulator empowers service providers to confidently introduce Triple Play service bundles on real-world IP networks for residential customers. This test solution is designed to bring the network into the test lab where prior to service deployment, service providers can determine and analyze how the system will perform under varying network conditions over time.

SPIRIT DSP
http://www.spiritdsp.com/voip.html
TeamSpirit

TeamSpirit is a multi-point voice conferencing engine based on SPIRIT. The industry-first full-duplex hands-free multipoint VoIP conference is available thanks to TeamSpirit. SPIRIT now actively markets TeamSpirit, the only voice engine alternative to GIPS software on the global market.

Sprint North Supply
http://www.sprintnorthsupply.com
Connection Central

Connection Central is an all-in-one voice and data gateway solution for the home and micro office markets. It is equipped with DSL modem, router, four Fast Ethernet ports, WiFi (802.11b/g) and support for two channels of VoIP. In addition, Connection Central incorporates the features of an advanced PBX, including a maximum of two PSTN CO lines, four wired extension ports and an on-board DECT base station supporting six wireless extension ports. Connection Central provides the user with complete flexibility and mobility.

Strix Systems
http://www.strixsystems.com
Access/One Network Outdoor Wireless System (OWS)

Strix's Access/One Network OWS employs a modular, multi-radio, multi-channel, and multi-RF wireless mesh architecture that provides the throughput and low latency needed to support wireless voice, video, and data applications. The multi-radio and multi-hop capability forms an intelligent high-performance mesh, where traffic is routed on optimal paths and the mesh automatically self-tunes and self-heals as conditions change.

Surf Communication Solutions
http://www.surf-com.com
SurfRider-812

The SurfRider-812 is a PTMC DSP farm providing simultaneous "Triple Play" processing capabilities for developers of voice and video gateways, CTI applications, Remote Access Servers (RAS), and a multitude of other voice, video and data applications. Featuring high port density and Surf's patent-pending Open Framework design, allowing seamless integration of user defined and proprietary algorithms, the SurfRider-812 is the ideal choice for such target applications. It integrates with PCI, CompactPCI (cPCI) and AdvancedTCA (ATCA) carriers.

Switchvox
http://www.switchvox.com
Switchvox SOHO

Switchvox SOHO is a full-featured, easy-to-use IP PBX solution starting at an affordable price of $995. Switchvox is built on open source standards. It works with all SIP compatible hardware and software phones as well as standard analog handsets. Calls can be sent over the Internet to Voice over IP providers worldwide and directly to remote corporate offices, by peering Switchvox systems using the SIP or IAX protocols.

SyChip
http://www.sychip.com
SyVoice VoWLAN 7100

The SyVoice VoWLAN 7100 CSM is a complete end-to-end Voice over Wireless LAN solution that can be quickly integrated into mobile phones to support the emerging dual-mode cellular and local area networks. The VoWLAN 7100 combines a VoIP processor and an 802.11b/g WLAN modem. The module is designed to easily interface with most baseband and application processors on the market.

Symmetricom, Inc.
http://www.ntp-systems.com
SyncServer 250

Symmetricom's high-performance GPS network time server, SyncServer S250 synchronizes clocks on servers for large or expanding networks and is optimized for the ever-demanding high-bandwidth Next-Generation-Network (NGN). SyncServer S250 features three independent 10/100Base-T Ethernet standard ports providing high availability and throughput of hundreds of thousands of network clients while maintaining 10 microsecond NTP timestamp accuracy.

SyncVoice Communications
http://www.voicemanagement.com
VXTracker

VXTracker is developed to tackle the unique management issues hybrid voice networks introduce regarding quality, performance, and compliance. VXTracker gives technicians the ability to analyze existing TDM performance, conduct VoIP pre-network assessments to estimate quality, measure QoS after installation, and provide a "rewindable" audit trail for troubleshooting.

SysMaster Corporation
http://www.sysmaster.com
Norfa Lite

Norfa Lite is a scalable and cost effective billing and call management platform, which enables service providers to quickly deploy IP Centrex/Hosted PBX systems. Norfa Lite utilizes Intelligence-at-the-Edge architecture in which central servers perform billing and call management functions whereas end user devices deliver IP Centrex/PBX functionality. Such architecture allows substantial increase in overall system capacity and reduction in per user costs.

Tadiran Telecom
http://www.tadiranamerica.com
Coral IPx Office

With Tadiran's new Coral IPx Office, enterprises can be assured that voice and data components are designed to work together. The Coral IPx is designed as a VoIP switch with available gateway interfaces for popular telephony interfaces. Other features include native VoIP design (SIP and MGCP), and support for third-party SIP devices (VoIP).

Tekelec
http://www.tekelec.com
Tekelec Fixed Mobile Convergence (FMC)

Tekelec's fixed-mobile convergence (FMC) solution lets carriers and MVNO operators migrate towards IMS while enabling services

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today. The Tekelec FMC solution consists of Tekelec 6000 VoIP Application Server for IP services and session control, Tekelec 3000 Media Gateway Controller and Tekelec 8000 Multimedia Gateway for wireless, wireline and broadband networks, and Tekelec's Wireless Convergence Gateway.

Telchemy Incorporated
http://www.telchemy.com
VQmon/SA-VM IPTV Performance Monitor

VQmon/SA-VM is a non-intrusive Video over IP performance monitoring agent that can be integrated into IP test equipment, network infrastructure products, and customer premise equipment. The product analyzes the performance of IPTV and video conference sessions, supporting a wide range of video codecs and transport protocols. VQmon/SA-VM is highly efficient, requiring only 0.1 MIPS per stream, and is able to monitor many concurrent video streams.

Telco Systems
http://www.telco.com
T-Marc

T-Marc devices provide a view of service delivery across the enterprise and service provider networks with a variety of infrastructure test functions and performance and provisioning monitoring, including latency, loss, jitter, and trace route measurements. T-Marc offers T1/E1 Pseudowire circuit emulation with extremely low latency and a robust feature-set that includes SLA monitoring and enforcement, traffic shaping, and rate limiting.

TeleVoce Inc.
http://www.televoce.com
TeleVoce-Connected VoIP Technology Platform

The TeleVoce solution delivers command and control to end users of PC-based VoIP services by leveraging the PC's power with TeleVoce's Internet telephony-bridging technology. TeleVoce's patented hardware and software provide the ability to connect the PC, now the end user's virtual "telco central office," with the functionality of Internet-based phone services. This provides freedom of choice of both VoIP service provider and advanced telephony capabilities.

TelTel
http://www.teltel.com
TelTel SVNO Program

The SVNO solution includes access to TelTel's platform infrastructure, SIP termination and origination, proven softphone available for customization, and full integration with the provider's existing operation and legacy networks. TelTel's SVNO solution helps service and content providers gain a strong competitive edge by leveraging TelTel's proven expertise and know-how to deliver a complete end-to-end VoIP solution to their customers.

Texas Instruments
http://www.ti.com
TMS320DM642 DSP-based digital media processor

Texas Instruments' (TI) TMS320DM642 DSP-based digital media processor running at 720 MHz, is designed to provide the foundation necessary to bring HD video and audio to the videoconferencing market. With performance of up to 5,760 million instructions per second (MIPS) at a clock rate of 720 MHz, the DM642 device offers cost-effective solutions to high-performance DSP programming challenges.

Toshiba America Information Systems, Digital Solutions Division
http://www.telecom.toshiba.com
Toshiba Video Communication System

Toshiba's Video Communication Solution (VCS) for its Strata CIX family of IP business communication systems provides affordable video communications and collaboration (desktop and applications sharing) capabilities. Using voice extensions, video can seamlessly be added to telephone conversations. Default settings (on, off, communication rate and window size for video) can be configured to meet each user's needs.

Touchstone Technologies, Inc.
http://www.touchstone-inc.com
WinEyeQ

WinEyeQ brings a unique new picture of the relationship of Voice and Video over IP traffic to other data components of the network in true "Triple-Play" fashion. WinEyeQ provides support for popular VoIP and data protocols (HTTP, SMTP, POP3, FTP, RTSP, SNMP, 802.1Q VLan), affording a clear, concise, and intuitive portrait of all of the components your network.

Transera Communications, Inc.
http://www.transerainc.com
Seratel

Transera's Seratel is an on-demand global IP call center platform designed to convert emerging Internet and telecommunications technologies into a solution that provides visibility, control, and quality management for organizations who are redefining their approach to customer care. Seratel leverages open industry standards such as SIP, Web Services and XML, rather than proprietary mechanisms, eliminating complex integration and reducing maintenance overhead.

Trinity Convergence
http://www.trinityconvergence.com
VeriCall Edge 2.0

VeriCall Edge 2.0 provides manufacturers with an embedded Voice + Video over IP software solution that offers all of the media processing, packet handling and call control functionality required to build voice and video capable communication and multimedia devices. VeriCall Edge is highly optimized software, allowing developers to build VoIP endpoints without the need for costly and power hungry external ASICs or DSPs.

TriVium Systems, Inc.
http://www.triviumsys.com
FloristCRM

The first in Trivium's SpecialtyCRM family of products, FloristCRM offers a unique integration of CRM and IP telephony allowing businesses to manage daily activities and capture customer interaction data within a single application. As a single point of access for customer information, Florist CRM provides IP telephony features like screen pops, click-to-dial, and call management with 360-degrees business intelligence.

Ubiquity Software
http://www.ubiquitysoftware.com
Ubiquity Voice Plus

Voice Plus is a pre-built residential voice over broadband solution for service providers who want to deliver IP-based voice and additional multimodal services to their customers. Voice is the main service at this time but the solution will add additional modalities and capabilities above and beyond voice moving forward, including presence, IM, file sharing, video calling, content sharing, and other features. Voice Plus is a beachhead application that will eventually deliver a full set of real-time, multimodal communications and collaboration capabilities using SIP as the underlying protocol.

VegaStream
http://www.vegastream.com
Vega 5048

Designed for enterprises looking to move efficiently and cost-effectively to IP networks, the Vega 5048 gateway connects analog phone systems to an IP network, while providing two additional network-facing analog ports for connection to a telephone line or analog PBX. The Vega 5048 provides 48 ports for connection to standard analog phones, and can act as a customer premises and PSTN gateway in carrier networks; and/or VoIP networking and IP telephony gateways in an enterprise setting.

2005 INTERNET TELEPHONY PRODUCT OF THE YEAR

Veraz Networks, Inc.
http://www.veraznet.com
I-Gate 40000 EDGE

The I-Gate 4000 EDGE media gateway was designed specifically for distributed low-density sites and offers carriers unique capabilities like 10:1 compression ratio and full redundancy. The I-Gate 4000 EDGE is compact, has a total capacity of 496 DSO channels and leverages the same in-house DSP technology as the other I-Gate media gateways.

Visual Networks, Inc.
http://www.visualnetworks.com
Select VoIP

Select VoIP is a software solution that utilizes actual real-time customer application traffic to assess VoIP readiness and, once VoIP is deployed, monitors site-to-site traffic to discover the root cause of VoIP performance problems. Select VoIP is a module of Visual UpTime Select, which provides in-depth, real-time and historical information that enables enterprises to intelligently manage application and network performance and availability across the entire enterprise.

Vodavi Communications Systems, Inc.
http://www.vodavi.com
Vodavi XTS-IP and XTSc-IP

The XTS-IP/XTSc-IP converged telephony solution is ideal for telecommuter, remote and traveling worker applications; the XTS-IP/XTSc-IP offers the ability to extend IP services via a remote IP services gateway, IP soft phones to laptop PCs, or WiFi handsets, all providing maximum business productivity and cost saving advantages when deploying IP communications.

VoIP, Inc.
http://www.voipinc.com
v911

More than the legally mandated minimum, VoiceOne's v911 services go beyond the requirements of the FCC order to help ensure reliability and quality of 911 calls. VoiceOne utilizes their private MPLS network along with their proprietary softswitch, selective routers, and media gateway controllers to provide the redundancy necessary for quality of service for 911 calls over IP.

VoIPshield Systems
http://www.voipshield.com
VoIPaudit

VoIPaudit is a vulnerability assessment solution built specifically for VoIP, enabling organizations to proactively identify and eliminate possible VoIP-specific attacks before they impact IP telephone services. VoIPaudit enables service providers, enterprises and security consultants to identify vulnerabilities across the entire VoIP network and is designed to be used before, during and following VoIP deployment to ensure the overall security of the IP network.

Vonexus Inc.
http://www.vonexus.com
Enterprise Interaction Center (EIC)

Enterprise Interaction Center (EIC) is a standards-based IP PBX designed to give small to medium-sized businesses productivity-enhancing voice applications all converged on the Microsoft platform. EIC includes applications such as SIP-based call processing, unified messaging, workgroup routing, conferencing, etc…, pre-integrated with a comprehensive set of Microsoft Business Solutions, such as MS CRM, Great Plains, Navision, Outlook and Exchange.

Voxeo Corporation
http://www.voxeo.com
Voxeo VoipCenter SIP Platform

Voxeo's VoipCenter SIP Platform delivers standards-based VoIP application creation, integration, and deployment to any enterprise or service provider. The platform comprises the VoipCenter SIP Application Server, SIP Media Server and SIP Fusion Server. The solution is a turnkey offering, with all the features in the SIP Application and SIP Media Servers, plus an optional built-in PSTN-to-SIP VoIP gateway, in one integrated, rack-mount device.

Whaleback Systems
http://www.whalebacksystems.com
Whaleback Systems SMB 1500

Whaleback Systems' SMB 1500 is a business phone solution built from the ground up for broadband. The solution is 100 percent premise-based and software-driven; it doesn't require any equipment purchases; and its all-inclusive, unlimited calling package makes VoIP technology immediately affordable for small and midsized businesses (SMBs).

WildPackets, Inc.
http://www.wildpackets.com
Omni Distributed Network Analysis Platform

By adding real-time expert VoIP analytics to OmniPeek, the software console used to view and control Omni remote engines, OmniPeek Voice offers both distributed network and VoIP analyses. The Omni DNX Engine captures, does expert analysis, and analyzes VoIP data and OmniPeek Voice views, troubleshoots, and performs in-depth analysis of VoIP traffic. Omni is a complete 24/7 VoIP monitoring and troubleshooting solution.

Witness Systems
http://www.witness.com
Witness Systems Impact 360

Witness Systems' Impact 360 brings together a solution for workforce management, quality monitoring/full-time recording, e-learning and performance management. This workforce optimization framework empowers organizations to access customer information — including actual recorded interactions — more quickly, easily and confidently than ever before.

XConnect Global Networks Ltd.
http://www.xconnect.net
XConnect

XConnect is a provider of 'Plug and Peer' VoIP interconnection services, dedicated to connecting disparate IP communications entities. XConnect address all the main issues of bridging the VoIP islands: ENUM-based directory services, multi-protocol interoperability, and security with both bilateral settlement and, through the XConnect Alliance, multi-lateral zero-cost arrangements. It is a neutral and trusted provider, with a global network of VoIP peering points.

