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The Enterprise PBX Is Dead!
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Broadband Over Powerline
Alternative Service Delivery Options

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

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By Greg Galitzine

Offer Me Alternatives

A recent post from Om Malik's GigaOm blog pointed out that AT&T, Qwest, and Verizon have seen cumulative losses of 1.75 million voice lines. The beneficiaries of this voice line flight? Cable companies. Now granted, the phone companies are more than holding their own when it comes to broadband access lines, outselling their cable competitors 1.5 million to 909,000 during the same period.

Win or lose, any way you slice it, we're talking about a duopoly that enjoys the lion's share of the customers in the voice market. Of course, we need to factor in the wireless carriers, who have enjoyed tremendous increases as customers shy away from subscribing to any fixed line service opting for an "all-mobile, all-the-time" approach, choosing to use their mobile phone as their only phone.

But what about the alternatives? What's going on with regards to Broadband over Powerline (BPL) or WiMAX (define - news - alert), for example?

WiMAX

Intel (news - alert) and the NDS Group (a supplier of open end-to-end digital pay TV solutions for the secure delivery of entertainment and information to set-top boxes and IP devices) recently announced a joint venture to offer television and videos via WiMAX.

The solution is designed to deliver live TV streams and video on demand (VOD), including an integrated electronic program guide (EPG), to a laptop with an Intel Centrino on the basis of 802.16 and 802.11.

Also, MediaRing announced the official launch of AngkorNet, the first Internet Service Provider in Cambodia to offer tiered and bundled packages of high-speed Internet access and other data applications using WiMAX technology. AngkorNet delivers up to 90 percent WiMAX coverage in Phnom Penh.

BPL

Current Communications Group (news - alert) recently announced that it had secured $130 million from investors. The company, which has developed technology to provide high-speed Internet access over power lines, is already offering BPL service to consumers in Cincinnati and Texas.

New investors in the company include TXU Corp., General Electric, EarthLink, Inc., which will resell Current's broadband services via the retail channel, and Sensus Metering Systems. Existing equity investors include Duke Energy Corporation, EnerTech Capital Partners, Goldman, Sachs & Co., Google Inc., Hearst Corporation, and Liberty Associated Partners, LP.

It's interesting to see that EarthLink and Google are major investors in a BPL provider. These two companies are prime examples of service providers that do not own their own infrastructure and are, therefore, basically at the whim of phone and cable companies when it comes to gaining access to pipes to deliver their services to customers. As the debate over Net Neutrality evolves, it will be fascinating to see if this opportunity pans out.

The point is that there are options out there, and serious investors with deep pockets are placing bets on the success of these other alternatives. I'm curious to see how this plays out, and to see what new technologies might emerge to pose a challenge to today's status quo. The possibilities are limitless. IT

Greg

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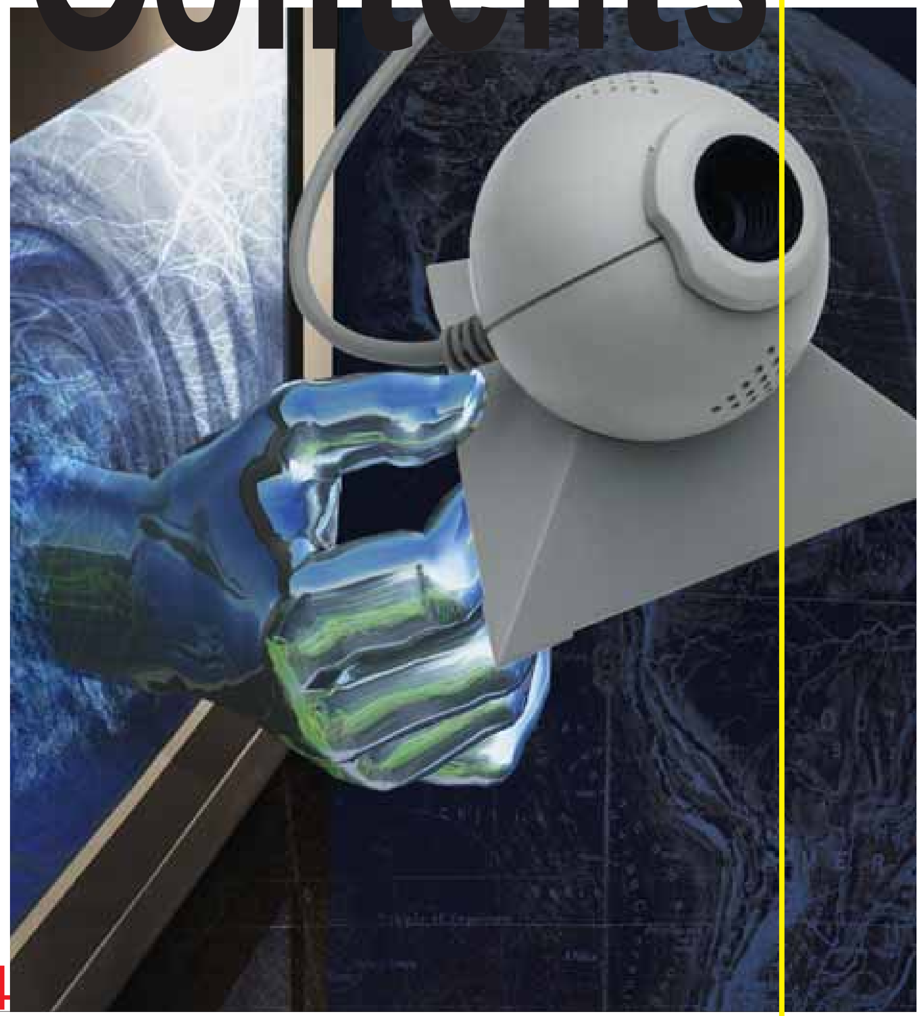

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QUOTE OF THE MONTH:

"There are some that believe that names will ultimately dominate the domains and that last remnant of the PSTN, the telephone number, will be relegated to the Smithsonian. There are others that know nothing of this concept and are trying to build businesses and revenue models on numbers through ENUM. Telling the future is tricky business, so look to the past to see what has worked. Are we all just a Social Security Number, or are we really known by our name?"

— Hunter Newby, page 64

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FEATURE ARTICLES

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2. Milton, Australia
3. Marian Del Rey, California
4. Middletown, New Jersey
5. New York, New York
6. Singapore
7. Cambridge, Massachusetts
8. San Jose, California
9. Toronto, Canada
10. London, UK

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The VoIP for SMB Community is the leading online resource for business-class broadband services that combine voice and data. This site is your primary destination for the most relevant news, education, and trends analysis. But that's not all. VoIP for SMB will help you navigate through the myriad of service providers and plans — Empowering you with the knowledge to make the right decision for your business. Come check it out at http://www.voipforsmb.tmcnet.com. Sponsored by Covad.

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

WHAT'S ON TMCNET.COM RIGHT NOW

To stay current and to keep up-to-date with all that's happening in the fast-paced world of IP telephony, just point your browser to http://www.tmcnet.com for all the latest news and analysis. With more than 16 million page views per month, translating into more than 1,000,000 visitors, TMCnet.com is where you need to be if you want to know what's happening in the world of VoIP.

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

Is IP Telephony Really an All-Or-Nothing Proposition?

IP Telephony is presented in the marketplace as an all-or-nothing proposition. This all-or-nothing characteristic comes at us from two different angles and, either way, the customer is looking at a big stick approach to something as simple as a desktop phone system. It seems that the value of the "phone service to the desk top" model is diminishing rapidly. http://tmcnet.com/297.1

Will video break the Internet?

Every day, it seems, a new service pops up offering to send you video over the Internet. *Desperate Housewives*, Stephen Colbert heckling the president, clips of bad dancers at wedding parties: It's all there. You may be up for it, but is the Internet? http://tmcnet.com/298.1

Teleconferencing Helps the Global Delivery Business Model

Gone are the days of the traveling salesman. Today, small and medium size companies worldwide have found that traveling for work is more likely to be motivated by knowledge sharing and building relationships rather than boosting sales. IP technology has revolutionized the way businesses operate and communicate. http://tmcnet.com/299.1

The Folly of the Empowered Agent

More and more contact centers are trying to take advantage of the interaction time during a routine service call. By adding a sales component to a standard service request, companies believe they have found an easy path to increasing revenues. But without equipping agents with the proper tools and training to sell, the transaction can frustrate or even anger the customer and may cause the opposite effect — declining revenues. http://tmcnet.com/300.1

Enterprise Communications Market Shifting to Software-Based Value Model

Enterprises are increasingly abandoning their legacy PBX and 1st generation IP PBX "closed box" solutions and moving to 2nd generation business communication software (BCS) platforms based on open standards. http://tmcnet.com/301.1

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By Rich Tehrani

Coming Soon: The VoIP IPO

Somewhere around 1998, a stockbroker acquaintance called me to tell me about a great opportunity — and proceeded to tell me how it was a great time to purchase AT&T. I explained the whole concept of IP telephony to the broker. I told him that buying AT&T when VoIP is taking off is just plain stupid. He basically told me I didn't know what I was talking about.