Xelor Software
http://www.xelorsoftware.com
XelorRate Service Quality Manager Software

XelorRate is a Quality of Service solution that creates a new category of software known as Service Quality Management. XelorRate unleashes an enterprise network by using SNMP to automatically discover network elements and topology and then configuring these elements according to industry best practices and the network equipment provider's reference manual for QoS. It is a lightweight Linux application for the Red Hat Enterprise operating system.

Zultys Technologies
http://www.zultys.com
WIP 2

The WIP 2 is a WiFi phone designed to address a real need for mobility in IP communications. Combing all of the power and functionality of a desktop IP phone, the WIP 2 is a wireless phone for employees to stay connected and productive. Competitively priced, the WIP 2 offers four hours of talk time and 12 hours of standby time. Based entirely on open standards, the WIP 2 works with any SIP-enabled system, giving companies the flexibility and feature set they need in their modern telephony systems.

IT

IMS: When Will the Hype Become Reality?

In the 1990s the industry hyped "broadband," "the Internet," and "next-generation networks." Around 2000, we hyped 3G. Now it's time to hype IMS, the 3GPP's IP Multimedia Subsystem. So, is this just more hype, or is IMS real? What will it take for IMS to succeed?

Let's start by looking at history a bit more fairly. Most technological and economic development is, in fact, a series of hyped bubbles followed by sober reexamination and, more often than not, success. For example, speculation around railroads was far ahead of its ability to deliver real services and profits but, ultimately, the industry boomed, crashed, consolidated, and then enjoyed an era of dominance until it, too, was dethroned by the next generation of transportation — namely, automobiles and then airplanes.

Our own communications industry has experienced a series of similar hype cycles around broadband, the Internet, dot coms, NGNs, and 3G, leading many to question the next buzzword, IMS. But several factors indicate that the time is right for next-generation IP networks to mature and become the foundation of communications infrastructure for decades to come. Critical broadband technologies are more mature. Service capabilities are in place and being proven in the market and the industry is investing again. Together, all this says that our industry needs to transform exciting — but incomplete — IP technologies into a viable platform for commercial success, which where IMS comes in — and that's why the time is right.

The first thing to keep in mind is that no single technology — IP, NGNs, IMS, dot coms, broadband, and 3G — is sufficient enough on its own to deliver commercial value. Rather, each is a necessary component in a larger economic ecosystem and value chain that delivers services to customers. A delivery network without services or vice versa offers little value. Architectures like NGN or IMS are effectively the network middleware to bring services, transport capability, customers, billing, and ease of use together in a standardized, user-centric, commercially viable manner. Therefore, IMS may be the capstone that holds together the technologies and services that have been developed over the years, as well as the solution that makes them economically viable.

We are now seeing, for the first time, maturity in services, access networks, IP technology, and NGN/IMS specifications. Broadband access is becoming commonplace, reliable, and affordable. IP-based content is becoming rich (although hard to access, charge for, and personalize). Core IP network architectures are maturing. It is worth pointing out that, while IMS is the industry buzzword, it really is a catch phrase for at least three similar, yet distinct, architectures: 3GPP (define - news - alert) has IMS, 3GPP2 has MMD, and Cable Labs has PacketCable 2.0 MultiMedia. Each of these is slightly different, reflecting the unique transport technology, legacy, and market needs of the constituent network operators that support them. Yet 3GPP, 3GPP2, and Cable Labs are all working to harmonize them so they interwork and support similar services. This truly is a breakthrough.

So why is IMS — in its broadest sense — ready for commercial success in the near future? And why IMS, rather than some future standard? Quite simply, the business drivers exist to support it, and practical reality has made IMS the de facto selection in addition to

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By Grant F. Lenahan

being the *technology du jour*.

The business drivers are in place.

IP networks, based on widespread dial-up and broadband Internet connectivity, have demonstrated their viability. Everything from music to blogs to VoIP to chat to news is delivered and consumed via the Internet today — most voraciously by younger generations who will become the adult majority very soon. Broadband IP networks have also demonstrated that they can deliver a mix of news, voice, chat, e-mail, video, music — anything, really — at a very minimal incremental cost per additional service. This means that IP networks, combined with a rich set of services, are essential for any operator to compete effectively in the future. Economists refer to these as economies of scale, and they provide an insurmountable competitive advantage to any operator who can offer them. Thus, economics demands that operators move to multiservice, broadband IP networks. The only questions are: Who provides the networks and Who offers the services?

IMS specifications are largely complete and ready for prime time.

While much of the communications industry was in turmoil during the early part of this decade, the mobile industry — 3GPP, in particular — plowed ahead with its own NGN standards called IMS. IMS was not conceptually different from earlier NGN efforts: It uses accepted IETF (define - news - alert) protocols, like SIP, RADIUS, and Diameter and largely adopts the IT philosophy of service oriented architectures. So, in many ways, it adopts the underlying approaches that both network and IT experts have been advocating for years. What makes IMS unique is that 3GPP, fueled by rapid mobile growth, sufficient money and enthusiasm, and the impending deployment of IP-based 3G, largely completed their task. Thus, the world had a ready-made, well thought through set of standards. It is also clear that many operators see the benefit of a next-generation standard that supports cable, broadband, and wireless access at the same time. IMS is ready, and it offers the additional benefits of a consolidated network architecture (not more technology-specific stovepipes) and support for convergence across those networks. It is attractive to service providers and defvelopers alike.

Still, some will ask, do we need core network intelligence at all? The Internet vision is one of distributed, peer-to-peer and edge-based services, largely without any in-network intelligence. Many believe that "the Internet" operates sufficiently well today without any sort of centralized intelligence, control, or registries. But does it really?

Significant improvements can be made in how well IP networks deliver services, support various business models, and make users' lives easier. IMS offers the components to do just that. The biggest problems facing Internet-based services today are: lack of security, difficulty of use, minimal interoperability, and almost no uniform charging models.

Simple examples include inconsistent passwords scattered across non-trusted parties, weak authentication of users and devices, "islands" of users and directories without inter-domain routing, and charging that mimics retail credit card use — with all the associated baggage. The biggest internal problem facing operators today, however, is their inability to innovate and introduce new services quickly, largely because networks are a series of service-specific, proprietary stovepipes that do not adhere to any sort of SOA conventions, like shared user data or shared charging practices.

In fact, the current telecom business environment may be the most powerful driver for deploying IMS now. Many forces are conspiring to increase competition across segments (cable, voice and data). While cable companies are exerting pressure on voice prices and market share, an even larger threat exists: the "dumb pipe" scenario. A simple IP network is conceived to connect edge users to edge services and, consequently, encourages the hollowing out of the value added by telecom providers of all kinds, as VoIP and information services providers fill that space.

IMS is an excellent tool to help solve both sets of problems. It provides modular value-adding capabilities that can improve the usability of almost any service, whether it is provided entirely

IMS Security

By Nathan Franzmeier

At their most basic level, networks interconnect people and machines and provide for the free and secure flow of information between them. In a perfect world, simple designs based on homogenous access and consistent infrastructure accomplish this. For simple networks in a benign world, there would be no need for security. Unfortunately the real world is not simple and it is by no means benign. Today's networks have evolved from the simple network of Watson and Bell to a very complex web that spans the world. Using these networks, voice, video, and data are exchanged over a variety of competing access technologies and carried to their destinations using a variety of infrastructure technologies. As technology has allowed these networks to continue to evolve and expand, the opportunity to utilize these networks in ever more innovative ways to provide better communications has expanded. As opportunity expands, the temptation to intercept, interrupt, and redirect this communication for a variety of reasons continues to grow with it. This is why network security must exist.

Much of the technology used for security in IMS is the result of what has been learned from the creation and implementation of previous networks. IMS security is implemented at multiple levels. Security is maintained at the access level and at the network infrastructure level for both the signaling sessions and any resulting bearer sessions.

Borrowing from and expanding on what was learned during VoIP deployment in the wireline network, the P-CSCF, I-CSCF and S-CSCF elements provide services for both the access level and the network level. These CSCF elements share a large degree of functionality with the session border controllers in today's VoIP networks and, indeed, many companies are leveraging this to build the CSCF elements for the IMS networks as fixed-mobile convergence begins to become a reality.

The role of these elements is to provide a standards-based mechanism for controlled access to the mobile network, to provide for roaming, and to interface with applications. This includes providing various security functions, such as interfacing to existing HSS systems for authorization, preventing denial of service attacks, providing firewall and spam protection services, as well as providing mechanisms for legal interception while preventing unauthorized interception. Since IMS is an overlay technology, it does not rely on on the underlying IP transport technologies for security. IMS uses SIP for both access and infrastructure signaling, and the security work for IMS parallels the continuing work for SIP security, in general. Unlike existing VoIP networks, which are evolving to require security, the IMS network architecture specifies it at the outset. This "designing in" of security overcomes many of the problems that are beginning to appear in VoIP networks. These issues include unauthorized eavesdropping, theft of service, spoofing of network elements, and, in some cases, purposeful service disruption with the intent of damaging the competition.

As the world moves towards ubiquitous service offerings over converged networks, the technology powering the networks will continue to become more complex. As the complexity increases, the security mechanisms must also continue to evolve to ensure reliability and inspire confidence in the end users. The IMS architecture and the corresponding security mechanisms will ensure that this happens, which will lead to increased opportunities for everyone. IT

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by an operator, or by a third party (such as a BYOB VoIP company, a news service, a music publisher, or a gaming company). IMS specifically offers a rich set of capabilities to provide open, shared data; to provide flexible real-time charging and rating; to authenticate users and devices; to share those credentials, thus reducing password proliferation and increasing security. It also provides the basis of a rapid service delivery environment, both for the development of in-network services and to bond the network to IT-based services and third-party services. In essence, it allows for a more uniform user experience, more flexible charging and economic models, better security, and greater interoperability.

All of this indicates that it is time for IMS to become real. The business drivers are there. The services — and demand for them — are there. The desire for convergence is there. Unlike previous NGN standards, IMS is sufficiently developed for most in the industry to see the value in adopting it intact, rather than re-inventing the wheel.

If the industry implements IMS as a technology — as a way of moving from SS7-based voice to SIP-based voice, then it will fail. But, if the industry sees IMS as a way to add value to a wide range of existing and latent IP-based services, and if the industry sees itself as collaborating with the innovative firms on the Internet (rather than shutting them out), it will usher in a cornucopia of new services and new revenues.

Finally, IMS is a journey. Successful technologies are deployed based on business needs. IMS will not materialize instantly, fully deployed. Nor will it be implemented at a record pace, becoming fully deployed in a year or two. Rather, it will be deployed function by function, as operators use IMS to deliver converged voice services (e.g., WiFi-Cellular roaming), add content-based services more uniformly, and simplify their operations and cut down on systems integration and maintenance costs. Indeed, this sounds less like hype, and more like a solid business rationale. IT

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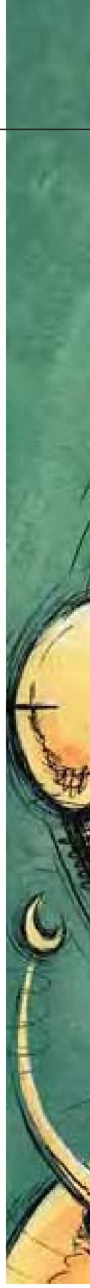
Glossary

3GPP	—	3rd Generation Partnership Project
BYOB	—	Bring Your Own Bandwidth
CSCF	—	Call Session Control Function
P-CSCF	—	Proxy CSCF
I-CSCF	—	Interrogating CSCF
S-CSCF	—	Serving CSCF
HSS	—	Home Subscriber Server
IETF	—	Internet Engineering Task Force
IMS	—	IP Multimedia Subsystem
MMD	—	Multimedia Domain
NGN	—	Next Generation Network
RADIUS	—	Remote Authentication Dial-In User Service
SIP	—	Session Initiation Protocol
SOA	—	Service Oriented Architecture
SS7	—	Signaling System 7



IPTV for Telcos: The Next Frontier

For the consumer, IPTV is the holy grail of entertainment: Imagine watching your favorite broadcasts delivered to your television set whenever you want, regardless of the market you are in, or being able make and take phone calls, or send and receive text messages or voice mails — while watching TV.

This combination of control of media content and access to hip, multimedia technologies is closer than we might imagine. Already, service providers are catering to consumers' appetites for more control of content by offering more personalized services through triple play offerings. Most carriers have the network capacity to handle voice and data, but it's the third component of triple play — video or IPTV — that is expected to be the "killer application" driving the widespread adoption of triple play services. It is also the one that presents the biggest challenge for telcos.

The Battle for the Consumer Begins

Today, service providers operating in the world's most competitive broadband markets are striving toward an ambitious, yet reachable, goal: They all want to offer triple play services to their customers over a single, packetized network infrastructure. Triple play promises to significantly reduce costs by enabling carriers to offer all these services via a single infrastructure. More importantly, telcos are hoping that the addition of IPTV will help replace their fading voice businesses with a growth business that will help them compete more effectively with cable operators for subscribers.

Recent announcements to offer new video on demand (VoD) services from some of the biggest TV networks and cable operators, including Disney, NBC Universal, and Comcast, demonstrate that consumers want — and will spend their dollars on — VoD services. In fact, analysts are predicting that, in the next five years, we will see a tremendous increase in the uptake of VoD services. According to research from analyst firm Informa Media and Telecoms, by the end of the decade, one third of the world's TV viewing households will be using VoD services. It also is predicted that cable companies will benefit most from this trend. This is not good news for the telcos and they are counting on IPTV to give them the edge in the fierce race for consumer spending on bundled services.

The promise of IPTV is that it will enable carriers to deliver TV, video signals, or other multimedia services to the home via a consumer's existing broadband connection. Benefits to the consumer include integrated features, such as the ability to personalize and interact with their video content in new ways. For example, users will be able to listen to their voice mail from their TV directory, or order a pizza from their TV before their HD movie begins.

IPTV is already a multi-billion dollar trend dominating the telecommunications industry. The projections are that $4 to $6 billion will be spent in this market by telcos annually for the next five years. Telcos know IPTV has the potential to give them the advantage in the race against cable operators; however, to deliver IPTV services, their networks must be upgraded to support this new world of on demand, integrated communications.