Fast forward many years to this afternoon. I was catching up on my reading and came across an article about France Telecom and how the new CEO of the company was forced to lower guidance twice in three months because of competition from VoIP. (define - news - alert)

Avid TMC readers know that we recognized VoIP's potential to wipe out the PSTN in many of the articles we wrote back in the 1996–1997 timeframe — before Internet Telephony Magazine was launched in 1998.

It seems to me that some large service providers don't actually read magazines, or is it a case of them just not believing what they read? Maybe they prefer to close their eyes to the painful truth. The fact that France Telecom is now looking for new revenue sources tells me just how plain out of touch they have been with reality for nearly a decade. Shouldn't they have had planning meetings and have been far more prepared for this eventuality? After all, Dialpad was the Skype of the 90's and must have shown up on every service provider's radar screen.

What is France Telecom doing now to deal with the diminishing PSTN revenue? They are looking at new sources of revenue like broadband, security over IP solutions, and digital television. The company is also moving into content and has started to broadcast details of soccer games to mobile phones. In addition, the company is making music videos available via mobile phones.

My take on all this? These are great ideas that are late to market. How do billion dollar companies not anticipate change and react so slowly?

You have to give tremendous credit to AT&T, (quote - news - alert)who saw this transition coming and purchased cable companies. They weren't able to figure out how to leverage these acquisitions, but at least they had the vision to make the right decision. They were likely too early. Perhaps they can be accused of being too visionary, if that's even possible.

A friend of mine recounted a recent conversation with his VC where the investor told him that in any market the cycle is too early, too early, too early, too early, just right, too late. How appropriate.

Adding to this misery, Skype just announced that calls to telephone numbers in the U.S. and Canada using their SkypeOut service will be free. As mentioned above, Dialpad was the first company to offer such a service. Now Skype will do it. I am not sure what the revenue model will be but I suppose ads may come into play.

If France Telecom thought life was difficult before, wait until other VoIP providers are forced to follow suit. Will the VoIP market become free and ad supported? If so, isn't Google in the best position to reap the rewards of this brave new ad supported VoIP world?

Large service providers don't actually read magazines, or is it a case of them just not believing what they read?

Regardless, while companies like Skype put more and pressure on companies like France Telecom, Vonage plans on filing for their very own IPO to reap the rewards of the low-cost long distance environment they helped create.

But how does one convince investors that your IPO makes sense when Skype just cuts your legs out from under you with precision not seen since the last Quentin Tarantino flick?

Vonage (quote - news - alert)was on its way to taking over the world. Can it still? Do they have a model that competes with free? Do they have advertising software available?

Worse, what if Skype leverages eBay listings as its supply of ads. Yes you read that right. What if the world starts using Skype to call the U.S. and every time they do, they are presented with myriad impulse items or other products listed on eBay? Skype (news - alert)may not even need an ad supported model. After all when you have rich parents you really don't have to work for a living — right?

Many of you have asked me if I would buy into the Vonage IPO and I am unsure. It is likely to be volatile as Vonage has said repeatedly they will sacrifice profit for market share. So the institutional investor is probably not so interested in this offering — At least the institutional investor of the past who wants to see an immediate plan towards profitability. If the institutional investor avoids Vonage then the individual investor will have to pick up the slack.

An amazing amount of people have told me they are Vonage customers and want to invest. This reminds me of the Boston Chicken IPO of last decade that spiraled upwards for some time based on happy customers.

Of course, VoIP is no ugly duckling and the comparison to Boston Chicken, now called Boston Market is only meant to reinforce how strong a brand Vonage has built. But what is a brand worth if you can't monetize it?

To answer this question we look to Amazon, which had a "take-over-the-world" mentality and continued to invest in new markets after the tech bubble burst. Eventually, shareholders told Jeff Bezos that he had better start producing profit or his hallmark laugh would be heard on the unemployment line.

Bezos eventually found a way to sell everything on Amazon but, instead of having to carry all the inventory, they started to partner with all sorts of merchants. Shortly thereafter Amazon became profitable. In my opinion, any day that Vonage wants, it too can turn a profit. But in doing so, of course, it will lose share. Right now, seeing a Vonage ad is like playing video games, eating mom's apple pie, and playing baseball. It is part of the American culture. Frankly, the brand is probably strong enough to take a slight marketing hit.

In short, I am not sure if I would invest, but I believe there is greater than 50 percent chance that the stock will do well after and during the IPO. The shares will likely see some good growth and then come back down to reality, based on analysts who point out how competitive the market is. Then Vonage will release earnings that are better than expected and shares will go up. Everyone seems to want me to go out on a limb here; the question is will my readers saw off the branch if I am wrong?

Some in the industry are afraid. They wonder, if Vonage does poorly, will it be bad for all VoIP investments? While these fears are based in reality and may be well-founded, from where I stand, the VoIP industry recently had a two-billion dollar acquisition. There is obviously value in this space.

However, a poor Vonage IPO would hang like a dark cloud over the IP communications space. But I don't think this is bad. There are more VoIP companies popping up now than ever.

In my opinion, it won't be bad to see investors take a step back and figure out what they are doing. After all there are likely 5,000 VoIP service providers worldwide and many have affiliate programs, making it seem like even more. There can't be 5,000 phone companies in ten years. Frankly, many of the companies in VoIP have great products and lousy marketing. They need some pressure on them so they can sell their companies and get out of the way. Fewer companies will mean each of the remaining organizations can be more profitable as there will be less competition.

Besides, there are other areas of VoIP that have yet to be exploited. There is much opportunity in WiFi telephony, SIP, and IMS. There is room for many more companies in these areas. VoIP quality and security are areas that need more attention. And, there are more and more opportunities in open source each and every day.

Let's not forget that VoIP is experiencing massive growth and the FCC and other government agencies are till trying to figure out how to regulate this space and exactly what to do with it. In a recent interview with Bryan Martin, the CEO of Packet8, he told me he is surprised there is not more foreign acquisition of VoIP companies. He thinks that acquisitions have been slowed because investors aren't sure what will happen to the space.

A bad Vonage IPO may be an additional drag on VoIP investment, but that may not be a bad thing for an industry with so many similar competitors.

And the government is, indeed, struggling to figure out what to do with VoIP as it blurs lines and changes the ways things are done. They are also concerned about other telecom issues, like being able to spoof Caller-ID and the fact that a chat session can escalate to a VoIP session. How do they enforce do not call laws in such situations? The de-siloing of

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communications due to the proliferation of IP must be a nightmare for anyone trying to regulate different media types in an orderly manner.

I wonder if we would see even more investment in IP communications if the government cloud were lifted.

To the investors in this market I would like to say be careful. Try to understand what the company you invest in is doing. I would also advise steering clear of companies run by engineers who don't plan on bringing on business people to manage the company. If they don't understand marketing, branding, and positioning, just pass. There is one exception here. Sometimes good technologies make a company an attractive takeover candidate. If you fill a hole in Lucent's or Cisco's portfolio and you think you can get them to bite, then you may have a winner. Still, I think it is risky to have technologists as CEOs in today's environment that is so focused on profit.

You should focus on companies that are generating profit or have a sound plan. If you see the 5,000th company in a market, you may want to steer clear. But, if you find a company that you think is dynamic enough to exploit a niche in the market, you may have a winner. Part of determining if a company fits in this category is whether the management has experience in launching successful companies in the past. What is their track record? The IP communications space moves faster every day and you need a team that can keep up with it all.

So, in the end, a bad Vonage IPO may be an additional drag on VoIP investment, but that may not be a bad thing for an industry with so many similar competitors.

Perhaps the ensuing Mitel IPO is a better bet as there are fewer competitors in this space. Mitel has always been a great technology company. The one weakness is that they aren't as visible as some of their competitors. An IPO may force them to focus more on awareness and branding, which would help them as they go up against Avaya and Cisco.

The discussion surrounding these IPOs is timely, as this August TMC will be hosting the IP Communications Business Summit in Santa Clara where we will bring together the investment and VoIP communities. This event will occur in concert with the newly launched VoIP Demo show, which will take place at the same time and place as the very successful VoIP Developer Conference. I hope you will make it to Santa Clara this August to join the rest of the TMC community for what is bound to be a great week for the VoIP industry. IT

China vs. USA

In the battle for technological supremacy, it seems that China has lost a major battle in its efforts to end its reliance on imported DSPs and other chips. The New York Times reported that Chen Jin, a dean of Shanghai's prestigious Jiaotong University and the leader of a government-funded high-tech research project, was dismissed from his university posts this week and stripped of other government titles and perks. The government also said that Professor Chen had been permanently banned from taking part in any government-funded science projects.

In a statement, Jiaotong University — one of the nation's elite schools — said, "Chen Jin has breached the trust of being a scientist and educator. His behavior is despicable."

Certainly, this should be great news for companies like Texas Instruments and others that rely on DSPs to provide valuable revenue. In addition, Intel can breathe a sigh of relief, as this news will likely set the entire country's independent tech discoveries back a bit.

This news shows how some in China see the need to import technology from beyond China's borders as something so horrid that it becomes better to fabricate research than import. It shows you how seriously many in China take their role as an up and coming technology powerhouse.

Perhaps this discovery of fabricated research will actually slow China down a bit and force them to become a better tech citizen. Hopefully, this embarrassment will make others in the country to think twice before they try something similar.

In addition, this news also shows how difficult it is to duplicate research that has been done in the U.S. and other countries. It would seem that even infinite resources do not let you become the equal of more established countries overnight. In the end, it takes time, hard work, and honesty to produce worthy products in the technology space.