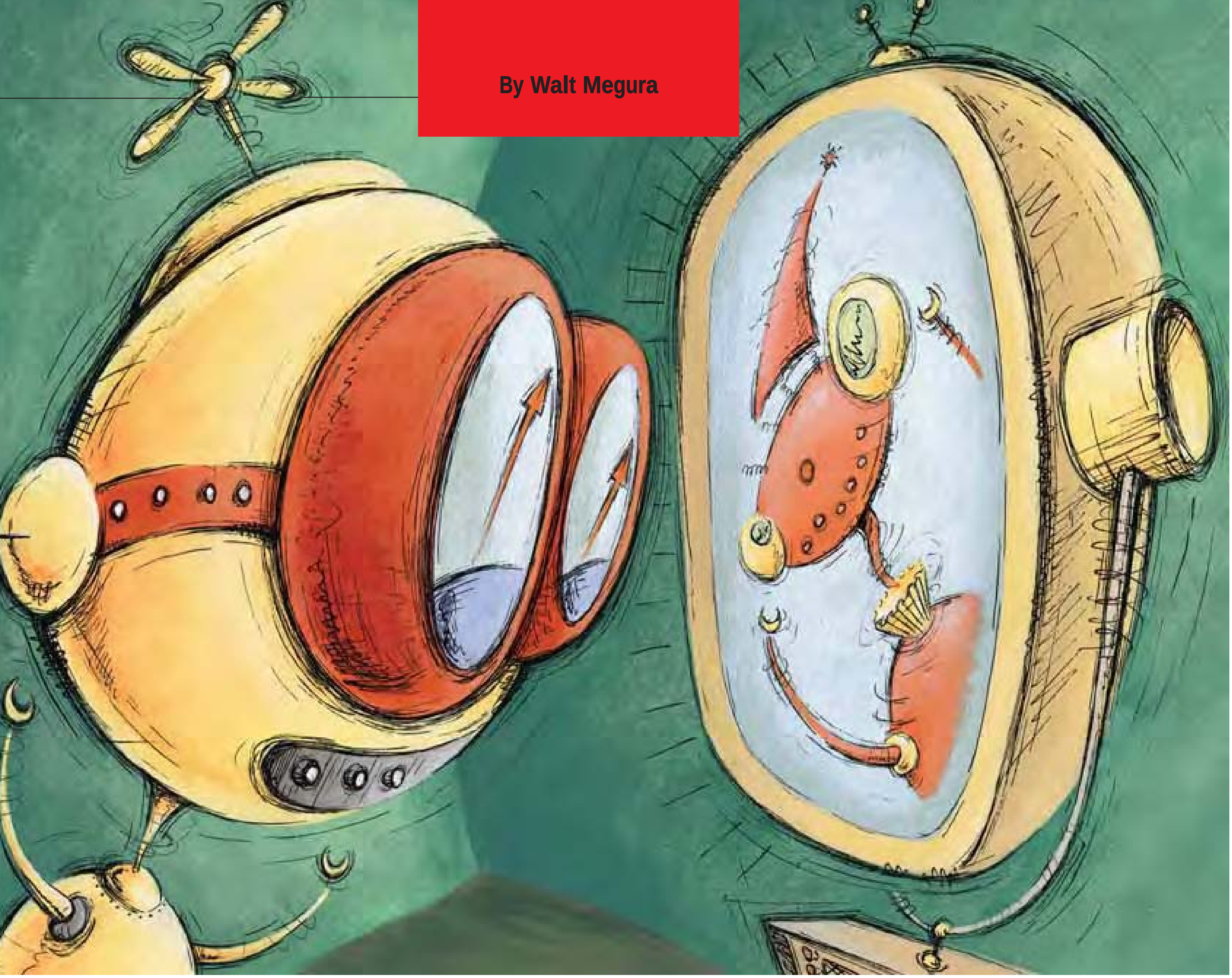
IPTV Components

With so many telcos and cable operators vying for position in this hot sector of telecom, and with IPTV-mania sweeping the networking landscape, there is much confusion about how to deliver these kinds of revolutionary new services. Many vendors are talking about a complex set of systems that will cost billions of dollars to acquire. However, the overall solution can be cost-effectively subdivided into four key categories:

- Video Head end,
- Network Backbone,
- Access Network,
- In-Home Network.

The Video Head End

The IPTV head end is where content (such as television channels or movies) is received and prepared for transmission across the operator's private IP network.



By Walt Megura

Perhaps the most complex piece is simply capturing this content, as it can come from analog satellites, digital satellites, or antennas, and can be standard definition, high definition, or music. Once formatted, the operator will add local commercials and FCC-mandated emergency alert information. The Conditional Access (CA) system then encrypts the signal to prevent service theft or unauthorized copying. Finally, the signal is placed onto the network for delivery to the subscriber. All of this happens under the control of network middleware, which also controls what the electronic program guide (EPG) displays, thereby controlling access to content and services.

The Network Backbone

The network backbone is often overlooked for enabling IPTV services. However, as many of today's provider backbones will not be able to handle the bandwidth required to offer even basic IPTV services, it is the most critical component service providers need to address. Consider a television service of 100 standard definition television (SDTV) channels using MPEG2, which is the existing standard for digital television. Since each channel requires roughly four Mbps of throughput, the backbone must support a total of 400 Mbps. This is certainly within reach for many existing backbones. Video, however, drastically changes the bandwidth requirements in a service provider's network backbone. IPTV essentially requires giving each subscriber a personal television channel of four Mbps. For a network of 10,000 subscribers, this means the backbone must now support 40 Gbps. As IPTV grows in use and popularity, service providers will need to address the need for increased capacity in their network backbones by expanding their optical infrastructure, as well as by building out local video office locations in major metropolitan areas.

The Access Network

The most basic decision about the access network is determining how much of the existing copper or DSL loop can be used. The general guideline for offering IPTV service is the network must support at least 20 Mbps of throughput for the long term viability of a service offering that includes HDTV service. But, whatever the service provider's target for access bandwidth, DSL throughput is highly dependent upon the length of the local

IMS creates an opportunity for IPTV services to follow end users wherever they go.

loop. Often fiber must be run to the outside plant to reduce the length of the copper run.

For new construction or overbuilding, fiber is often deployed right to the subscriber's premises, since laying fiber costs about the same as running copper. In this case, Passive Optical Networks (PON) are most commonly used to support the desired throughput. In denser areas requiring higher bandwidth, Ethernet may be a better solution.

In-Home Network

This is where the magic happens for consumers. But, before consumers get to watch their favorite TV show on demand, telcos must uncover a cost-effective way to distribute the new IPTV signal to their TV sets. First, each consumer site requires a demarcation device, such as a DSL modem or Optical Network Unit (ONU).

There are many ways to connect this demarcation device to the set-top box, but Ethernet-capable wiring is the most common. However, Ethernet cannot run on most existing in-home telephone wiring systems, so telcos must run new wires and, typically, absorb this extra cost. Third-party devices, however, will allow Ethernet to run on existing coax cables and enable the re-use of existing television wiring.

There are also many features available on set-top boxes. The lowest cost equipment supports standard definition television services. Higher-end units support high definition television, integrated hard disks for recording programs, digital audio outputs for connecting to the home audio system, Web browsers, USB ports, and many other options. Offering one set-top box is not sufficient for all subscribers. On the other hand, offering too many alternatives drives up testing and support costs, so a balance is necessary.

Evolving To IMS

A key component to the access network for IPTV is IP Multimedia Subsystem (IMS) capabilities. Though the first goal for service providers is to provide advanced services to the home, their ultimate goal is to be able to deliver those same services to end users on any device, across any broadband connection, wireless or wireline, wherever they happen to be. To realize true service and application ubiquity, though, today's disparate access networks must be linked together. Even with the convergence of voice, video, and data services, management systems and subscriber data remain segregated and service-specific. By deploying an IMS network architecture, service providers will be able to link these disconnected facets.

IMS provides a common SIP session control mechanism, enabling subscriber registration from home or visited access networks, as well as a centralized subscriber database that securely manages subscriber profiles and provides a single point for user authentication. By decoupling the subscriber from specific access networks and specific device types, and by consolidating subscriber management and authentication/authorization, IMS creates an opportunity for IPTV services and applications to follow end users wherever they go.

In addition to separating services from the underlying network infrastructure, IMS makes it possible for service providers to implement many more new applications much faster, easier, and more cost-effectively. For example, because they are unhooked from networks and devices, service providers can merge multiple applications into exciting new combined services. With IMS, service providers will be able to offer increased content flexibility and targeted content. End users will be able to create and control access to video content, and play or broadcast stored programs (personal videos, pictures, music, and recorded video) no matter where they are located.

As IPTV and IMS (define - news - alert) mature, IPTV will become even more personalized and better suited for each end user's specific preferences, interests, and favorite methods of use. Service providers will be able to deliver — and, just as importantly, bill for — voice, data, and video services that flow back and forth across previously separate networks, which will be joined by IMS on the control plane. IPTV features, functionality, and applications will be portable across multiple devices and networks. And like many other communications devices, the TV set will be multifunctional.

Challenges & Opportunities

The move to triple play, IPTV, and, eventually, converged IMS networks will dominate all aspects of the telco market place for years to come. The same is true for cable companies vying for competitive advantage and, in time, even wireless service providers. Their very existence depends on the adoption of these on demand services. All the major telco equipment vendors and many new startups are aggressively moving to provide solutions to deliver on the market opportunities created by this demand.

Delivering video is by far the single biggest challenge, due to its complexity and strains on the network. The current telco network is inadequate for delivering video — they were originally built for voice, then augmented to support data. But, these network and associated operational challenges present an opportunity for a range of equipment vendors. However, as consumers come to expect more of the on demand world, it will be interesting to watch who will be the first to successfully deliver IPTV services to consumers and, once all is said and done, admire the players left standing. IT

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IPTV Trends

The most obvious drivers for deployment of IPTV make it clear that it is far more than the latest fad. IPTV gives users a choice. IPTV gives users control over content alternatives. And IPTV allows content producers the ability to bring truly interactive programming into a subscriber's living room. Sports fans can get the statistics they want on their favorite player during a live broadcast or select which camera angle they want to watch. Drama fans can pick from more than one ending to their favorite program. Hot spotting from creative advertisers would tease program viewers to click on an item in order to view a short video clip on that item while the main program is on pause.

Another dramatic trend is the shift from voice-centric revenue to broadband IP-centric revenue among the global service providers. They all have finally seen the light at the end of the tunnel and it is rapidly nearing. With less than 25 percent of household revenue allocated to wireline voice and 45 percent (and growing) now devoted to video and broadband, there will be a fierce battle between Telco and cable industry operators to capture the largest pieces of consumer business.

Video services are a natural progression for broadband providers of any type and the cornerstone of their triple play service offering of voice, data, and video.

In their quest to pull themselves from the commodity voice business, telcos have one major hope: IPTV (Internet Protocol Television). The technology promises to transform television and, in the process, re-energize telcos by reducing churn and introducing new revenue streams. Yet, IPTV is as uncertain as it is exciting, and making it work will be most challenging.

With the potential for IPTV to increase per consumer revenue by nearly 50 percent, the market is in a rush to develop and deploy the technology through profitable business strategies, improved broadband networks, evolving access technologies, and innovative delivery devices, while focusing on achieving and sustaining a solid ROI.

IPTV is expected to grow at a fast pace in the coming years, as broadband is now available to more than 100 million households worldwide. Many of the world's major telecommunications providers are exploring IPTV as a new revenue opportunity from their existing markets and as a defensive measure against encroachment from more conventional cable television services.

The convergence of broadband and broadcast delivery will inevitably disrupt existing telecommunications and television industries as they collide and collapse conventional boundaries. The network television revolution will change channels of distribution and fundamentally affect the way television is viewed in the future. Billions are being bet on the outcome that could radically transform the media landscape forever.

The television we know today, with a few hundred channels, will soon seem archaic. Viewers will eventually be able to choose from thousands of programs and channels from around the world, much as do with streaming audio already. The network will, in effect, become an infinitely expandable personal video recorder. Viewers will truly enjoy the interactive programming and other benefits from the convergence of Internet and TV.

IPTV is a big step forward when combined with the latest advancements in picture quality, such as high definition (HDTV) and new AVC (Advanced Video Compression) standards like as MPEG4, H.264, or WM9. Not only are set-top boxes becoming smarter, but they will also interact with other devices, such as PDAs, mobile phones, and the Internet to provide a truly flexible solution, allowing local information to be tailored to specific regions and personal preferences.

Success in the IPTV services market requires understanding of the core value chain, including how content flows from creation to consumption and making the right infrastructure decisions to support growth and competition. DRM (Digital

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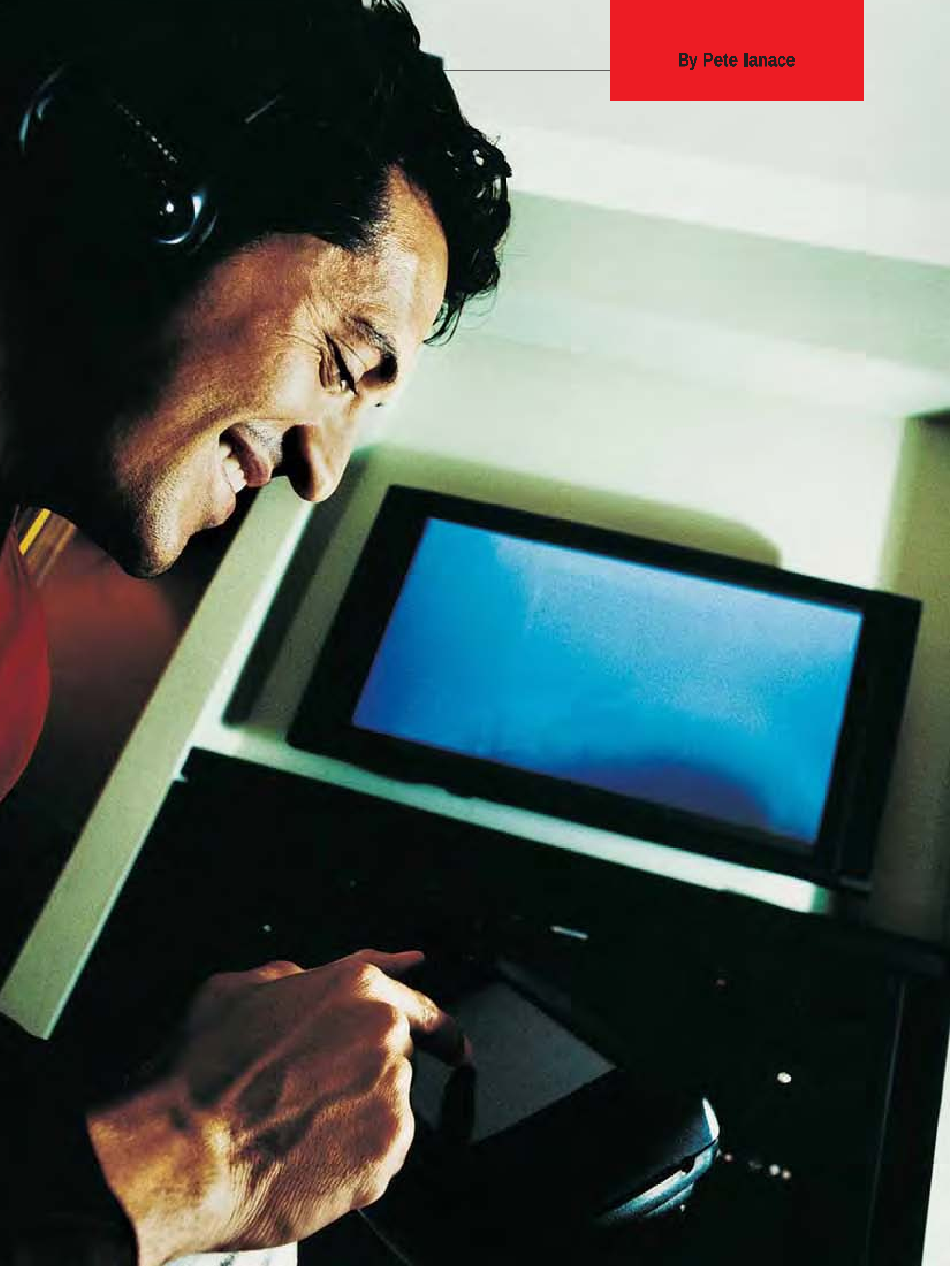
By Pete Ianace



Rights Management) is the single largest challenge. The ability to secure the content is paramount in convincing content providers to stream their most valuable content over IP networks.

The network is the essential enabler of a competitive IPTV deployment. The pay TV market in the United States is reaching saturation, so IPTV providers need to capture market share through more effective differentiation. Compelling content packaging, on demand, interactive applications, and, ultimately, integration of video and video devices into the digital home are more than the goals: They are the requirements.

What will be IPTV's biggest competitor?

Internet Television will be IPTV's biggest competitor for our attention. Internet Television enables anyone to offer a service or TV channel and empowers video producers and programmers to build broadband businesses. Internet Television also gives viewers more choices and control over their use of video and television. This approach gives the content producer and the viewer much greater control over what gets published and what gets viewed. The content publisher is able to directly reach the consumers on multiple devices independent of any specific carrier or operator. Internet Television also aims to be as device independent as possible. Thanks to open standards and formats, which have helped create this opportunity, Internet Television wants to be just as the Web is today, accessible from any computer and IP connection in the world, and not physically tied to the user living room or set-top box.

Internet Television unites the visual impact of television with the dynamic interactivity and measurability of the Internet. This grassroots approach is an outgrowth of the existing Internet user's experiences. People want flexibility, convenience, and as many choices as possible. Internet Television is open to any rights holder, no matter whether it

A Guide to Entering the IPTV Universe

By Mitch Auster

In the coming years, more and more telephone companies will begin offering IP-based television service, or IPTV, to round out their triple play offering. There are several theories as to how successful this will be. Some believe it will be a huge money pit that won't turn a profit; others feel the technology has such an advantage over the cable MSO's platform that success is a slam-dunk. In reality, several independent telephone companies and municipalities have already proven that it is possible to operate a profitable IPTV/Triple Play franchise.

Certainly, it is not just a matter of "if you build it, they will come." There is no service functionality associated with IPTV that the MSOs can't provide. It may be more efficient to send everything via IP to the home and may, therefore, be a logical architecture for a greenfield deployment. When and if the efficiency gain is called for, cable can start sending video over the DOCSIS (Data Over Cable Service Interface Specification)-based IP pipe alongside VoIP and data and the game is essentially even. So, it comes as no surprise that the levers are revenue, CAPEX, and OPEX.

On the revenue front, given the fact that the platform is based on IP, the potential application set is limitless with the ability to integrate telephony, chat, polling, click-to-purchase, and a wealth of other interactive functions into the TV experience. The ultimate keys to revenue generation, besides the economy, will be the level of creativity and the relative brand strength — as evidenced, for example, by the success that Google is experiencing.

In polls taken over the past few years, telephone companies have been cited as the preferred provider for a bundle of services that includes video. For the telcos to be successful, they'll need to be mindful of several network-related considerations in order to maintain their brand, contain costs, and consistently generate revenue. These include scalability, reliability, service quality, and openness.

Today, most people watching TV are tuned into a broadcast channel either watching the program "live" or having it captured to their recording device. Using IP-layer multicast as well as physical layer broadcast, a video stream appears at most only once on any network segment from the point of origination to anywhere in the network — even if it is viewed or recorded by many subscribers. Video on Demand (VoD) requires a unique video stream for each subscriber watching a particular program. Until recently, this consisted of movies, a set of programs from premium channels, and some free niche programming content. This resulted in a peak simultaneous usage of about 15 percent. In the future, it is expected that almost all of the most popular channels and programs will be available on demand. As they are broadcast, they will be ingested into a server and ready for on demand viewing within five seconds and stored for, perhaps, one to two weeks. Simultaneous usage could grow to 50 percent or more.

For bandwidth scalability, the transport network should be based on 40-wavelength DWDM with reconfigurability at the wavelength and subwavelength level to rapidly add capacity in 1 Gbps or 10 Gbps increments to the central office and remote terminal locations where it is needed. For service scalability, the access node should perform IP forwarding. This serves to hide

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all the Ethernet MAC addresses of all the subtending subscriber devices from the rest of the network. Instead of the aggregation switches and routers needing to know and store the 250–2,000 or so MAC addresses per access node, they only need to know the MAC address of the access node itself. This avoids having to partition the network and allows it to scale almost indefinitely.

While there are no E911-type regulatory requirements for video, some might consider its full time availability to be a life or death matter. Nobody has ever died because the TV went out, but I'd run for cover if it happened during the fourth quarter of the Super Bowl or the decisive final episode of American Idol.

To ensure that video is always available, the operator should employ fault-tolerant servers and redundant broadcast feeds, as well as a robust distribution architecture between the service edge router (SER) or broadband remote access server (B-RAS) at the video serving office and the access node at the wire center or remote terminal. Per-user/service packet treatment should be employed at the SER/B-RAS as well as the access node. This will ensure that each subscriber and application receives the contracted quality of service without adversely impacting any other subscriber.

Transport should be based on carrier-grade Ethernet aggregation configured in a logical hub-and-spoke design over a physical ring. At layer 2, the end-to-end system only requires class-based queuing, but must support at least two strict priority packet queues, so that both VoIP and video have their own, and are transmitted immediately. Other data services should have separate queues to ensure an appropriate quality of service without impacting the more sensitive VoIP and video traffic. In conjunction with VoIP and video being managed by a connection admission control function, so as to avoid oversubscription, this will ensure that those packets aren't dropped or delayed.

By utilizing transport nodes with integrated Ethernet switching arranged in a logical hub-and-spoke over physical ring architecture, the aggregation network is drastically simplified as compared to routers or MPLS/VPLS switches interconnected in a Layer 2 or 3 ring. It enables protection at Layer 1 with restoration in less than 15 milliseconds compared to several seconds when done at higher layers. This design also enables Layer 1 drop-and-continue of the broadcast traffic to all nodes on the ring. Carrier-grade Ethernet leverages the ITU-T Optical Transport Network standards to provide the robustness of SONET on the cost-curve of Ethernet.

To minimize capital expense and maximize service flexibility, the network must be built using open standards. It may seem expedient to choose a single supplier to provide an integrated solution when first deploying IPTV, but that comes a degree of risk. Cable operators have been locked into a duopoly of suppliers for the video head end to set-top box due to proprietary systems. Again, it has been demonstrated by several independent telcos that it is possible to build a network using best-of-breed components from several vendors. At a minimum, a service provider should test that a second vendor can plug into every portion of the network to ensure the system is based on open standards. This will result in price-value competition and unrestrained innovation.

As telephone companies deploy video, it is apparent that they are at the very beginning of what is sure to be a continually evolving triple play network. As such, it is imperative that the network be simple to scale and operate, uncomplicated so as to ensure robustness and service quality, and open to change. IT

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is an individual creating a video for a very small audience or a traditional publisher offering linear cable channels.

As the explosion of blogging has shown, many of us want our moment of fame through content creation. The ability to create any type of video content and post it for distribution will give us all more options than we will have time to view. The ultimate goal is for each user to have the ability to build their own custom channel.

Internet TV also is ideal for marketing and distribution; it facilitates a distributed and collaborative environment for media production. The Internet is capable of counting how often content is viewed and by whom — This is an advertiser's dream. But it challenges advertisers to come up with new, creative, and entertaining ads that capture the viewer's attention, since Internet TV gives the user the ultimate choice on what they watch.

Internet Television is able to ride on existing infrastructure including broadband, ADSL, WiFi, cable, and satellite. In addition it doesn't tie the user to a specific service provider.

If the user wants to use the TV as the viewing device, there will be numerous ways to accomplish that, thanks to the hoard of new devices that will make it easy to connect the Internet TV channel of choice to the TV set in your living room.

The real question is what option will the user select? If nothing else, the fact that Internet Television will be an option will force IPTV to put its best foot forward and we, the users, will be the ultimate winners. IT

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Measuring Quality for IP-Based Voice and Video

Can broadband service providers afford to ignore customer experience metrics?

There is little doubt that the abundance of choice for consumers regarding voice and video services is due largely to the deployment and success of IP-based packet networks, access technology improvements, and deregulated telecom infrastructure. We are on the brink of a digital revolution in the way people communicate and consume media over fixed and mobile networks. Convergence of services has meant the Triple Play: the combined Broadband Internet, VoIP, and Digital TV are springing up from a variety of service providers as competition for the consumer heats up. There are also a variety of different consumer devices, including PCs, PDAs, and mobile phones that are now capable of receiving and sending live video images along with audio.

Consumers are aware of picture video quality, lip synching, and overall quality within audio and multimedia applications; so how will service providers assess the customers' perception of the quality of voice and video applications? Rating the customer experience from a quality perspective is a new way to look at an old problem: Can service providers objectively measure and predict a customer's perceived quality of experience of what they see and hear with new IP-based services like VoIP and IPTV?

Subjective Measurement

The well-known tagline, "Can you hear me now?" underscores the importance of voice quality and the role of subjective testing in today's service providers networks.

Service providers perform subjective testing for speech quality by selecting a panel of ordinary consumers and then subjecting each of these people to a set of voice or video samples. The consumers are then asked to evaluate the quality of voice or video calls under different testing environments and scenarios by rating each sample call in terms of a Mean Opinion Score (MOS) from 1 to 5. This subjective test data is collected and analyzed to help service providers rate the performance of real-world conditions that consumers would experience in terms of service quality. Service providers and equipment vendors continue to use subjective testing process methodologies to validate — or contest — their perceived service or product quality.

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By Nav Chander

Subjective testing, however, is time consuming, expensive, and must be very carefully designed and executed. It also requires a specially designed testing facility.

Objective Measurement

Objective measurement methods for voice and video quality assessment represent a cost-effective alternative to subjective tests. Such methods can analyze the performance of impairments and network parameters and produce a set of MOS scores that correlate well with available subjective data — without requiring a panel of human subjects. There are two main classes of objective measurement: intrusive (active) and non-intrusive (passive). Such measurements are repeatable, efficient, and fast.

Active testing techniques inject sample voice or video signals into a network so they can be captured and assessed at another point in the network allowing comparison of the degraded voice or video to the reference sample.

The world standard for intrusive assessment of end-to-end speech quality is the perceptual evaluation of speech quality (PESQ) model, ITU-T Recommendation P.862.

Non-intrusive techniques monitor live network traffic to determine the perceived quality. As a result, network capacity is not lost to testing and the service provider is assessing the actual performance experienced by customers. Benefits of non-intrusive techniques include:

- allowing wider scale and denser testing;
- not losing valuable network capacity for test calls;
- not requiring access to end points for test signal injection;
- providing measurement data directly reflecting actual customer use.

Objective Measurement Methods for VoIP

Core 3G networks and next-generation networks will be IP packet-based. Objective measurements of voice quality in VoIP (define - news - alert) can be assessed with measurement methods that can accurately predict packet network performance from packet statistics and analyze payload to detect echo, delay, levels, and voice quality.

Simple objective measurements of network statistics, such as packet loss, delay, or jitter in a typical VoIP network

The service provider's future success will hinge on the quality of the voice or audio visual experience.

provide some measurement of network QoS, but cannot alone predict consumer experience.

There are numerous examples across many technologies but, by way of example, consider a VoIP network with two percent packet loss. In the first case, the packet loss is evenly distributed in time and the packet stream is converted back to speech by an edge device with effective error correction. In this case the user experiences two percent packet loss and high voice quality. In the second case, the packet loss is bursty and the packet stream is converted back to speech by an edge device with poor error correction. In this case, the user is subject to two percent packet loss and low speech quality.

Objectively predicting MOS scores using bulk assumptions about the network and averaged performance data, such as certain methods derived from the E-Model, simply cannot reflect the perceived performance for an individual session or a real consumer experience.

Objective customer-centric quality assessment requires a complementary set of voice, audio, and video measurement methods. These cover active (known test signal) and passive (live traffic) testing requirements. Passive methods can also determine the separate performance of the IP-bearer and IP-payload. For example, in a typical VoIP network carrying a voice service, the IP-bearer measurement determines if the packet transport network service is affected. It uses a sophisticated mapping of packet and jitter temporal distribution to determine if packet loss and jitter is degrading the MOS performance of the bearer.

IP payload measurement analyzes the packet payload to measure echo, delay, speech level, noise level, and the speech quality of the payload. This analysis can be applied to a sample of the call traffic. Sampling is sufficient, since it is used to diagnose slowly varying network problems such as defective Customer Premise Equipment (levels, acoustic echo path, speech distortion, etc.), under-provisioning of echo cancellers, and quality issues.

This alternative, and more robust, VoIP monitoring measurement method provides a powerful and unique combination of IP-bearer and payload analysis. The combination of measurements, in addition to packet loss, jitter, delay, speech level, noise level, or echo, provides a comprehensive representation of customer experience and allows diagnostics and root cause analysis of service-affecting faults. This type of objective measurement provides accurate statistics that genuinely reflect customer experience.

The transient and potentially erratic nature of IP data packets means that the service provider's future success will hinge on the quality of the voice or audio visual experience it delivers. And while it would be tempting to rely on conventional diagnostic measures to ensure that service levels remain high, the ultimate definition of success will depend on something rather more subjective — the experience of the viewer or caller sitting in an armchair watching an IPTV stream or having a conversation on their VoIP phone. What is needed is an automated system capable of reporting back to the carrier with data that accurately reflects human perception.

Customer Experience Metric

Today, Operation Support Systems (OSS) that manage service assurance on service providers' packet networks have been designed to handle data services rather than real-time video and multimedia services, where packet loss and delay have a much more disruptive effect on the end user's experience.

In IPTV, VoIP, and other real-time triple play services, quality of service can only truly be measured if it relates to the end user's experience. All other data becomes a string of meaningless figures unless the customer is satisfied with an excellent audio or visual experience, which is highly subjective, hinging on human perception.

Objective measurement metrics that predict customer experience are invaluable planning and monitoring tools for service providers that are concerned with the introduction and penetration of newer IP-based voice and video services in their markets. Accurate correlation to a wide range of subjective testing results provides essential validation and credibility to using these metrics.

Ultimately, the end user will not tolerate anything that negatively impacts the quality of their communications, whether voice, data, or video. Carriers need to ensure that the customer's high expectations for quality are met. IT

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Getting products ready for market is not easy. To do it right, you need a Q/A staff with up-to-date skills and the latest test equipment. That's where CT Labs comes in. A respected testing authority, CT Labs specializes in real-world testing, test automation, and Q/A services for a wide range of converged communications products.

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A New Energy for the Contact Center: $E = MP^2$

New Multimedia VoIP Contact Center Solutions Provide Innovative Enterprise Power

Energy is a powerful force, as Albert Einstein discovered 100 years ago when revolutionizing the world of science with his breakthrough formula, $E = MC^2$. This same force, with the advent of new Voice over Internet Protocol (VoIP) capabilities, can now be applied to enterprise customer relationship management (CRM) solutions.