For now, China has received the equivalent of a technical foul. Hopefully, they will come out and play in a more sportsmanlike manner for the rest of this game. IT

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News Analysis By Robert Liu
TMCnet Wireless and Technology Columnist

LM Ericsson See Dividend from Marconi and 3G

NEW YORK — The advent of true, mobile broadband through technologies, like High Speed Packet Access (HSPA) and consistent growth for fixed line service is creating tremendous demand for traffic through the core network and for long haul transport and transmission service, according to officials at LM Ericsson Telephone Company.

With the Swedish telecommunications giant roughly seven months into its merger integration of Marconi Plc's optical networking business, the renewed capacity demand is creating new market opportunities, Carl-Henric Svanberg, President & CEO, told analysts, investors, and the media at its Capital Markets Day event in the Big Apple's Financial District.

"We are talking about, on one hand, the build out of mobile broadband... more subscribers that can do more. Download music, TV, whatever we want to do, we can do that in the network," Svanberg told the gathering that included even Swedish Monarch, King Carl XVI Gustaf. "At the same time, on the other side, we have the roll out of fixed broadband. All of that creates a tremendous opportunity in growth and it's all going through the same converging core network. But on top of that, it's also got to travel bigger distances and that's where transport and transmission comes into play. So this is building up a tremendous potential also and demand in the transport/transmission networks and that is also where our acquisition of Marconi comes into play."

Against the backdrop of the Volvo Ocean Race (formerly known as the Whitbread Round the World Yacht Race, which Ericsson both participates in and sponsors), Ericsson offered its latest contract win, the metro DWDM optical network for Telefónica de España, as proof-positive that it stands to benefit from the greater demand for capacity. The Telefónica multi-haul contract almost entirely represents the business of the former Marconi assets and helps to justify the multibillion-dollar bet placed on the struggling U.K. telecom equipment vendor.

Despite the encouraging signs, though, the Marconi restructuring remains ongoing and Svanberg estimated the cost by year end will likely reach 2 billion Swedish Kronor ($275 million) and include the layoffs of 1,600 workers — which resulted in considerable flack for the Swedish company, even prior to formalizing the acquisition last fall. That said, from a business standpoint, the acquisition has provided many of Marconi's legacy customers with the reassurance that Ericsson won't abandon support of their network infrastructure.

Pictured from left to right: Ambassador Gunnar Lund, Sweden's newest ambassador to the U.S. and King Carl XVI Gustaf with Carl-Henric Svanberg, President and CEO of LM Ericsson, attending Ericsson's Capital Markets Day.

Race sponsorship has created opportunities with more than half of the world's carrier community for Ericsson.

"The acquisition ... the reception of it ... the appreciation from customers was beyond what we expected it to be," Svanberg said.

However, that still doesn't alleviate investors' concern about the impact fierce competition is likely to have on Ericsson's overall profitability. The competitive threat of encroaching equipment vendors was only highlighted recently when French firm Alcatel announced its intentions to merge with Lucent Technologies to form a combined $25 billion company.

Yet, the Marconi transaction has taught Ericsson's top brass that merger synergies on paper aren't as easy to realize in practice. Even if product lines overlap, existing customer relationships empower a vendor to keep supporting both product lines and that is likely to keep both Alcatel and Lucent folks busy for some time, the Ericsson chief official explained.

"The problem we have with vendors is that if you go and acquire another vendor with overlapping products, you still can't take out the synergies. You must keep those installed networks alive because if one company buys the other one, and the other one's customers say are you going to keep developing my technology, my network…if you're not going to do that and say no, then you're going to have to swap that network. If that is up for swap, then everybody's interested again. Every vendor wants to come in. Then it's better to swap the networks than make the acquisition," Svanberg concluded.

Unlike its competitors, Ericsson's position is unique in that its technological edge extends into other realms of the wireless ecosystem. On top of the core and long haul businesses, Ericsson Mobile Platforms (EMP) is helping operators build out HSPA-based high-speed networks scaleable to the IP Multimedia Subsystem architecture. The company also has an extensive global services business as well as its handset joint venture with Sony. By its own measures, today's 3G networks already have enough capacity to meet subscribers' demands for entertainment like mobile TV, Hakan Eriksson, Ericsson's Chief Technology Officer, said during his presentation.

Ericsson takes a more realistic approach to mobile TV. Using a combination of unicasting and multicasting technologies to satisfy fluctuating demand because, the more popular channels could be broadcast using Multimedia Broadcast Multicast Service (MBMS), other, less popular content offerings could be sent via unicast.

"Why spend output power and resources in the base station of the cellular network to broadcast something that nobody wants to watch," Eriksson contended.

If 100 percent of subscribers wanted to access mobile TV, the 3G network would only be able to handle MBMS traffic of less than 20 minutes of average usage, but, if roughly half of the total subscriber base is interested, that would allow for nearly 30 minutes per day. Based on internal surveys, end user interest in 30 minutes of MBMS service currently stands at only 15 percent.

"There's plenty of space in the cellular system," the Ericsson official explained. IT

Robert Liu is Executive Editor at TMCnet. Previously, he was Executive Editor at Jupitermedia and has also written for CNN, A&E, Dow Jones and Bloomberg.

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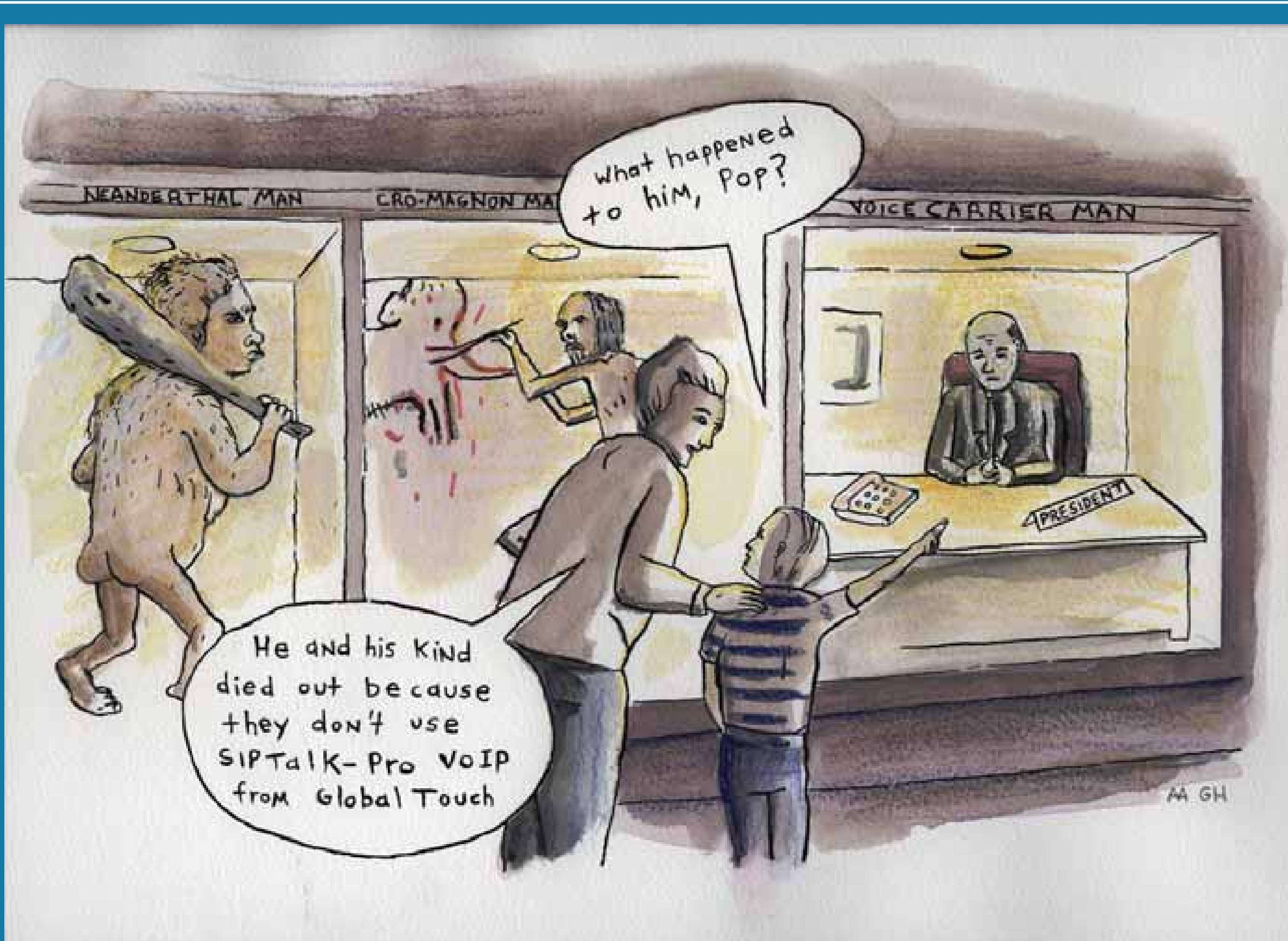

NEANDERTHAL MAN
CRO-MAGNON MA
what happened to him, Pop?
VOICE CARRIER MAN
PRESIDENT
He and his kind died out because they don't use SIPTalk-Pro VoIP from Global Touch
GH

Encentuate Allows Easy, Secure Access from Anywhere at Anytime

Encentuate Inc., (news - alert) a provider of enterprise access security solutions, announced the release of Encentuate Web Workplace, a solution that simplifies, strengthens, and tracks access to information systems via Web portals, extranets, and browsers. The solution also ensures that single sign-on, strong authentication, incremental security and compliance tracking are available consistently wherever a user gets on the network, providing secure and easy access anytime, anywhere to authorized users.