With feature-rich VoIP contact center suites, enterprises can now gain tremendous new sources of energy by extending — at the speed of light — voice, multimedia presence, collaboration, and productivity enhancements throughout the enterprise. Such next generation suites go vastly beyond the province of call centers. They are helping to empower all areas of the enterprise to deliver first contact resolutions, improved organizational effectiveness, and superior levels of customer satisfaction. As an added bonus, many of the next generation VoIP contact center solutions are very easy to use and deploy.

$E = MP^2$ is the new customer relationship management model that organizations must embrace to remain competitive in the future:

- "*E*" for Enterprise-wide Energy Engagement
- "*M*" for Multimedia VoIP
- "*P*" for Productivity and empowerment and, finally, the second
- "*P*" for Presence-enhanced collaboration.

Here's how the formula works for enterprises with advanced VoIP (define - news - alert) contact center solutions:

E: Enterprisewide Energy Engagement

Customers have higher expectations. The speed and pace of life and business has many customers unsatisfied and unwilling to accept company delays in resolving issues or providing answers. In the past, an enterprise would not suffer from statements to customers like:

- "*That's not a question this department can answer, call this new number.*"
- "*Call us back later. I don't know where the expert is or when she'll be back.*"
- "*I really don't have the authority, so I'm transferring you to our Chicago office. When someone answers, tell them your issue or leave a voice mail.*"

Today, if an enterprise is responding consistently these ways, it is undoubtedly losing money, losing customers, and losing its competitive position.

Many industries are establishing benchmarks and measurements for first contact resolution (FCR), moving away from the old model of average speed of answer (ASA). These businesses are coming to the realization that a key to improving FCR is the engagement of departments and company areas outside of the call center to support and respond to customer requests. Engaging these enterprise areas into the customer contact model — in real time — is critical to reaching resolution on the first contact. These areas can be as close as subject mater experts who are tightly associated with the call center, or as far away as departments located in other cities and states focused on associated service areas such as billing, collections, downstream processing, or shipping. The concept is to reduce the layers of time, distance, and resources required to

By Al Baker

meet customer issues and eliminate the need for multiple contacts for the resolution of one issue.

New VoIP contact center solutions are enabling companies to empower all areas of the enterprise to assist in delivering first contact resolution and providing a path for the traditional call center to easily transition into a multimedia contact center.

M: Multimedia VoIP:

Many companies are now adding VoIP into parts of the call center infrastructure. The clearest value proposition for VoIP in the call center has been in extending the call center functionality seamlessly to remote agents and agents working at home. This has allowed companies the flexibility to hire and retain resources needed for the call center. Moving into the future, the next competitive level for VoIP goes beyond voice to embrace multimedia functionalities.

Customers want media options (e-mail, Web chat, and voice, for example) to reach businesses. At the same time, they want the same simplicity and service treatments (like gold card or platinum memberships) that they now receive with voice. Historically, multimedia options for call centers have been bulky adjuncts requiring separate systems, routing strategies, reporting and administration. For the enterprise, this meant extensive integration and support costs. Customers, attempting to use new media options, were left in uncharted waters and they often lost preferred status treatments.

The next generation of multimedia VoIP contact center suite solutions can deliver — anywhere in the enterprise — inbound voice, Web, chat, e-mail, and outbound voice customer contacts. At the same time, these solutions can utilize the same routing, reporting, CRM strategy, and administration tools across all media. Competitively, this allows enterprises to deploy customer segmentation strategies and treatments consistently across all media. Because these are packaged suites, customization and professional services is greatly reduced, if not eliminated altogether. The extended enterprise also benefits from multimedia options, allowing voice, e-mails, and chat sessions to be extended throughout the enterprise and still be monitored, managed, and reported on. This can provide a new level of flexibility for increasing first contact resolution (FCR).

Multimedia VoIP solutions can provide innovative power to enterprises.

P: Productivity and emPowerment:

Contact center agents are being asked to shoulder more and more of the daily interaction with customers. For many companies, especially those that are driving more sales through the Internet, a customer's interaction with a contact center agent or CSR is often the first contact he has with a real person in the company. Also, for many vertical markets, the "agents" may be highly paid professionals — doctors, nurses, engineers, private bankers, or real estate agents. Today, perhaps more than ever before, enterprises must assume the truth of the old adage that first impressions make for lasting memories.

Like never before, an agent's actions will communicate to customers the quality of the company. A positive interaction with a company's contact center, service, or sales representative represents an important opportunity to create a positive perception of the company. Equally and sometimes exponentially worse, a negative interaction with a company's call center creates a negative perception of the company. All agents, including the new extension to white collar agents, expect and need tools and applications that enhance the company's aura of professionalism in the eyes of customers. All of these areas are benefiting greatly from the multimedia VoIP suites.

Today's contact center applications must also coordinate customer data from multiple sources to provide real-time information about overall call center performance (SLAs) as well as individual agent performance (MBOs). Next generation multimedia VoIP contact center suite solutions are also delivering on this. Embedded in these suites, enterprises need IP-based desktop tools with built-in customer data screen pops, real-time agent personal productivity data, connectivity to corporate LDAP directories, streaming desktop "ticker tapes" showing contact center activity, historical trend data, and real-time linkage to other enterprise applications and scripts. Enterprises should choose multimedia clients that are easy to use and simple to personalize.

P: Presence-enhanced Collaboration:

Contact center agents are the first responders for engaging customers as they enter the enterprise. Often, these first responders need the assistance of others. From a customer's point of view, this hand off must be handled quickly and efficiently or the customer's urgent need is considered unmet. New multimedia VoIP contact center solutions should come with presence and collaboration features built-in and include specific desktop applications for agents and for employees who work beyond the initial line of response in the contact center.

Such intra-enterprise collaborations are the "moments of truth" for the contact center and the enterprise, in regards to reaching their first contact resolution (FCR) objective. Having desktop tools that provide real-time presence information about extended enterprise workers is critical. Who is available now, by department or skill set? Which media are they available on, regardless of location? Presence-enabled collaboration answers all these questions in real time and allows the agent, via a single click, to communicate and share data by VoIP and the enterprise WAN. This is real power that creates new customer response energy. Companies throughout the world are using this new paradigm to develop new competitive positions in customer relationship management.

Innovative VoIP Contact Center Solutions

Multimedia VoIP contact center solutions can provide innovative power to enterprises. Traditional call center environments and many vertical markets — healthcare and insurance, to name two — are reaching beyond the traditional options provided to agents in the past.

Insurance companies are using presence and collaboration tools to empower contact center agents to collaborate with more seasoned colleagues and other knowledge workers to solve customer issues on first response. In the healthcare sector, incoming calls frequently require expertise that goes beyond a call center's capabilities. Several large healthcare enterprises with multiple hospitals has empowered its contact center agents with presence and collaboration tools so that agents can make one-click contact with triage nurses to initiate immediate consultations. The hospitals, as a result, have vastly improved patient service.

One cable entertainment and broadband service provider estimates yearly productivity gains to be about $300,000 thanks to presence-enhanced collaboration tools that link calls center agents with others in the organization via a virtual caller queue display. The company said its average call handling time has been reduced by half.

Conclusion

In response to how he came up with a number of scientific revelations, Einstein said that the true sign of intelligence is not knowledge but imagination. "We cannot solve our problems with the same thinking we used when we created them," he noted.

The enterprise deployment of new multimedia VoIP contact center solutions opens the door for quantum leaps in CRM. A new source of customer response energy is waiting to be tapped for enterprises savvy enough to empower all of its people resources to drive first call resolutions. Einstein would be proud. IT

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Best Practices for Successful VoIP Adoption

In the private sector, companies strive to use technology as a competitive advantage. In the public sector, organizations endeavor to use technology to deliver myriad services to a vast array of locations in the face of shrinking budgets — to "do more with less." In both instances, telephony systems are critical to efficient ongoing operations.

However, both private and public sector IT and telecom staffs are plagued by numerous issues with current TDM (define - news - alert) telephony services, including:

• Redundancy in deploying services for new users, voice mailboxes, and contact center agents in various locations;

• Delays in implementing moves, adds, changes, and deletes (MACD), especially during periods of growth or consolidation;

• Difficulty in delivering uniform services across multiple locations;

• Requirement for specialized system management expertise to ensure implementation of best practices.

To address these limitations, as well as deliver high-quality voice service, obtain superior functionality, and gain cost advantages, many organizations have decided to transition from TDM-based systems to VoIP-based IP communications systems.

During this transition, organizations often overlook the importance of IP communications system management and prioritize other issues, including security, quality of service (QoS), reduction of latency and jitter, and role changes driven by voice and data convergence. Since IP communications offers a new network infrastructure, enterprises and public sector institutions have an opportunity to elevate IP communications management as an integral element in IP telephony deployments. Doing so allows them to positively transform internal processes for voice service delivery throughout the organization.

Proper Planning to Maximize Benefits

As part of an IP communications telephony system's architecture, in which most or all applications are centralized at a main location or in primary facilities, VoIP services can be delivered to virtually all users regardless of their physical location. This environment adds significant new capabilities to IP communications, but it also changes the management requirements.

As a result, IT and telecom managers have an opportunity to institute a VoIP system management framework that:

• Provides management control to local users and administrators;

• Allows deployment of configuration changes to all servers and server clusters;

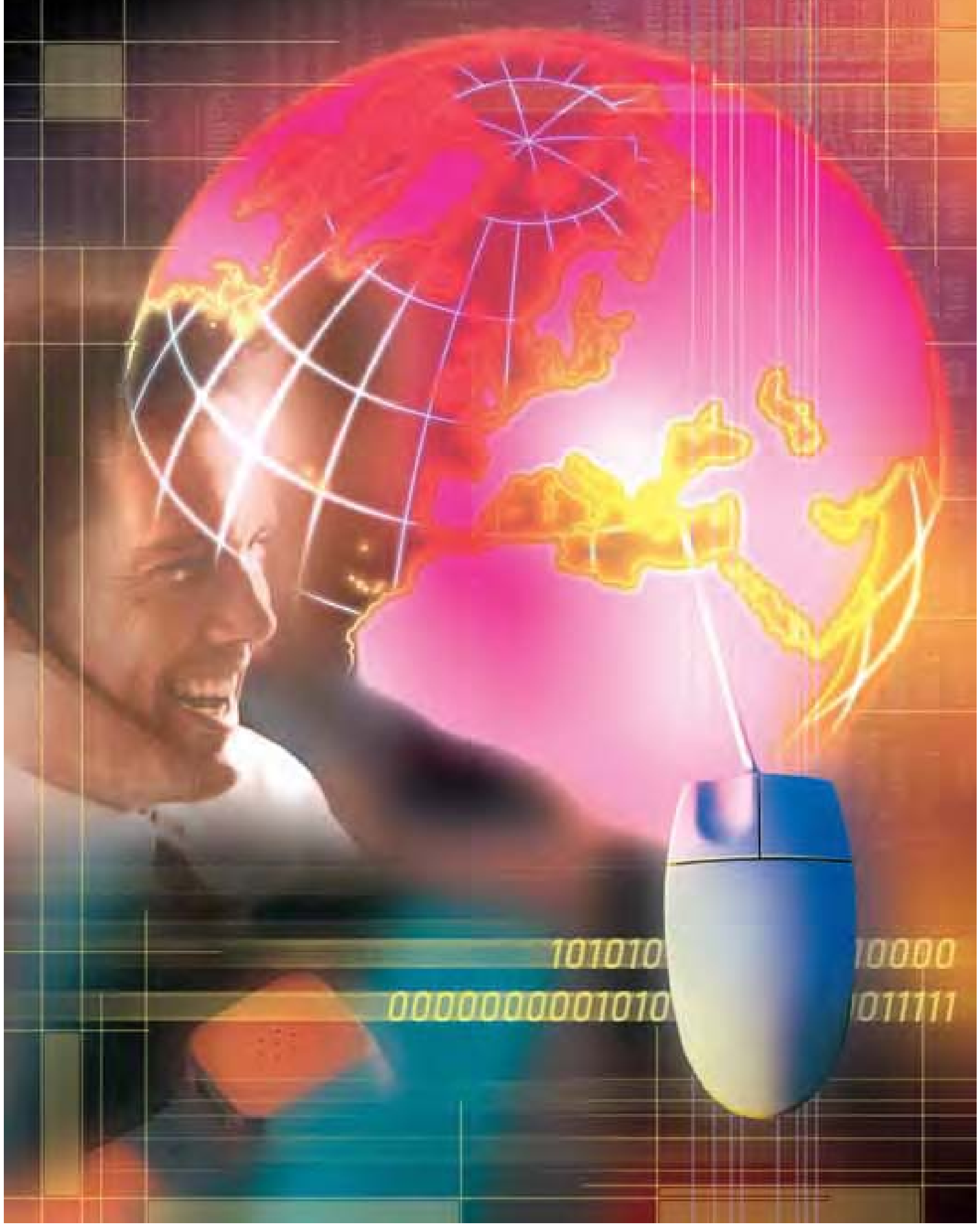
• Supports the expanding need for enterprise information (key metrics).

System management architectures must expand administrative and service capacity by using business rules to enable intelligent delivery of user services and by allowing common administration of distributed VoIP system components. This must be addressed in the pre-planning stages, not after the systems are installed. Companies that plan for system management are better positioned to build it right the first time, controlling configuration consistency across all offices, enabling scalablity of their system without adding complexity, and, ultimately, lowering the cost of ownership.

Essential VoIP Management Solution Elements

A key objective for a VoIP network is to deliver superior services while improving efficiency. An IP Communications system management environment that controls both user and system functions can easily accomplish this goal. Its structure should include the following components:

• *Multi-tenancy* — Virtual partitioning of a VoIP network to group users based on department, group, individual, By Tom Sullivan



and location.

• *An application resource and user manager* — An engine based on configurable business rules to administer and manage all IP communications functional elements, such as ACDs, call managers, IVRs, and unified collaboration systems, as well as all user-specific parameters and information.

• *A unified dashboard* — A centralized interface to integrate separate applications for multi-level administration and provide information to users regarding IP communications call metrics and quality monitoring data.

By introducing these elements, the right IP communications system management framework allows organizations to transform how they provide and manage voice services. Specifically, they save IT resources by incorporating a service provider-like environment through multi-tenancy. They also enforce best practices by providing well defined scripts and rules for configuration changes, eliminating redundancies and they provide critical metrics to drive resource management decisions.

Multi-Tenancy: Enabling Local Administration Across the Enterprise

IP Communications systems can provide a centralized system architecture in which servers and clusters of servers deliver services to multiple locations, including headquarters buildings, branches, remote sites, and home offices. This approach allows system capacity to be shared across all locations, eliminating the capacity issues of TDM systems, in which excess capacity may be available to only certain locations. By enhancing call resource availability, IP communications systems simplify capacity planning and service delivery. In a VoIP environment, users and groups of users can act as "subscribers" to services delivered by a centralized system. However, to enable users and groups of users with the same and similar service attributes to autonomously administer their needs, IP communications networks need multi-tenancy.

Multi-tenancy allows IT staff to partition groups of users with common requirements and manage the groups securely and independently. For system management purposes, each partitioned group can be treated as a subscriber or a group of subscribers.

Multi-tenancy gives groups and users day-to-day control, while reserving higher level control for IT staff without requiring extensive training on VoIP application configuration rules. With this functionality, senior level administrators can concentrate on enterprise level issues, allowing local administrators and users to oversee minor administrative details. As a result, partitioned users can perform moves, adds, changes, and deletes, select desired services within allowable limits, and obtain call usage reports.

Application Resource and User Management to Enforce Business Rules and Best Practices

With multi-level administration through multi-tenancy, IT managers need to ensure that local administrators and users follow uniform system practices — naming conventions, dial-plan patterns, and specific rules for making moves, adds, changes, and deletes. To prevent configuration errors and to ensure rapid fault isolation in the case of network problems or failure, IT staff should design these rules based on organizational best practices.

IP communications management needs to simplify and automate implementing system changes across the network. Specifically, it should make change requests trouble free by providing rules for each level of users. In addition, it should automate change requests by maintaining a database of user data and a set of system commands that comply with relevant procedures. It also should track these changes. This is a central part of the IT Infrastructures Library (ITIL) framework for instituting best practices.

As a result, enterprises and public institutions can optimize the use of IP communications services across all users through a mechanism that helps rapidly introduce new services, maximizes VoIP system uptime, and addresses potential problems or isolates faults.

Unified Dashboard for Consolidated Administration and Metrics

IP communications management must provide consistent, multi-level reporting with appropriate partitions based on roles and permissions. It should allow users, group administrators, branch managers, and top level enterprise administrators to securely generate reports that include only the information they are authorized to view. For each level, an IP communications management solution should exclude

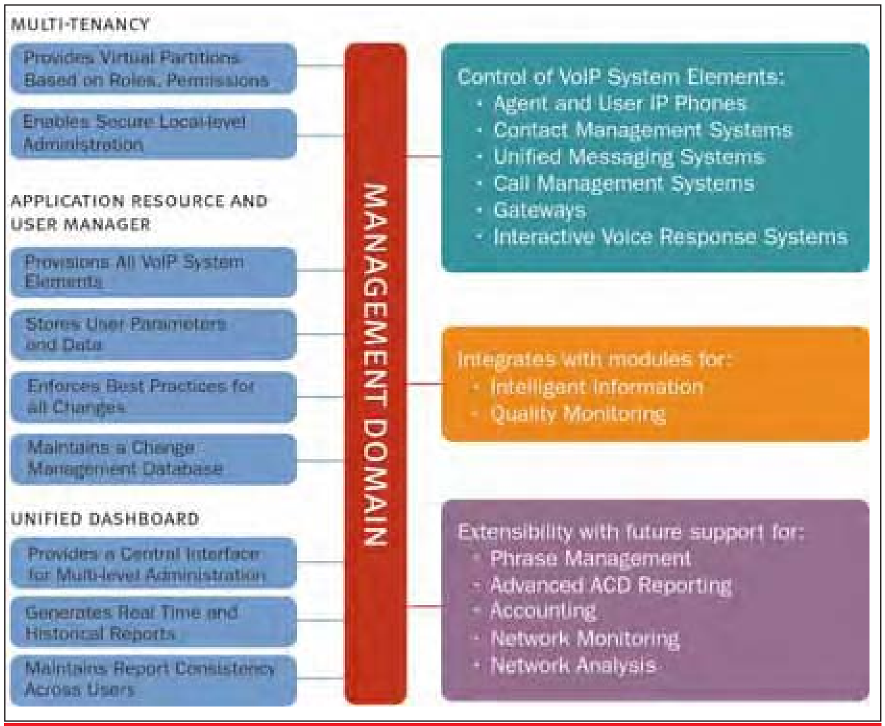


Figure 1. A VoIP management solution simplifies overall delivery of voice service applications, facilitating the transformation of IT and telecom processes by involving users and local administrators control of the VoIP network.

data for higher levels, guaranteeing security by providing a closed reporting structure, where access to broader data is only granted to appropriate users.

A multi-level reporting structure provides IT staff with visibility of critical metrics across locations, users, and groups to help drive informed decisions regarding VoIP resources. Available reports should include:

• *Audit Trail* — Provides a record of every system administration action.

• *Configuration* — Shows the overall telephony plan and configuration of each VoIP system and IP phone.

• *Billing* — Details billing information for each partition, including user, group, and enterprise levels.

• *Contact Center* — Provides real-time and historical data at multi-tenancy levels for call summaries, wait times, and agent utilization.

• *Quality Monitoring* — Delivers network layer information to allow tailoring of the VoIP network based on QoS and jitter characteristics.

In addition, to eliminate issues with report consistency across users and groups, IT staff, groups, and individual users should have consistent reports that reflect like-for-like criteria and fields. Having consistent reports allows IT managers, local administrators, and users to gather valuable information by comparing data across all locations, facilities, and groups. Consistent reports help IT professionals plan and implement new programs or apply resources to other locations as needed. Users can manage their individual environment to coincide with the business requirement of their group, department, division, or location.

The Advantages

An IP communications management solution helps enterprises and public sector organizations simplify the planning, deployment, and administration of VoIP services. By integrating local level administration and user control, delivering critical metrics, and implementing routines that support best practices for configuration and maintenance, VoIP management promises organizations numerous advantages that drive RoI:

• *Greater Reliability* — Implementation of best practices and availability of critical metrics ensures greater reliability of the VoIP system.

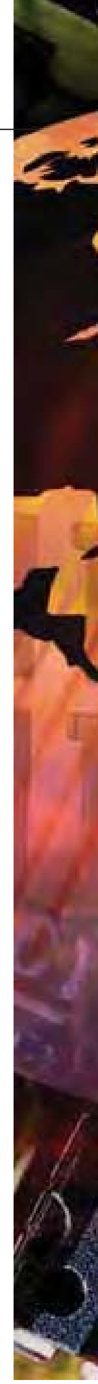
• *Greater Efficiency* — A converged voice and data network enables better capacity planning through a centralized call resource architecture. Disparate system components can be configured through a single interface with multi-tiered administration access and a faster mechanism for MACD.

• *Lower Costs* — Reduces demands on IT staff resources by allowing local administrators and users to manage MACDs and day-to-day administration. A consistent set of rules and templates for performing configurations and change requests minimizes user problems and avoids calls to service desks. Reports can be electronically entered into enterprise accounting systems, avoiding manual re-entry of data, saving time in charging calls back to appropriate departments or groups and helping identify anomalous call patterns.

With the opportunity to transform the process for voice service delivery across the organization, enterprises and public sector institutions can maximize the benefits of a VoIP communications system. They can track cost savings and perform better capacity planning through consistency in reporting of critical metrics across all facilities and departments. Additionally, IT managers can find more time to handle crucial projects, knowing that IP communications management is under control. Through multi-tenancy, users and local administrators are able to efficiently configure VoIP services based on roles, without redundancy or need for specialized training. Finally, IT and telecom managers can appreciate modularity in their IP communications management functionality, allowing them to add and expand new modules as required during the system's life cycle. IT

An IP communications management solution helps enterprises and public sector organizations.

Tom Sullivan is a senior product manager for Spanlink's CentralControl suite of IP communications system management products. For more information, please visit the company online at http://www.spanlink.com. (news - alert)



Interactive Communications — Connecting Employees, Customers and Affiliates Over The Internet

Voice over IP (VoIP) has gone mainstream in recent years in both the consumer and enterprise markets. Consumers are adopting VoIP for economical, broadband-based home-telephone service and for PC-based calling applications. According to company statistics, Vonage boasts an installed base of more than one million lines and usage of more than 35 million calls per week. As of December 2005, more than 220 million PC users around the world have downloaded Skype Internet phone clients, this according to the company's Web site. Google, AOL, MSN and Yahoo! have all announced new or improved PC calling offerings in recent months.

PC calling services such as Skype are used almost exclusively by consumers. Concerned about potential viruses, security threats, and the chatty nature of PC calling protocols, many businesses and institutions prohibit the use of these services by employees and individuals within their enterprises.

But businesses are increasingly deploying VoIP to reduce operation costs and telecommunication carrier expenditures. Industry analysts report IP PBX line shipments have overtaken traditional PBX line shipments, and IP handset sales have outstripped traditional handset sales.

Enterprises are deploying IP PBXs as alternatives to traditional PBXs for office telephone services and to connect remote sites over private IP networks or Virtual Private Networks (VPNs). Some organizations now offer Internet-based voice services to mobile users or telecommuters.

While enterprises have gained economic and administrative benefits by transmitting voice and data over the same physical media, they have not leveraged VoIP to fundamentally advance their business applications or to inherently improve communications with their customers or partners.

A new class of software product is delivering the next logical step in the evolution of converged voice and data services. The Interactive Communication Platform (ICP) integrates voice, video, and other services at the application layer. This enables organizations to add interactive communication services to software applications and to deliver integrated Internet-based voice and video services to customers and affiliates.

ICPs enable enterprises to leverage VoIP not only for cost savings, but also to boost the top line with new interactive business solutions that fundamentally improve customer, partner, and employee communications. In addition, ICPs allow enterprise users to enjoy features found in popular consumer PC calling programs while granting corporate IT staff the added control, security, and reliability of a trusted enterprise solution.

Interactive Communication Platform Overview

An ICP is a next-generation software platform for delivering interactive voice, video, and other communication services within an enterprise or extended community of interest. Unlike conven-

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By Alan Rosenberg

tional circuit-switched PBXs and new IP PBXs that were designed to switch media flows between internal parties, an ICP is engineered to enable internal communications as well as to easily and flexibly extend communication services globally to employees and affiliates over the Internet. Additionally, ICPs can broker communications between external parties such as customers, partners, and suppliers.

In its simplest form, an ICP can be deployed as an economical software-based IP telephony system, either as a PBX adjunct or alternative. ICPs support any SIP phone as well as traditional analog telephone instruments, and support communication over private IP networks, VPNs and the Internet. An ICP provides a core set of voice/video calling features, incumbent PBX and PSTN gateway functions, and supplemental services such as n-way calling and voice messaging.

Enterprises can roll out ICPs today to deliver IP-based voice and communication services to new locations, employees, and applications in cap-and-grow scenarios; to extend corporate voice services to mobile users, telecommuters, or small offices; and to deliver voice services to customers and affiliates.

ICPs can also offer open APIs to integrate voice and video services with Web pages and various business applications. ICPs meld interactive services with Service-Oriented Architectures (SOAs), allowing voice and video services to be invoked as a program call that enables enterprises to fundamentally extend their use of voice, video, and other communication services.

Interactive Communication Platform Applications

ICPs are suitable for a variety of applications in virtually any industry or market segment.

Click-to-Talk Applications. Enterprises can utilize ICPs to integrate voice or video services with Web sites; back-office call centers and CRM applications; and other business applications. Businesses can improve customer experiences by routing calls to subject-matter experts or by diverting priority customers to dedicated service representatives. Flow-through APIs link call agents to customer database records so routed customers are not forced to repeat account information or transaction data.

Peer-to-Peer Communications. Businesses can install ICPs to deliver brokered communication services for clients and affiliates. Peer-to-peer communications enables customers to communicate directly with authorized partners or with other customers to exchange information or conduct business transactions. While the enterprise controls and initiates the session, the communication flows directly between the participants — not over the enterprise network. Transporting media streams across the Internet provides massive scale advantages and minimizes enterprise network capacity and re-engineering requirements.

Internet Calling Services. Enterprises can leverage ICPs to deliver a controlled and trusted Internet-based phone service for constituencies and extended communities of interest. Organizations can offer basic PC calling services plus premium services such as public telephone network connectivity; personalized concierge or information services; or enhanced feature packs (voicemail, n-way calling, call forwarding, call transfer, call waiting, and so on). Internet-based users can enjoy features found in popular consumer PC calling programs with the added control and reliability of a trusted enterprise solution.

Peer-to-peer communications enables customers to communicate directly with authorized partners or with other customers.

Traditional PBX or IP PBX versus Next-Generation ICP

Traditional PBX or IP PBX	Next-Generation ICP
Centralized monolithic architecture — designed for intra-enterprise communications	Distributed modular architecture — adaptable to intra-enterprise and peer-to-peer communications
Proprietary APIs — aimed at conventional applications such as IVR and unified messaging	Services-oriented APIs — adds voice/video services to Web sites and other applications
Hardware-based solution — costly, vendor lock-ins	Software-based solution — exploits general purpose computing platform economics and performance trends. Provides flexible deployment options.
Proprietary phone instruments — limited choices, expensive	Standards-based SIP phones — choice of instruments to meet feature, form, and cost needs; well suited for click-to-talk applications
Proprietary subscriber management — redundant adds/moves/changes	Standards-based subscriber management — unified adds/moves/changes via RADIUS, LDAP or Active Directory.

Conclusion

Interactive Communication Platforms enable organizations to deliver Internet-based voice and video services to employees, customers, and affiliates. They allow enterprise users to enjoy features found in popular consumer PC calling programs with the added control and reliability of a trusted enterprise solution. ICPs deliver a new echelon of voice/data convergence, empowering businesses to increase customer satisfaction and revenue by fundamentally improving customer communications. Finally, by melding interactive services with SOAs, ICPs will change the very nature by which enterprises deploy and utilize voice, video, and other communication services. IT

Alan Rosenberg is director of Product Management at BlueNote Networks. For more information, please visit the company online at http://www.bluenotenetworks.com. (news - alert)

2006: The Year of VoIP Peering

VoIP peering has become recognized as the catalyst towards the true IP communications revolution where calls are end-to-end IP. As Rich Tehrani, publisher of *Internet Telephony*®, has proclaimed, 2006 will be the Year of VoIP Peering. This year is when we can state with confidence that VoIP peering industry has finally migrated to the mainstream with the associated technology, infrastructure, services, and solutions reaching a level of sophistication and robustness that provide the foundation to this revolution.

As often happens with new technology, there is some confusion about what VoIP peering actually is. Simply put, VoIP peering is the complete bypass of the legacy Public Switch Telephony Network (PSTN). The VoIP Peering motto is "Keep Calls on Net."

While conventional wisdom would suggest that calls that start on IP and finish on IP should remain on IP, surprisingly, this is not the case. With the proliferation of VoIP on various networks, many of which compete with each other for subscribers, interconnecting would mean sleeping with the enemy.

The vast majority of all Voice over Broadband service providers today exchange call traffic with each other only via the global telephony network. For example, calls originating from Tokyo via SoftBank's Yahoo! BB Phone, and terminating on Telio take the tortuous journey from IP to PSTN (define - news - alert) via a PSTN Gateway, travel over the PSTN (incurring all the usual PSTN charges, and limited to PSTN-only functionality), are translated back via another gateway from the PSTN to IP, and are finally delivered to Mom in Norway.