Encentuate Web Workplace expands Encentuate's vision to provide identity management at the enterprise endpoints and complements the existing support for personal and shared workstations, Citrix, and Terminal Services. Web Workplace ensures that users need just one password and no installation of desktop software to access multiple information systems remotely from home offices and Web cafes or through browsers on handheld devices like PDAs and other mobile devices.

Web Workplace also allows IT administrators to leverage the Encentuate IMS Server to centrally manage identities and access policies. One of the key benefits of Web Workplace is the ability to track and centrally report all access events to the IMS Server. IT can then develop comprehensive user-centric reports to audit access to information systems across enterprise end-points.

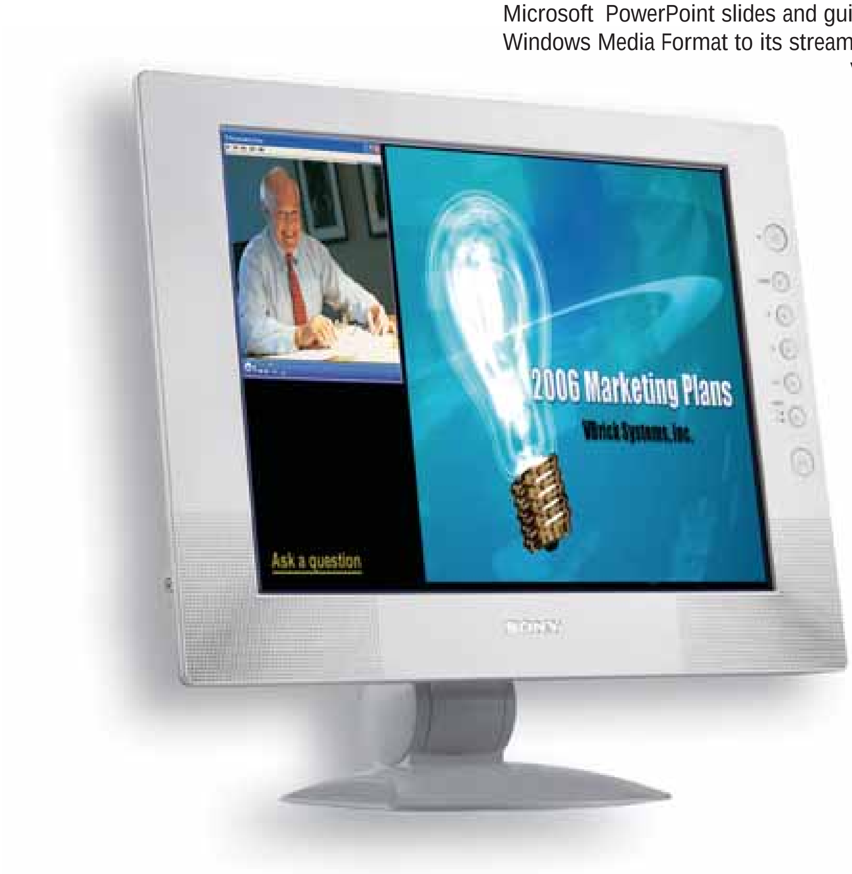
http://www.encentuate.com

VBrick Adds Presentation Materials to Streaming Video Capabilities

By Erik Linask

VBrick Systems (news - alert) has added presentation capabilities to its Webcasting repertoire. VBCorpCast is the firm's latest release and is designed to facilitate simple and even more effective streaming than ever before via the public Internet.

Key features of the two new releases include the incorporation of live, synchronized Microsoft PowerPoint slides and guided Web browsing, as well as the addition of Windows Media Format to its streaming video format library. Both new kits, the VBCorpCast and VBEduCast — targeting corporate users and educators respectively — enable users to initiate all multimedia presentation capabilities using the product's Microsoft PowerPoint interface, eliminating the need to learn new applications.



The VBrick Webcast kit is a complete, out-of-the-box presentation streaming solution that combines video with multimedia slides and Web content, as well as interactive audience polling and Q&A capabilities. The package comes complete with everything users will need to set up and complete live Web broadcasts — the VBrick appliance, a camera microphone, wires to connect the appliance, and 50 GB reflecting service, in case you don't have enough bandwidth available for your presentation.

VBCorpCast also records the entire presentation, including slides, questions, and online polling, and automatically archives it in the Internet for later viewing. At the completion of the presentation, a simple click of the mouse button will begin the archiving process, which takes only as long as it takes to upload the file to the server.

http://www.vbrick.com

Clear-Com Launches SoftVoICE 1.0 to Deliver Superior Intercom Quality

Clear-Com, (news - alert) an intercom solutions provider, announced SoftVoICE 1.0, intercom software for use on standard personal computers running Windows XP. SoftVoICE works in partnership with VoICE (Voice over Internet Communications Equipment), a 1-RU 4-way VoIP interface frame that connects remote users seamlessly and efficiently over low-cost house LANs, private WANs, and other communication links using Internet protocols. Each VoICE frame enables an Eclipse Matrix to convert four of its physical station ports into SoftVoICE connections.

Connecting remote personnel with fixed sites can challenge studios and switching centers because links are tricky and costly to coordinate while telecom solutions suffer quality and reliability issues. SoftVoICE, on the other hand, is as simple to use as an instant messaging client while delivering broadcast-quality audio from around town or around the globe.

SoftVoICE significantly reduces the cost and complexity of audio connections between studios and talent in remote locations or tight spaces. And with advances in image compression and video streaming, nothing prevents the talent from receiving moving or still pictures on a desktop or laptop to fully engage in the conversation. The same holds true for public safety employees, live event engineers, critical business conference attendees, or even military forces.

http://www.clearcom.com

Citel Enables Gradual Enterprise VoIP Migration

By Johanne Torres

Citel (news - alert) introduced the EXTender IP6000, a system that enables enterprises with multiple locations to gradually migrate their telecommunications to an IP telephony-based network.

With the EXTender products in place, businesses will be able to connect remote call centers, home workers, and branch offices to a central digital PBX over an IP network. The system offers reduced telecom operating costs, single voicemail and call center applications, central reception, and four-digit dialing throughout the enterprise.

As the business prepares to complete the migration to SIP, the EXTender IP6000 can be software-upgraded to accommodate the premises or service provider hosted IP PBX, integrating the existing handset and wiring infrastructure at each location. This process allows businesses to experience a smoother migration to SIP telephony in the future, without having to replace existing infrastructure.

http://www.citel.com

iPass and Nokia Develop WiFi Connectivity Client for Nokia

iPass Inc. (news - alert) and Nokia Enterprise Solutions (news - alert) announced they are developing iPass wireless connectivity software for the Nokia 9500, the Nokia 9300i, and the Nokia Eseries devices. Based on the iPassConnect universal client, this software will extend availability of the iPass secure remote access service to users of Nokia Business devices.

In addition, Nokia began a user pilot of the iPass Corporate Access service, ahead of a planned company-wide roll-out around the world later this quarter. Through the iPass service, Nokia plans to provide its remote and mobile employees with secure and reliable connectivity to company resources via the Internet.

Users of the WiFi-enabled Nokia mobile business devices will be able to join the hundreds of thousands of active quarterly iPass users who can securely connect to the Internet and key corporate applications from approximately 50,000 WiFi venues worldwide.

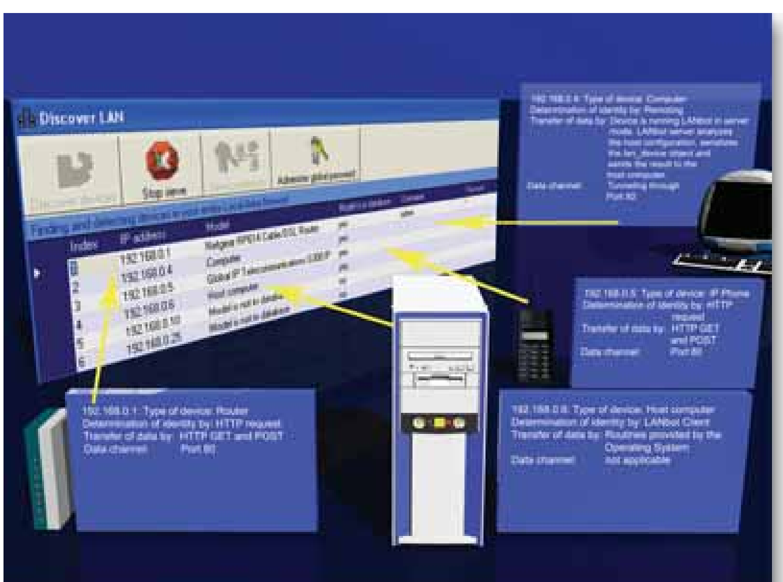
iPass Corporate Access gives remote and mobile workers a safe and simple way to connect to the Internet from over 160 countries around the world with dial, ISDN, wired, and wireless broadband connectivity. It offers a full range of services, including centralized policy-based management and end-to-end security enforcement that allow IT managers to maintain control over how users connect to the Internet without jeopardizing key corporate assets.

http://www.ipass.com
http://www.nokia.com

Auto-LAN Configuration Device Simplifies Network Optimization

By Erik Linask

Global IP Telecommunications (news - alert) has successfully completed the proof of concept version of its software that will automatically configure LAN devices. LANbot resolves frequent discrepancies and redundancies that inevitably are found in network installations with multiple devices sharing limited resources.