There are two principal drawbacks of utilizing the PSTN for transiting these calls. First, every PSTN call incurs a per minute charge due to legacy PSTN settlement fees and accounting charges. Second, all the richness of IP communications is lost by connecting via the lowest common denominator, the PSTN.

VoIP peering allows a service provider to offer "on-net" calls, thereby reducing the transit cost to near zero and delivering IP rich multimedia communications services to not only their own customers on their own IP service, but to any other provider's customer. Thus, the number of calls a provider can provide on-net is a direct function of the number of other providers with whom it peers.

Bridging the hundreds of islands of voice over broadband service providers to ensure that multimodal communications remain entirely in the IP universe is a necessity for the VoIP industry. End-to-end IP communications enables the full spectrum of benefits of high-fidelity codecs, encryption, video telephony, and widespread presence.

Independent providers that wish to peer together can either create and manage multiple — even hundreds of — bilateral peering relationships (technically, operationally, and commercially), or can have one relationship with a federated VoIP peering service provider. This service would typically deliver:

- Federated ENUM-based Number Directory system with added security, privacy, and provisioning capabilities together with policy management;
- Advanced call signaling management;
- Security features, including Spam over Internet Telephony (SPIT) prevention;
- Support for various types of commercial agreements between the VoIP parties.

While VoIP peering requires underlying IP connectivity (i.e., Layers 1, 2, 3), either public Internet or private meeting points, and connectivity, this is IP peering and not VoIP peering. VoIP peering is, in the OSI model, from layers 5

By Eli Katz

onwards..

VoIP peering is a vital step in the fundamental migration from an all-PSTN communications world to an all-IP communications world. A recent iLocus report stated that 14.5 million customers (as of June '05) worldwide are using a VoIP phone line, with service being provided by 300 providers. Analysts predict that, by 2008, that number will grow to more than 200 million customers being served by thousands of ITSPs. This will represent, in many countries, 15 to 20 percent migration from PSTN to VoIP.

A number of significant events in the VoIP peering space have driven this process in the past few months. The key foundations of rapid deployment of an IP technology are usually:

- Global standards
- Market demand
- Available service and products based on the standards.

E-mail is a classic example. The POP3 and SMTP standard was defined at the IETF, e-mail software based on these standards was developed, and, finally, an increasing demand by businesses and consumers for e-mail services has led to the almost ubiquitous, interoperable, and interconnected nature of e-mail today. VoIP peering, despite being in its embryonic phase, now has all three elements in place.

Standards

Many of the key standards for VoIP peering, such as the basic signaling protocols — SIP and H323, and ENUM for numbering and addressing — have been around for many years. In the last six months, major developments have occurred in ENUM. In addition, a new IETF working group — Session PEERing for Multimedia INTerconnect (SPEERMINT) — is in the process of being chartered, focused exclusively on VoIP peering.

Also in the IETF world, Spam over Internet Telephony (SPIT) is recognized as a serious potential nuisance and already there are proposals for its prevention.

Market Demand

In November 2005, CableLabs issued

VoIP peering is vital in the migration from all-PSTN to all-IP.

an RFI for VoIP peering with U.S. cable operators as well as with other service providers. The MSOs, with their millions of VoIP customers, have collectively more digital voice subscribers than the rest of the sector combined, and they are rapidly gaining additional market share. VoIP telephony is a key strategic component of cable operators' future business models, both for inducing customer retention and as a revenue generator.

In fact, on a global level, CableLabs was pipped to the post by an initiative called 'Sip-Exchange', which was announced in October. The consortium of cable operators (UPC, Essent, Casema, Multikabel, and CaiW) in the Netherlands, where cable penetration is almost 95 percent, announced the intention to set up a VoIP peering system to enable all the Dutch cable operators' VoIP traffic to bypass the PSTN and remain on-net.

Likewise, a growing number of in-country ITSPs globally have demonstrated a clear desire to peer and respond to the positioning of global brands like Skype, Yahoo!, MSN, and Google. These initiatives by the leading VoIP service providers demonstrate that the demand for VoIP peering has arrived and is being recognized worldwide.

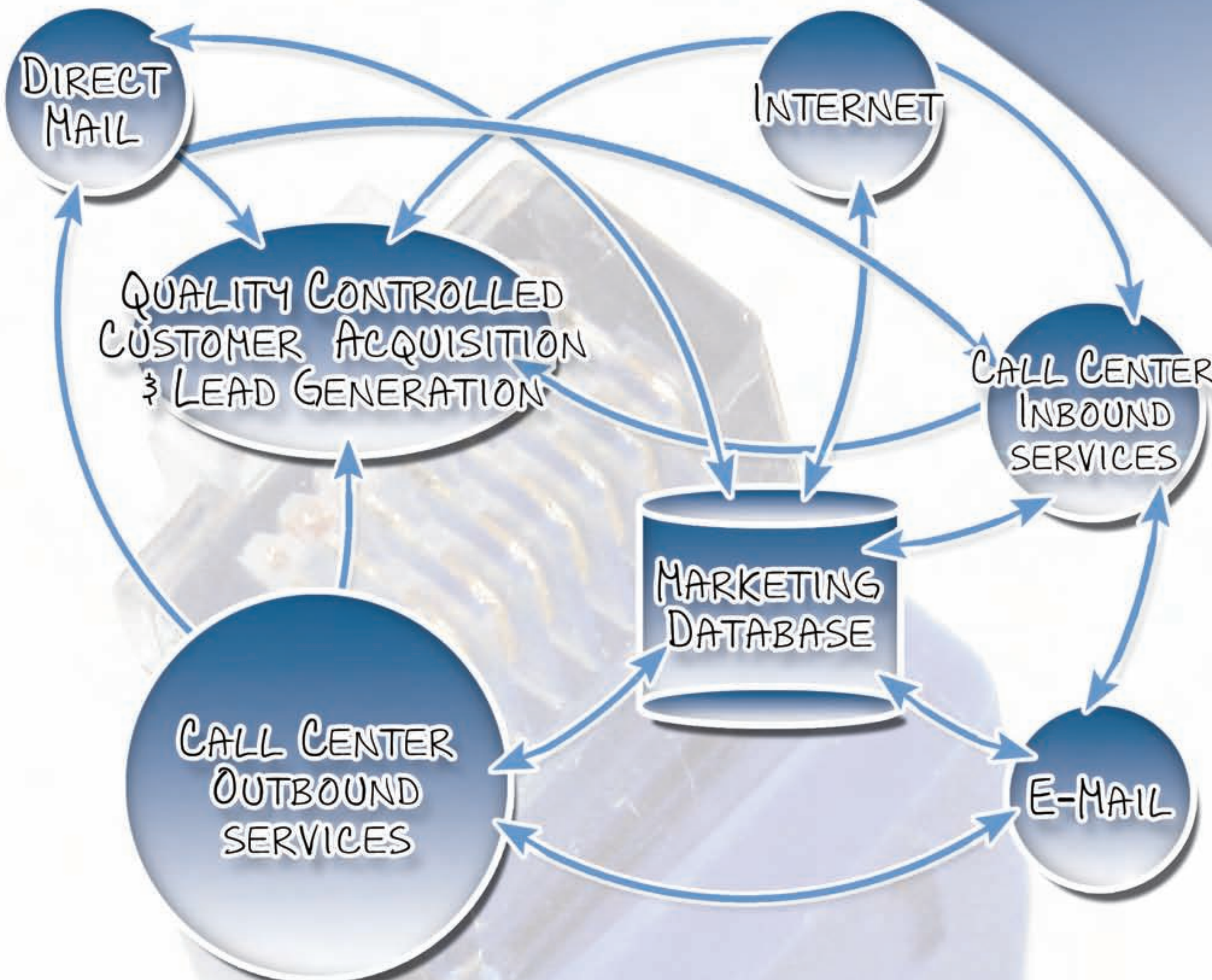
Services and Products

In the past year, companies like as Neustar, Verisign, XConnect, and e164.info have launched VoIP peering services for the global market, covering various elements of VoIP peering. The new year will undoubtedly bring significant research and development, largely focused on security, identity, provisioning, and global architecture.

Driven by new research, costumer needs, and product developments, VoIP peering is certainly ready to take off in 2006. Indeed, it would only be appropriate to label the coming year, "VoIP Peering — Ready for Mainstream Launch." IT

Eli Katz is Founder and CEO of XConnect a provider of 'Plug and Peer' VoIP interconnection services dedicated to connecting IP communications providers and by-passing the legacy PSTN. For more information, please visit the company online at http://www.xconnect.com. (news - alert)

Customer Acquisition & Lead Generation



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What Telecom Equipment Makers Need to Know About ATCA, AdvancedMC, and MicroTCA

Telecom is rapidly evolving as service providers build out their broadband packet networks to offer enhanced wireless, data, video, and VoIP services. Most TEMs have historically built virtually everything in-house. As deregulation has made the service and equipment landscape increasingly competitive, delivering home-grown equipment in a timely, cost-effective fashion has become much more of a challenge.

By utilizing open platforms like AdvancedTCA, AdvancedMC, and MicroTCA, suppliers to TEMs can now offer a wide range of system level hardware and software that makes outsourcing both convenient and cost effective.

What is ATCA?

Defined in January 2003, AdvancedTCA is an open architecture framework designed for building high performance, high density, high availability, rack mountable telecom shelves. ATCA defines the mechanical form factor, electrical interfaces, switched fabric configurations, transport protocols, and system management interfaces for the ATCA chassis, ATCA plug-in cards, and shelf management controllers.

Because they utilize multiple independent point-to-point serial links for blade-to-blade communications, TEMs prefer switched fabrics over general purpose parallel buses. These point-to-point connections increase availability by making the overall system less vulnerable to individual blades or link failures. They're also more scalable, making it easier for TEMs to squeeze extra bandwidth out of existing platforms.

ATCA's hot-swappable switched fabric provides a peak throughout of 10 Gbit/sec per link and supports a full mesh interconnect, which maximizes availability by enabling each blade to simultaneously communicate with every other blade via dedicated channels. The ATCA switched fabric is also protocol agnostic, enabling it to support multiple packet-oriented protocols, including Ethernet, Infiniband, PCI Express, and Rapid I/O.

In addition to its high speed fabric, ATCA provides a number of other features that are critical for TEMs. Its large form factor (8U) and high power capability (200W per blade) give it the capacity to support complex functions and high density configurations. And its redundant fabric, redundant power, and hot-swappability reduce susceptibility to point failures and enable individual blades to be serviced and upgraded without disrupting overall service.

One of the greatest contributors to overall CAPEX (define - news - alert) and OPEX savings in ATCA systems is ATCA's redundant Intelligent Platform Management Interface (IPMI) system control framework, which facilitates active monitoring of and control over individual ATCA blades. This capability is especially important for high-density systems utilizing large numbers of high-performance processors, where thermal control and power management are major concerns.

IPMI utilizes an I2C-based physical interface known as the Intelligent Peripheral Management Bus (IPMB) to link chassis management with board-level FRUs (field replaceable units). IPMI can be used to monitor physical system health characteristics such as voltages, fan speeds, temperatures, and power supply status. It can also be used for automatic event notification and remote shutdown/restart. This information simplifies system design by enabling TEMs to monitor, test, and diagnose systems at the blade level during the development phase. It also enhances availability by enabling IPMI technicians to isolate problems faster with a finer degree of granularity, thereby reducing mean time to replacement

By Stuart Jamieson

(MTTR).

Modularity & Flexibility

AdvancedMC is a field replaceable mezzanine interface that enhances ATCA flexibility by extending ATCA's high bandwidth, multi-protocol interface to individual hot-swappable modules. The resulting fabric gives TEMs a versatile platform for building modular telecom systems that can be outsourced, designed, manufactured, stocked and spared at lower cost. Modular ATCA/AMC fabrics also reduce operating costs by enabling service providers to scale, upgrade, provision, and service their systems with a finer degree of granularity.

AdvancedMC provides a high speed, protocol-agnostic, serial packet interface with up to 21 I/O channels, each supporting data transfer rates of 10 Gbit/sec per channel. AdvancedMC modules are hot-swappable, enabling service providers to replace individual modules in the field without taking entire ATCA blades off line. They offer high power handling capability (currently up to 60W per module), which enables TEMs to implement complex functions at the module level. They also provide an IPMI interface, which enables shelf management to monitor and control individual modules residing on ATCA blades.

ATCA carriers can be equipped with up to eight AdvancedMC modules, which currently come in four sizes: half-height single-width, half-height double-width, and a full-height version of both. The modules have escalating power limits of 20W for the smallest module to currently 60W for the largest module. This mechanical flexibility enables designers to partition their blades for maximum scalability, upgradeability, and field serviceability.

Reducing Carrier Expenses

AdvancedMC facilitates a modular approach to ATCA blade design that greatly reduces time to market and cost. By using AdvancedMC, TEMs do not need to develop a custom blade for each application. Instead, they can create application-specific blades by combining a generic ATCA carrier with generic AdvancedMC components such as network interfaces, control processors, network/signal processors, and mass storage devices. Because the ATCA blade and modules are generic, they can be reused across multiple applications, thereby reducing design time and production

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In some ways, MicroTCA is a repackaging of modular ATCA/AdvancedMC blades for small form factor.

cost. The generic nature of the blades and modules also makes them easier to outsource or purchase off-the-shelf, further reducing design time and cost.

Modular, field-replaceable ATCA/AdvancedMC systems are also easier and less expensive to scale and upgrade, reducing equipment costs by enabling carriers to deploy the minimal hardware needed to service their subscriber base. With AdvancedTCA/AdvancedMC, TEMs can stock a single generic carrier board that spans several products, along with the handful of generic I/O, mass storage, control, and signal processing modules needed to configure that carrier for specific applications.

Because they are field replaceable, ATCA/AdvancedMC systems can tolerate failures to individual blades/modules with minimal disruption to overall service. Modular ATCA/AdvancedMC blades also reduce provisioning cost by enabling service providers to scale and provision their systems according to actual demand.

MicroTCA tackles Small-Form-Factor Applications

The same capabilities that make AdvancedMC attractive as a mezzanine architecture make it equally attractive as a blade-level specification for MicroTCA systems. Its hot-swap capability enhances availability by enabling live systems to be serviced and upgraded in the field. Its large form factor and high power capability make it ideal for implementing complex functions. Its high bandwidth, protocol-agnostic packet interface provides an ideal interconnect for linking multiple modules in a chassis. Its IPMI interface facilitates centralized, fine grain monitoring and control. And its flexible form factor makes it possible to create mechanical MicroTCA chassis packaging options that are optimized for particular applications.

PICMG performed the first physical MicroTCA demonstration in June 2005 using a 2U/300mm/19” rack based system. The system simulated a wireless application servicing millions of subscribers. Final approval for the specification is expected in May 2006.

In some ways, MicroTCA is a repackaging of modular ATCA/AdvancedMC blades for small form factor, cost sensitive applications. ATCA’s large form factor, though ideal for building high density central office telecom infrastructure equipment, precludes its use in many outside plant and enterprise applications with tight size constraints. High cost also hampers the use of ATCA solutions in many outside plant, enterprise, and customer premises applications. ATCA carriers equipped with AdvancedMC modules have an added cost premium, as the carriers must be equipped with expensive card-cage-style connectors in order to house field replaceable AdvancedMC modules.

MicroTCA reduces size and cost by eliminating the ATCA carrier and enabling AdvancedMC modules to be used directly in a variety of compact, low cost enclosures, from standalone pico cells, to standard rack mount systems. The OEM production price for a baseline MicroTCA system, including a MicroTCA chassis, switching hub, and power module, is projected to range from $1,500–$2,000.

To accommodate a broad range of applications, MicroTCA is designed with scalability in mind. In addition to its scaleable packaging and power options, MicroTCA provides scaleable aggregate bandwidth from one to 650 Gbit/sec, and scalable availability ranging from three nines (.999) to five nines (.99999).

MicroTCA’s compact format, low cost, and low power consumption make it a perfect complement to ATCA for small form factor central office and outside plant applications, like wireless base stations, digital loop carriers, optical ADMs, and Fiber to the Curb optical network units. Some also see a role for MicroTCA in enterprise networking applications such as workgroup routers, modular servers, and SAN storage boxes.

Virtual Carrier Environment

A MicroTCA enclosure acts as a virtual carrier, emulating the ATCA carrier environment. The virtual carrier provides the interconnect, power conversion, clock distribution, and system management functionality need to support up to 12 AdvancedMC modules. Some of this functionality may be implemented using components integrated as part of an active backplane. However the most cost effective approach is to implement this functionality using a dedicated virtual carrier management (VCM) module. Systems requiring high availability would deploy these VCM modules in redundant pairs in order to eliminate the VCM as a single point of failure.

The virtual carrier interconnect fabric provides the main connectivity among AdvancedMC modules in a MicroTCA enclosure. The VCM module acts as a dual-star hub, providing a central switch and high speed lanes to each module. The half-duplex, serial lanes provide a scaleable bandwidth ranging from 3.125 Gbit/sec to 12.5 Gbit/sec per channel, compatible with the data rates supported by individual AdvancedMC modules.

MicroTCA’s small size, low cost, field replaceablity, and scaleable performance, coupled with its ability to utilize off-the-shelf AdvancedMC modules and ATCA/AdvancedMC infrastructure, make it an ideal platform for low- to mid-range telecom applications. Together with ATCA and AdvancedMC, MicroTCA provides an end-to-end framework that addresses the full spectrum of high availability telecom applications, from core routers and WDMs, to converged customer premises equipment. IT

Stuart Jamieson is director of advanced technology for Artesyn Communications Products, and he is also currently the draft editor for the MicroTCA PICMG standard. For more information, visit Artesyn online at http://www.artesyncp.com. (news - alert)

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VoIP MARKETPLACE

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TMCnet Traffic Analysis

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

TMCnet.com Traffic vs. Technology/IT Web Sites

Web Site	Alexa Site Rank
TMCnet.com	2,379
eWeek.com	2,826
Computerworld	4,671
InfoWorld	6,618
Network World	8,394
Light Reading	14,655
Pulver.com	36,063
Wireless Week	40,701
Destination CRM	48,598
Telephony Online	58,251
VoIP News	76,801
Telephony World	121,573
Call Center Magazine	183,448
America's Network	185,033
Telecomweb	204,159
CommWeb	249,258
Wireless Review	317,334
Communications News	984,904

TMCnet.com Traffic vs. Business Magazine Web Sites

Web Site	Alexa Site Rank
TMCnet.com	2,379
Fortune Magazine	2,484
Smart Money	2,980
Inc. Magazine	4,984
Fast Company	5,259
Business 2.0	5,986
Barron's Online	6,560
Weekly Standard	8,996
Technology Review	9,624
CIO Magazine	11,330
BtoB Online	23,419
Worth Magazine Online	174,723

TMCnet.com Traffic vs. Prominent Web Sites

Web Site	Alexa Site Rank
TMCnet.com	2,379
Sharper Image	4,152
Volkswagen	4,258
Nokia USA	4,351
Coca-Cola	7,670
Brookstone	10,045
GE Appliances	11,058
Brooks Brothers	14,899
JVC	15,692
Black & Decker	32,061

Source: Alexa.com 12/5/05

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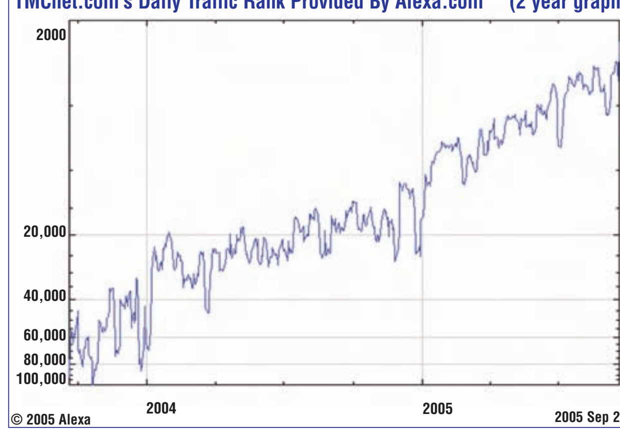
• Over 690,000 Unique Vistiors'*

*Source: Web Trends January, 2006

No Other Communications Site Even Comes Close!

TMCnet.com's Daily Traffic Rank Provided By Alexa.com** (2 year graph)



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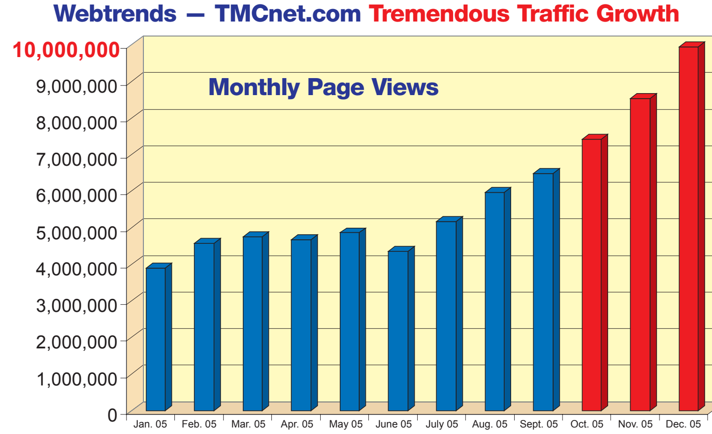

**Source: Alexa.com ranks Web sites by traffic. The number indicates a site's proximity to being the number one most visited Web site. Date: 12/5/05 Alexa is an Amazon.com Company. Neither Alexa.com nor Amazon.com endorse, or are affiliated with, TMCnet.com in any way.

CEO Spotlight

John Legere
Chief Executive Officer
Global Crossing

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 John Legere, Chief Executive Officer of Global Crossing. (news - alert)

GG: What is Global Crossing's mission?

JL: First off, we start with our vision — which is to be the recognized leader in next-generation global communications. Our mission, which says how we execute against that vision, is to grow our business using direct and indirect distribution to sell our advanced, secure IP solutions while providing a superior customer experience.

As a global IP service provider, we serve our customers and their reseller carrier partners with the first-to-market implementation of a leading edge, global MPLS-based IP platform. Combine that with unparalleled levels of network security, a highly responsive global customer support and service model, and increased customer control with our Intelligent Front Office, and you have a pretty good 30,000-foot perspective of what Global Crossing is all about.

By the way, our industry-leading customer satisfaction scores, five nines of network reliability and award-winning products and services all speak to the great job our 3,500 employees do every day to consistently deliver on our mission.

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

JL: Our vision at Global Crossing is very clear — to be the recognized leader in next-generation global communications. We're accomplishing that through innovation, partnerships, and investment in IP services, collaboration, and convergence infrastructure.

We're well positioned as a next-generation provider of converged communications, delivering a totally integrated, interoperable suite of IP services. Our go-to-market strategy includes partnering for best-in-class IP solutions, such as our managed services, converged IP applications, VoIP, collaboration services, and IP telephony solutions.

As consolidation plays out in the industry, we feel our position will only grow, especially since we're one of the few remaining independent players in the global IP services niche.

GG: Now that growth and opportunity are the trends in the VoIP industry, what possible hurdles do you see that might upset this momentum?

JL: VoIP has seen tremendous growth among early adopters and is gaining steady acceptance in the mainstream marketplace now. I don't see major hurdles derailing customer adoption. However, adoption isn't always easy or immediate. What I mean is that customers must derive a business benefit to convert to VoIP. IP convergence is that key driver, and as customers converge voice, video and, data networks into a combined IP infrastructure, VoIP adoption should accelerate.

Another issue to note is the globalization of VoIP services. As a global provider, we're experiencing first hand the different approaches to VoIP adoption in different countries. Government regulation, licensing, service types, and availability vary significantly. For customers, the complexity of managing the VoIP nuances among their different global offices can pose a convergence challenge.

So as Global Crossing deploys VoIP services, we're keenly focused on developing consistent offerings around the world. As I mentioned earlier, our customer satisfaction leads the industry — 98 percent of our enterprise customers state that they are satisfied with Global Crossing and more than two-thirds are "very satisfied." And we won't quit until every customer is singing our praises.

Our network was built for IP and we're beginning to reap the benefits. Today approximately 70 percent of the voice minutes that traverse our global network infrastructure are IP. That translates to more than two billion minutes per month, and by the end of the year we expect virtually all of our voice traffic to run over our VoIP backbone. In addition, our IP VPN traffic, which supports converged IP solutions, grew 300 percent on an annualized basis during the first three quarters of 2005.

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

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In a converged network voice will be no more than another application similar to data and video.

JL: We now put an increased level of focus on the services and applications that our customers use — IP video, audio, and Web collaboration, and a mushrooming set of advanced voice applications. These services are where the real value is derived from our network, and they change the way that our customers conduct business.

In addition, our customers demand an ever increasing amount of control of the products and services they purchase. So we continue to make improvements to our customer portal, and integrate it with our back-office systems to create something we call the Intelligence Front Office, or IFO. This will allow customers clear visibility of all their applications across a converged IP network.

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

JL: The world has been waiting to see what the application would be that would drive people to consider IP convergence. Many thought it would be data or video oriented, but it turns out that voice — and the cost savings of VoIP — are driving increased adoption. With that in place, businesses can immediately see the tremendous benefits of converged IP services, which is where we come in.

As cited in a recent *Harvard Business Review* article, companies that harness VoIP to achieve business objectives will find it much more than an undifferentiated commodity technology. In a converged network enabled by IP, voice will be no more than another application similar to data and video.

Global Crossing has been and always will be a future-thinking company. We built the world's first IP network and we're now starting to realize its tremendous advantages. We'll continue to invest in and discover new technologies, allowing our customers to always be ahead of the curve in their communications. IT

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CEO Spotlight

Marc Zionts
Chief Executive Officer
Excel Switching

In the CEO Spotlight section in *Internet Telephony*®, we recognize the outstanding work performed by exemplary companies. Each month we bring you the opinions of the heads of companies leading the Internet telephony industry now and helping to shape the future of the industry. This month, we spoke with Marc Zionts, Chief Executive Officer of Excel Switching Corporation. (news - alert)

EAS Group, parent company of Excel Switching recently announced the acquisition of Brooktrout, Inc. The combined company now has greater resources to invest in technology that enables service providers and enterprise customers to develop new products, introduce new services and cost-effectively transition networks to IP.

GG: What is Excel Switching's mission?

MZ: Excel Switching's mission is to deliver the broadest range of products to enable those creating and delivering communications solutions to develop new products, introduce new services, and help them cost-effectively transition their networks to IP.

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

MZ: My vision for Excel/Brooktrout is to create the telecommunication's industry leading provider of enabling technology and to continue to expand into open-source standard-based technologies such as IMS, IP video, and beyond. We want to continue to provide our customers and partners with our proven expertise and experience to help them bring their innovative solutions to life. Excel is positioned quite well in the next-generation telecom market. We address the needs of both enterprises and service providers and have a variety of products that can help them as they make their transition from traditional networks to IP-based solutions.

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?

MZ: The past year we have certainly heard a lot of hype about IMS and have seen companies invest in the new technology. If IMS does not take off as quickly as people expect it to, then we could certainly see an upset in the VoIP momentum. In addition, it is a crowded marketplace with many companies seeming to offer similar products. Customers will need to ensure that they work with a company that has proven products, strong partners, and a stable environment. One of the major trends in the industry is the continuation of the outsourcing of various corporate functions. For example, you can expect to see more collaboration between companies when it comes to research; more commodity hardware and software; and more sharing of ideas.

We are still in a very challenging time for the telecommunications industry. Many of the network equipment manufacturers are teetering on bankruptcy and have historically low market capitalizations. The incumbent service providers are seeing their cash flows becoming dramatically reduced due to new technologies and new competitive and regulatory pressures. Even though they're still making billions of dollars, their profits for the first time are being seriously challenged.

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

MZ: Excel is continuing to focus on emerging technologies, such as speech recognition, and is certified to work in Microsoft's Speech Server platform. We continue to develop our IP Video messaging solution by growing our partnerships with companies like Openwave and a deployment at Portugal's largest mobile provider, TMN. We also invest in emerging technologies such as IMS gaming with partners like IBM. In addition our media gateway product helps companies making the shift from TDM to IP. We don't see "all IP" as a reality yet, and our unique position in both TDM (define - news - alert) and IP allows us to help our partners as they make their transition.

GG: Please describe the acquisition and integration of Brooktrout Technology. How does this affect your future plans?

MZ: The acquisition creates a combined company that has greater resources to invest in the research and development of the enabling technology that customers — service provider and enterprise — require to develop new products, introduce new services, and cost-effectively transition legacy networks to IP. We are pleased to be able to deliver our customers a wide array of enabling technology products. We will continue to look for opportunities to grow and strengthen our business. We plan to officially launch the company in March and will be excited to share more of our mission and plans for the future then!

> If IMS does not take off as quickly as people expect it to, then we could certainly see an upset in the VoIP momentum.

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

MZ: At the start of 2006, we are going through another change in the telecommunications ecosystem. The reality of disaggregating services from location in the network is far more profound then we had envisioned in 2000. In 2000, we believed VoIP would dissolve the distinction between long-distance and local providers, as well as blurring the distinction between a network offering and a service bureau offering. However, we have come to learn that for many services they can be equally deployed as network services or as enterprise, premises services.

This change in how services are being delivered is impacting our traditional business models. Today, we find demand for products that was traditionally sold to carriers also within the enterprise. For many enterprises, they need the scale and reliability of the traditional products. However, there's an even larger number that will require the reliability, but not the scale, of the traditional products.

The availability of tools such as .NET, SALT, and Live Communications Server from Microsoft, is a manifestation of the promise of voice just being another modality to access enterprise applications. Likewise the popularity of service-oriented architectures is bringing technologies such as VoiceXML and the stimulus markup paradigm to the enterprise. What is different is that rather than simply building telecommunications applications with these new development paradigms, we see the application model turned upside down. These new technologies are allowing a different, less technical set of people create telecom applications. This will help grow the telecom industry and provide opportunity for new and exciting applications. IT

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