LANbot is a versatile LAN setup and installation utility designed to allow users of all levels to automatically configure their LAN devices for optimal performance. LANbot automatically resolves conflicts (e.g. port conflicts), configures both hardware and software, and provides a common user interface in at least three different languages. Global IP Telecom says LANbot will be the tool of choice for bringing new and existing customers online for ISPs.

The patent pending LANbot can automate the analysis and setup of LANs. It can detect routers and all other network devices and, rather than require a separate user interface for each device, it provides a single uniform user interface capable of showing all devices and conflicts in the same window.

The goal is to have just one button to configure the whole LAN automatically. When this button is pressed, the software analyzes all network and devices and automatically configures them with the most optimal settings.

http://www.globaliptel.com

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Sona Mobile Rolls Out BlackBerry Media Player Application

By Laura Stotler

Sona Mobile Holdings (news - alert) is launching a BlackBerry Media Player software application, designed to offer multimedia applications on the latest generation of RIM devices. The new application will offer near TV quality layback of synchronized video and audio files.

Sona Mobile provides secure mobile solutions for access to financial and enterprise information as well as gaming and entertainment applications. The company's technology enables delivery of a rich client experience with top performance and security.

The Sona Wireless Development Platform architecture and 3D mobile methodology work across a wide range of mobile devices and operating systems. These include Research In Motion's BlackBerry and Microsoft Corp's Windows Mobile lines.

"For the very first time, BlackBerry users can receive either BerryCast (PodCasts wirelessly updated) or streaming video on their mobile devices," said John Bush, CEO and president of Sona Mobile. "RIM customers take advantage of a download-and-play method of delivering multimedia files to BlackBerry devices. We believe that this application will be well-received in the marketplace."

http://www.sonamobile.com

Pandora and ESPRE Deliver Video Conferencing Solution

By Mae Kowalke

Media technology (news - alert) company ESPRE Solutions, Inc. (news - alert) announced a licensing and integration agreement with IP communications service provider Pandora Networks for the delivery of a video conferencing solution.

The agreement integrates video products from both companies: ESPRE's eViewChat and Pandora's Worksmart On-Demand IP Communications solution.

eViewChat is an 8-way video conferencing solution. Worksmart is a software suite that gives small and medium-sized businesses control over communications services including voice, video, messaging, and collaboration.

Integrating the two products means that Worksmart users will be able to "take advantage of ESPRE's proprietary video compression to deliver business-quality video communications at far less bandwidth than that of competing video solutions," the companies said.

The integration includes ESPRE's virtual private network solution, eViewNet, allowing users "to operate through corporate firewalls in order to make seamless video conferencing connections without the need to involve IT or other technical resources," the release stated.

http://www.espresolutions.com
http://www.pandoranetworks.com

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XO Holdings Launches Nextlink

By Johanne Torres

XO Holdings (news - alert) announced that it launched Nextlink, (news - alert) a new broadband wireless service provider that will offer services to mobile and wireline operators, businesses, and government agencies. Using licensed wireless spectrum covering 75 metropolitan markets across the United States, Nextlink will offer customers broadband wireless services as an alternative to conventional broadband services delivered over copper.

Nextlink's services will first be available in Dallas, Los Angeles, Miami, San Diego, Tampa, and Washington, DC, with additional market launches over the next two years. Nextlink is currently providing broadband wireless services to a national wireless company, delivering wireless backhaul and network redundancy and diversity services across markets in south Florida. In conjunction with the launch of its services, Nextlink also announced today the selection of Hughes Network Systems, LLC as its strategic wireless equipment supplier.

Nextlink's services are "fixed wireless" broadband offerings that rely on licensed local multipoint distribution system (LMDS) wireless spectrum in the 28GHz - 31GHz range. For locations up to seven miles and in line-of-sight of a Nextlink wireless hub, Nextlink provides wireless broadband services with speeds from 1.544 Mbps (T-1) up to 622 Mbps (OC-12). Nextlink's services include wireless T-1, wireless metro Ethernet, and wireless dedicated Internet access.

http://www.nextlink.com
http://www.xo.com

Seamless IP Networking Across the Great Lakes from Rogers, Sprint

By Erik Linask

Rogers, (news - alert) Canada's largest wireless voice and data communications services provider, has signed an agreement with Sprint (news - alert) allowing business customers with locations in both countries to operate on a seamless IP-based network through a new MPLS (multi-protocol label switching) network-to-network interface (NNI).

The newly founded relationship allows Rogers' customers to seamlessly connect to the Sprint network and enjoy a consistent communications experience north or south of Niagra Falls. Technology, SLAs, and support will all be seamlessly exchanged between the two providers.

Rogers North American MPLS delivers both the flexibility and reach of the Internet and the security and performance of a private network. Business customers have secure connections offices, customers, partners, and suppliers, but do not have to purchase, operate, and maintain additional hardware — which makes for an efficient, cost-effective solution.

The advantage of MPLS solutions is that companies will be able to avoid many capital expenditures and additional hires to maintain networks on both sides of the border. Cross-border offices will be able to operate seamlessly on an IP-based network, experiencing seamless connectivity with security, redundancy, and quality of service.

http://www.sprint.com
http://www.rogers.com

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Gizmo Project Now Includes VoIP Conferencing

By Erik Linask

Gizmo (news - alert) Project uses your Internet connection to make calls to other computers. With the click of a mouse, you're connected to friends, family, and colleagues anywhere on earth. You talk clearly for as long as you want — for free.

To address growing demands for conferencing capabilities, Gizmo Project has enhanced its VoIP services by tapping Vapps, a provider of VoIP conference calling solutions, for its conferencing platform in order to offer free worldwide conference calling services to Gizmo users.

Vapps (news - alert) provides a carrier-class system for service providers and large enterprises to deliver high-quality reservationless, attended and operator assisted conferencing solutions. The solution is based on the company's Conference Bridge 1000 (CB1000), a SIP-based conferencing platform that offers customized conferencing on a customer by customer basis both on legacy TDM and newer IP-based telecom systems.

With the Vapps VoIP conference calling solution, Gizmo subscribers can connect multiple users over the Internet for cost-effective and reliable conference calls. All the features of a traditional conference calling service are available through Gizmo's Web site.

http://www.gizmoproject.com
http://www.vapps.com

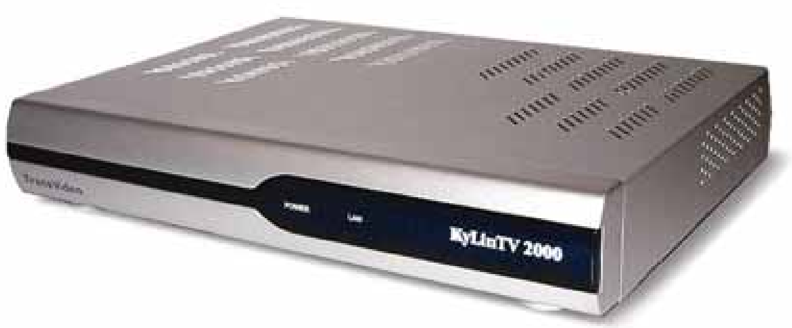
IPTV Goes Straight to the Screen

By Erik Linask

NeuLion (news - alert) is expanding video possibilities with its IPTV, going beyond the desktop, straight from the Internet to the TV. The NeuLion iPTV platform is an end-to-end solution that uses the public Internet to stream multimedia content to any TV or PC from a common library. The result is DVD-quality streaming video — in real-time — to any device.

The NeuLion iPTV platform encodes, delivers, stores, and manages an unlimited range of multimedia content, and the Operational Support System (OSS) maintains all billing and customer support services. The service delivers high quality video to the home or business over an existing high speed Internet connection, allowing viewing of high quality content, just as if it were being delivered by a local cable provider.

But, without traditional geographical limitations, NeuLion's platform can span the globe, connecting niche audiences and creating communities connected by common interests. What's more NeuLion's set-top box also supports interactive features, from VoIP connections to gaming and more.



The key is that the NeuLion platform is both platform- and vendor-neutral — it can deliver video to any platform or any media device, using surprisingly low bandwidth rates.

NeuLion's content partners supply media and identify the marketplace. The NeuLion iPTV Platform encodes, delivers, stores, and manages an unlimited amount of multimedia content, and the Operational Support System (OSS) maintains all billing and customer support services.

http://www.neulion.com

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AT&T Delivers Content to Broadband Customers

By Johanne Torres

AT&T (news - alert) and Starz Entertainment Group have joined forces to offer SEG's Vongo Internet movie delivery service to AT&T High Speed Internet customers. The service delivers movies and other video content over the Internet for playback on Windows-based PCs, laptops and select portable media devices as well as on TVs.

The new agreement will feature a co-branded AT&T and Vongo Web site with a special 14-day free trial offer to AT&T High Speed Internet subscribers. The companies will also market the Vongo service on the AT&T Worldnet portal.

"Vongo's compelling content increases the value proposition for AT&T High Speed Internet customers," said Scott Helbing, chief marketing officer-AT&T Consumer in a statement. "With Vongo, we're positioned to deliver quality content, as we build a digital lifestyle platform for our customers."

http://www.thenewatt.com

CTI Selects Netrake To Launch Caribbean VoIP Services

Netrake (news - alert) announced that VoIP carrier CTI (news - alert) has selected Netrake to support its launch of VoIP services throughout the Caribbean region. Netrake's nCite session border controller provides CTI with secure traversal of firewall and network address translation (FW/NAT) systems, as well as denial of service (DOS) attack prevention through deep packet inspection at both the SIP signaling and VoIP media layers. Netrake's session controllers also provide end-to-end quality of service (QoS) assurance and SLA enforcement for each call across network borders.

CTI employs Netrake's nCite session border controllers to provide enterprise and wholesale VoIP services across the United States, the Caribbean and Latin America. Based in Miami, Fla., CTI is offering lower cost yet feature rich residential & enterprise VoIP services. Additionally, CTI will leverage its extensive IP network to offer termination services to local carriers around the region, thereby providing those carriers with competitive pricing. CTI will also be able to offer users their own Virtual Private Network (VPN) for delivering calls on and off of the public switched telephone network (PSTN), such as 'landing' calls from one nation's PSTN to another nation's PSTN, as well as utilizing a unique four (4) digit dialing plan world wide.

http://www.ivoiptech.com
http://www.netrake.com

American Cable and VoX Offer Triple-Play Bundle

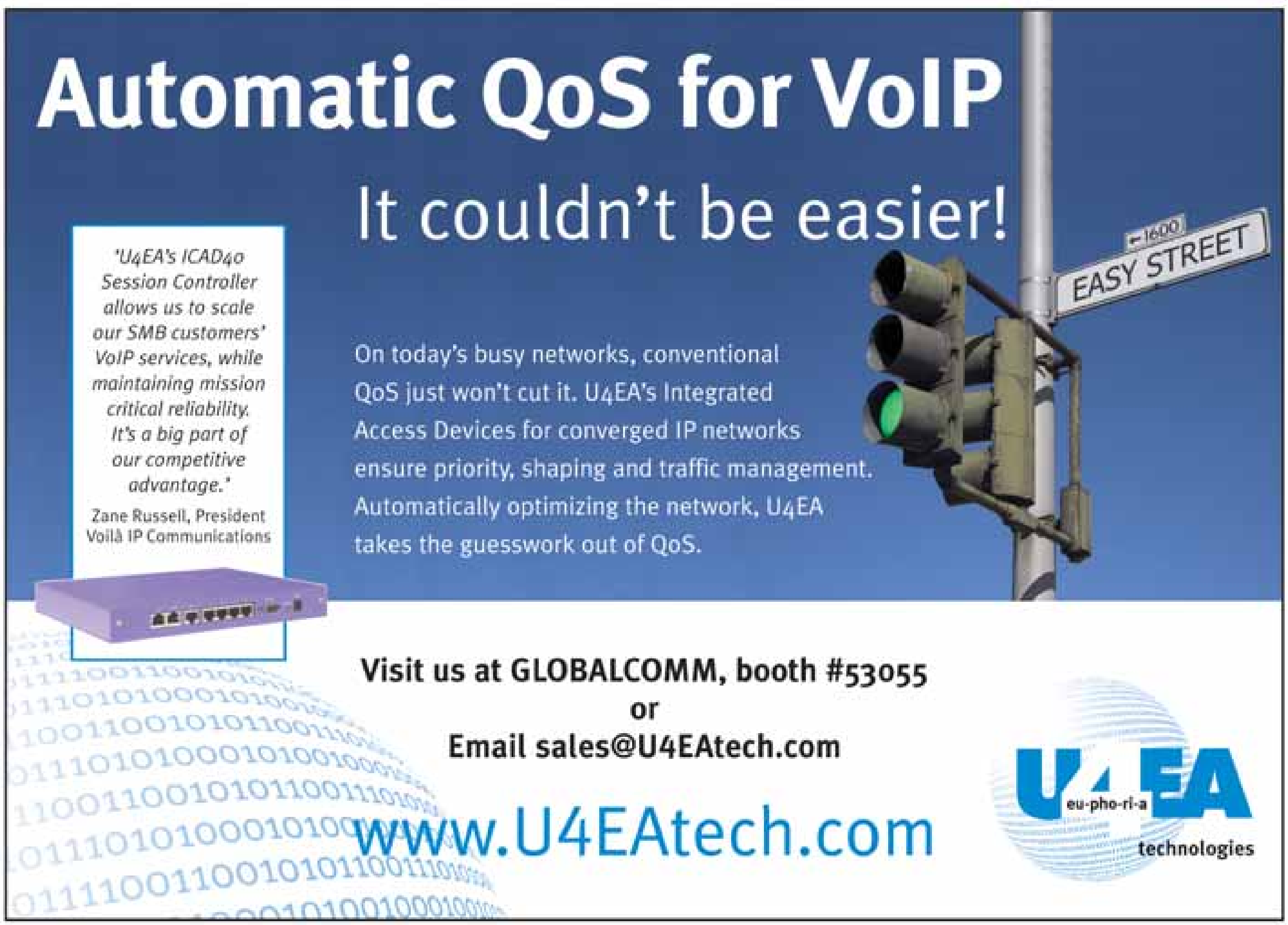
By Johanne Torres

Wholesale VoIP service provider VoX Communications (news - alert) announced that Florida cable operator American Cable Services (news - alert) will deploy VoX services as the broadband phone component of a new triple-play offer, which also includes cable TV and high speed Internet. The offering will first be available in Little Harbor, a coastal resort community of 2,300 residences in Ruskin, Florida.

"We've made a strategic decision to become the preferred broadband phone solution for cable operators nationwide. There are more than 1,100 independent cable operators like American Cable and many of them are weighing plans for phone services. This contract represents a large and meaningful market opportunity for VoX," said VoX's president and CIO Mark Richards.

American Cable is contracted for the next several years to provide the bundled services to approximately 150,000 multi-family residence units in the sunshine state, many of which will have a "mandatory phone" under their Home Owners Association covenants. The integrated offering becomes an amenity that is paid for as part of the resident's monthly dues.

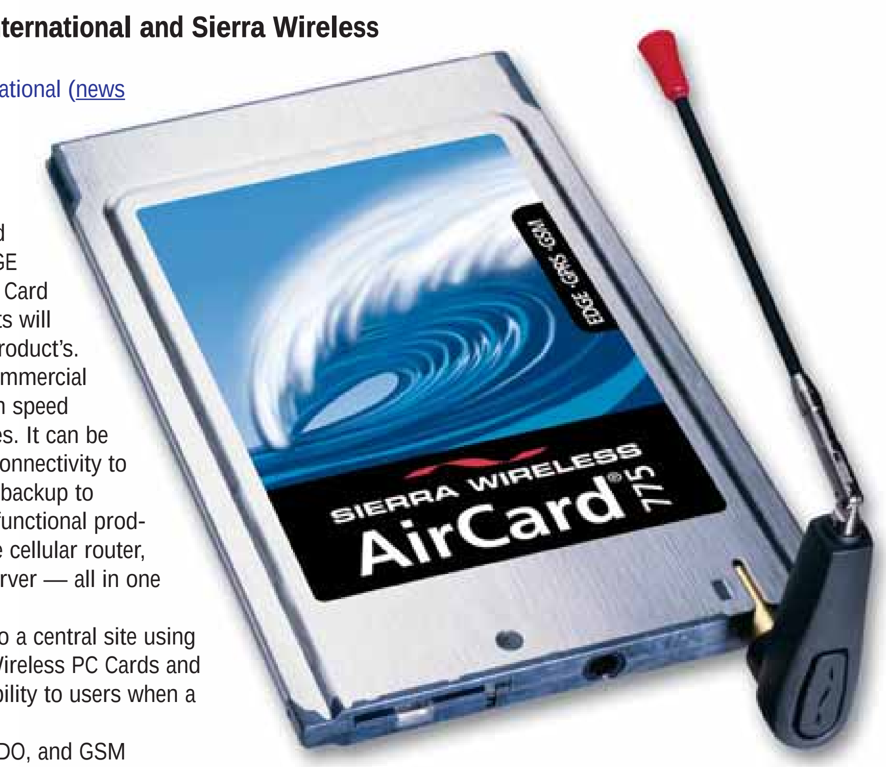
http://www.americable.us
http://www.voxcorp.net



3G Wireless Connectivity from Digi International and Sierra Wireless

By Erik Linask

Sierra Wireless (news - alert) and Digi International (news - alert) together will provide 3G wireless connectivity for customers using Digi's ConnectPort WAV VPN routers. Specifically, Digi, a networking solutions developer, will be offer compatibility with Sierra's AirCard 860 and AirCard 850 HSDPA PC Cards, AirCard 775 EDGE PC Card, and MC5720 EV-DO PCI Express Mini Card embedded module. The Sierra Wireless products will provide 3G wireless connectivity Digi's router product's.



Digi's ConnectPort WAN VPN product is a commercial grade, upgradeable 3G router that provides high speed wireless connectivity to remote sites and devices. It can be used for primary wireless broadband network connectivity to equipment at remote locations, as well as for a backup to existing landline communications. It is a multi-functional product, able to perform multiple network tasks, like cellular router, firewall, switch, VPN appliance, and terminal server — all in one device.

The ConnectPort router enables connection to a central site using a wireless wide area connection — via Sierra Wireless PC Cards and embedded modules — offering maximum flexibility to users when a wireline connection is not available.

ConnectPort WAN supports both CDMA, EV-DO, and GSM HSDPA/UMTS 3G networks. Currently, the Digi ConnectPort WAN VPN is certified by Cingular and Sprint, with certifications with additional service providers expected soon.

The AirCard 860 and AirCard 850 wireless WAN cards are Type II PC Cards that can be conveniently stored inside the laptop. The AirCard 860 utilizes the 850 and 1900 MHz UMTS frequency bands while the AirCard 850 utilizes the 2100 MHz UMTS frequency band and is targeted primarily for use in Europe. Both offer average data rates of 500 to 800 kbps, with bursts over 1 Mbps on HSDPA networks. In areas where HSDPA or UMTS network coverage is not available, the AirCard 860 and AirCard 850 will connect to EDGE and GSM/GPRS networks.

The AirCard 775 fits into a laptop's PCMCIA slot and provides data rates averaging 100 to 130 kbps, with peak rates up to 216 kbps. The AirCard 775 card also supports global roaming, functioning on EDGE and GSM/GPRS networks on all four GSM frequency bands used worldwide.

The dual-band MC5720 PCI Express Mini Card, built using the MSM6500 Mobile Station Modem chipset from QUALCOMM, offers typical download data rates of 400-700 kbps, with peak speeds up to 2.4 Mbps. To provide coverage in areas where an EV-DO network connection is not available, the MC5720 module is also compatible with widely available 1X networks and will seamlessly switch without interrupting the user.

http://www.digi.com
http://www.sierrawireless.com

Convergence, meet the new guy.

Finally, the world has a truly converged work environment. The OfficeServ™ 7200 platform from Samsung's Business Communication Systems integrates voice, data, VoIP, analog, digital and LAN and delivers them all via wireline and wireless technologies. With wireless in the mix, employees can remain connected, efficient and productive throughout your business. Even when they're away from their desks. Now that's convergence. Complete convergence.

All hail the new guy.

For more information visit **www.samsungbcs.com/OS7200**

SAMSUNG

Venturi Wireless Upgrades PetroCom's Gulf of Mexico Network

By Erik Linask

Venturi Wireless, (news - alert) which is focused on providing solutions that optimize the performance of wireless data networks for mobile operators, announced that mobile carrier PetroCom (news - alert) has deployed Venturi's optimization solution within its Gulf of Mexico network.

Venturi Wireless' Adaptive Airlink Optimization is a cross-layer technology goes beyond simple application layer compression to virtually eliminate the inefficiencies of TCP at the transport layer. Thus, it can provide mobile data users faster access and a generally better experience — it can deliver access speeds up seven times higher with improved reliability and consistency.

The patented Venturi Transport Protocol (VTP) lies at the heart of the solution. It was originally developed to cope with satellite data transmission issues, and has since been modified and transformed into a wireless solution. It overcomes the high latency and packet loss issues that result from poor RF conditions caused by weakened signal strength, interference, fading or high load. In addition to a better end user experience, Venturi Wireless optimization also maximizes spectrum and network resource utilization.

http://www.venturiwireless.com
http://www.petrocom.com

WildPackets Intros WiFi RF Environments Analyzer

By Johanne Torres

VoIP analysis technology provider WildPackets (news - alert) announced the OmniSpectrum, a portable RF spectrum analyzer that runs in a standard Windows laptop PC and identifies the devices causing interference on a WiFi network.

OmniSpectrum extends the existing capabilities of the OmniAnalysis Platform by making the 802.11 physical layer visible and intelligible, enabling network engineers to see which devices, WiFi and non-WiFi, are causing interference.

OmniSpectrum identifies 802.11 devices, others that are generating signals in the 802.11 frequency band, and those that are unauthorized or transient. The system also identifies the class of the interfering device, like a cordless phone, and the manufacturers and model numbers of the offending equipment. This facilitates problem identification and resolution.

WildPackets' OmniAnalysis Platform is a distributed network analysis system for optimizing network services and maximizing uptime on enterprise networks. The platform uses analytical techniques, including network forensics and application performance indexing, to troubleshoot network problems.

http://www.wildpackets.com

SOHOware Announces Outdoor WiFi Solutions

SOHOware, (news - alert) an integrated wireless systems pioneer, announced new outdoor wireless solutions that help WISPs and other Service Providers quickly deploy public Internet access at multi-dwelling (MDU), hotel, and resort properties. The AeroExtend family further enhances SOHOware's integrated MDU solutions by adding dual-band outdoor wireless capability, which combines robust 5.8GHz backhaul trunks with simultaneous 2.4GHz WiFi access.

There are two radios in the AeroExtend device, providing dedicated bandwidth to the access point and wireless backhaul. The dual-radio, dual-band platform offers multiple benefits; AeroExtend installations require less number of devices overall per deployment, providing a more visually discrete solution and reducing initial capital investment for cost sensitive property owners.

The AeroExtend family includes the WLG2502 dual radio AP-Bridge and a range of complimentary high gain antennas for 2.4 and 5GHz bands. Wireless network modes include point-to-point and point-to-multipoint bridging, plus simultaneous access point. The WLG2502 includes Power over Ethernet support and complete mounting accessories. Performance is 54Mbps for 11a and 11g, with turbo mode for 108Mbps.

http://www.sohoware.com

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Broadcom Ships Legacy-Friendly 802.11n WiFi Solutions

By Erik Linask

Broadcom Corporation, (news - alert) a producer of semiconductors for wired and wireless communications, has begun shipping its Intensi-fi wireless LAN technology to several top manufacturers of wireless networking gear. The Broadcom Intensi-fi draft-802.11n solutions provide increased performance wireless connectivity. More importantly, they ensure compatibility among next generation products from various manufacturers.

Broadcom's Intensi-fi chipsets combine high quality radio and digital technologies to deliver increased levels of wireless performance and reliability in the home or in the office. Intensi-fi's throughput of more than 180 Mbps enable consumers and businesses to take advantage of emerging voice, video, and music applications on their wireless networks. Because of its performance and compatibility, Intensi-fi technology is expected to extend beyond PCs and networking equipment into consumer devices to enable an entirely connected digital home. Look for Intensi-fi technology to make its way into WiFi phones, storage drives, set-top boxes, broadband modems, TV and stereo equipment, gaming systems, and digital cameras.

Not only is its Intensi-fi technology interoperable with a wide variety of 802.11a/b/g and draft-802.11n devices, but it can even improve the performance of legacy WiFi products. This is music to the consumers' ears, for it adds value to their previous investments and protects them from immediate obsolescence. Intensi-fi routers employ a "good neighbor" mode that ensures optimal performance in mixed networks when using the optional 40 MHz transmission mode, dynamically managing the use of 40 MHz channels and reverting back to 20 MHz channels when traffic is high or legacy clients need to communicate.

http://www.broadcom.com

ARRIS and UTStarcom Team on FMC Solution

By Mae Kowalke

Converged network solutions provider ARRIS (news - alert) announced today a three-part, wireless-related agreement with end-to-end networking solutions provider UTStarcom. (news - alert) The agreement — which covers development, licensing, and supply — will enable "the fourth leg of the quadruple play for cable Multiple System Operators (MSOs) worldwide," according to ARRIS.

ARRIS and UTStarcom's joint solution will enable seamless roaming between cellular and WiFi connections for end users with dual-mode handsets.

With the agreement, ARRIS' FMC solution—which enables MSOs to add wireless telephony to their service offerings—teams up with UTStarcom's Continuity FMC solution, designed "to increase network efficiency and coverage through either wireline or wireless networks," ARRIS said in the press release.

ARRIS will license UTStarcom's FMC software for an initial term of three years.Both companies have specified rights to create new features for emerging FMC solutions. Finally, UTStarcom will provide ARRIS with hardware and software to sell to cable MSOs. ARRIS is granted "exclusive marketing rights to the global MSO community" and UTStarcom retains "exclusive marketing rights to the telco community."

http://www.arrisi.com
http://www.utstar.com

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BroadVoice Intros WiFi/Cellular Phone

By Johanne Torres

VoIP service provider BroadVoice (news - alert) announced that it will begin selling a wireless VoIP/GSM phone, code-named "Falcon," this summer. The BroadVoice Falcon will enable users to use the company's unlimited VoIP calling plans utilizing WiFi networks. When the BroadVoice user is outside WiFi coverage, the Falcon switches to a cellular network and works as a mobile phone using a separate account with a GSM service provider.

"Whether you're at home, at work, or at a coffee shop, airport, hotel, or other location with a public hot spot, you can use your BroadVoice Falcon phone to make unlimited calls anywhere in the United States, and up to 35 countries, without paying for minutes," said Gene Cornfield of BroadVoice. "When you're on the move and WiFi isn't available, the Falcon automatically uses the GSM cellular network, just like a mobile phone."

Once a customer buys a Falcon phone with BroadVoice service, they can select and activate a telephone number online and begin making and receiving VoIP calls over WiFi immediately. As soon as the customer plugs in a standard SIMM card from any GSM900, GSM1800, or GSM1900-based cellular carrier with whom he has an account, he can place cellular calls without having to contact the cellular service provider. Calls are received on either the user's BroadVoice number, or the number assigned by his cellular carrier.

http://www.broadvoice.com

TANDBERG 3G Mobile Video Applications Drive Revenues

Assisting service providers in delivering video content and interactive videoconferencing to mobile consumers, TANDBERG (news - alert) introduced 3G video capabilities that enable service providers, including mobile operators and content providers, to differentiate their offerings and maximize their revenues.

With the TANDBERG Video Portal, service providers manage and stream live or archived video content such as news reports, sports highlights, and movie previews to 3G mobile users. The intuitive drag-and-drop interface makes it easy for a service provider to build a customizable user experience that 3G mobile users navigate to quickly access content over their mobile phones. Service providers seamlessly manage content from multiple content providers with the Video Portal, and content usage statistics are easily downloadable for invoicing and statistic purposes. In addition, anyone can record content to the Video Portal for applications such as real estate services and video dating from any 3G mobile phone, or a SIP or H.323 device. Furthermore, 3G content can be live streamed to the internet or television studios.

Videoconferencing is another innovative mobile video application for 3G mobile users. Whether responding to emergency situations, remote troubleshooting or participating in multiple-site business meetings, 3G mobile users have the ability to interact face-to-face instantly from anywhere. With the TANDBERG 3G Gateway, service providers can offer mobile videoconferencing to their customers using a scalable and redundant solutions built to integrate with carrier infrastructures using E1/T1 or SS7. Diverse billing rates based on premium numbers or video short codes give service providers a flexible solution for maximizing call revenues.
http://www.tandberg.net

Gig-E and PoE Stackables from Alcatel

By Erik Linask

Alcatel (news - alert)has introduced its OmniSwitch 6850 family of 10 Gigabit-capable stackable 10/100/1000 and 10/100/1000PoE workgroup switches. The OmniSwitch 6850s are the first Alcatel switches to provide Gigabit power over Ethernet (PoE) capabilities with extensive port diversity, full Alcatel Operating System (AOS) support, Gigabit to the desktop, 10 Gigabit uplinks, and enhanced performance for a flexible, intelligent, and highly available network.

The OmniSwitch 6850L models provide Fast Ethernet to the desktop and are software upgradeable to Gigabit. This latest release continues Alcatel's strategy to provide enterprises a cost-effective means for migrating to Gigabit Ethernet desktop connections through a software key.

The OmniSwitch 6850s and 6850Ls protect enterprises' investments by providing for future needs, such as advanced edge security, support for user mobility and dual IPv4/IPv6 capability. The switches' versatility afford customers the flexibility to buy only what they need today without sacrificing performance they will require in the future. They also will benefit from the wire-rate layer 2 switching and layer 3 routing supporting optimal QoS for voice and video.
http://www.alcatel.com

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JDSU Tackles High Capacity Networks

By Cindy Waxer

JDSU (quote - news - alert)has announced that its IP network troubleshooting and data analysis platforms, DA-3400 and DA-3600A, can now perform VoIP call quality monitoring on high capacity networks. The DA-3400 is able to support 8,000 simultaneous calls and the DA-3600A can support 64,000 calls to provide accurate and high-quality measurements when network load or utilization is at its highest.

The enhanced DA platforms also generate important industry metrics used to help ensure VoIP voice quality, such as MOS and R Factor. These can be measured for all calls on a gigabit Ethernet circuit that is being utilized 100% by VoIP traffic. Other features include display filtering, display segmentation, and users configuration of the DA line for simple network management protocol or e-mail notification so they are alerted to any network quality issues quickly.

Part of JDSU's Service Assurance Solutions portfolio, the DA-3400/3600A's new VoIP features also include the ability to show signaling details of VoIP calls using a graphic display that captures signal/packet set-up and tear-down during a VoIP call exchange. This feature also details the timing and response code, delivering prompts that warn the user the VoIP signal did not go through.

http://www.jdsu.com

Acme Packet Intros SBC Features

By Johanne Torres

Session border control provider Acme Packet (news - alert)introduced additional features and a partnership to offer a suite of session border controller capabilities to enable cable voice, video, and multimedia services.

The company now supports an ENUM server interface and has interoperability in field deployments with Nominum's Navitas, an ENUM-based IP-application routing directory server. Acme Packet is also expanding its hosted NAT traversal mechanisms with support for the IETF standard STUN (Simple Traversal of UDP through NATs), STUN-Relay/TURN (Traversal Using Relay NAT) and ICE (Interactive Connectivity Establishment).

Acme Packet is currently in commercial production at 4 cable operators in the U.S. and at some cable companies in Europe. For these operators, Acme Packet's Net-Net session border controllers provide access to the network edge and interconnect borders to protect and secure the MSO service infrastructure.

Acme Packet's ENUM server interface for the Net-Net SBC platforms supports private and public ENUM directory services and enables MSOs to use IP transport end-to-end for VoIP calls between subscribers on different networks. This eliminates any PSTN transport.

http://www.acmepacket.com

Elma Announces New 1U EasyPlug Compact PCI Chassis

Elma Electronic Inc., (news - alert)a global manufacturer of electronic packaging products, announced a new Type 39 1U EasyPlug CompactPCI enclosure. The Type 39c HA line features 9U cPCI backplanes from Elma Bustronic, which include pluggable 47-pin connectors for hot swapping power supplies. There are also pluggable fan tray headers and optional shelf management modules. Redundancy options can be built-in with all of these components. The backplanes are available in standard cPCI, H.110, or PICMG 2.16 options.

With a sheet metal design and full pluggability, the Type 39c HA family offers ease of manufacturing and saves costs. The chassis are available in 1U-4U heights in horizontal-mounting orientations. Compliant to the latest PICMG specifications and IEEE 1101.10/.11, the enclosures feature side-to-side 200CFM (300LFM) cooling, 300mm depths, and rear I/O options.

http://www.elma.com

Trenton Introduces System Host Boards With Dual-Core Processors

Trenton Technology (news - alert)has recently released several new PICMG 1.3 system host boards (SHBs) featuring dual-core processors from Intel.

Trenton's SLT is a server-class SHB that has two, Dual-Core Intel Xeon Processor LV 2.0GHz CPUs resulting four independent processing cores on a single system host board. The advanced power management capabilities of this new processor platform maximizes power efficiency, lowers system thermals and enables efficient high-density clustering of SHBs in a wide variety of telecom applications.

Trenton's TML is a graphics-class SHB featuring a single, long-life Intel Core Duo Processor T2500. This dual-core SHB supports a x16 PCI Express electrical link to a PICMG 1.3 backplane that enables full link support for the latest x16 PCI Express graphics and video cards. Another key feature of the TML is the ability to support RAID 0, 1, 5 or 10 implementations of the four, onboard SATA/300 interface ports.

Both class of system host boards support multiple PCI Express links from the SHB to a PICMG 1.3 backplane. These high speed, high bandwidth PCI Express links provide multiple communication pathways to and from the SHB that are capable of supporting a vide variety off the shelf PCI, PCI-X and PCI Express option cards.

http://www.trentontechnology.com

Globecomm Unveils SkyBorne IPTV Regional Headend

By Patrick Barnard

Globecomm Systems (news - alert)recently introduced its SkyBorne IPTV Regional Headend (RHE) for telephone companies and broadband carriers looking to add video services.

The latest in the company's line of pre-engineered systems, SkyBorne RHE provides content acquisition, content management, subscriber management, content packaging, and delivery to the carrier's distribution network — all in a single, cost-effective, customizable package. Optional revenue-generating features include systems for advertising insertion and video-on-demand.

According to Globecomm, SkyBorne RHE, which is based on field-proven modular building blocks, provides the performance, reliability and operational integrity of a custom broadcast system at a competitive price. It can adapt quickly and easily to changing customer needs. In addition to the Regional Headend, SkyBorne also includes lifecycle support services such as network monitoring, installation, field service and repair to preserve the value of the original investment and provide maximum uptime and throughput.

http://www.globecommsystems.com

Minacom and Electroline offer VoIP, Video, & Internet Testing

Minacom, (news - alert)a VoIP test solution provider for Cable MSOs, announced that its DirectQuality R7 Service Level Test Automation platform now supports remote testing to Electroline cable transponders, providing advanced VoIP, IPTV, Internet, and Fax over IP service quality and reliability assurance throughout the hybrid fiber-coax (HFC) network. Electroline status-monitoring transponders support non-intrusive IP-loopback, in addition to packet, RF, and power supply performance and reliability monitoring.

With this announced interoperability, operators can now use Minacom's PowerProbe service level test probes running the RTP Streamer test agent to perform loopback tests to Electroline transponders supporting the SMRP protocol. Loopback testing provides a complete, concise service and network performance assessment by measuring bi-directional VoIP speech quality (MOS, R-factor), packet jitter, delay, loss, and other IP impairments, without interrupting transponder service transmission functions. Supporting Minacom's award-winning E-Model implementation, the RTP Streamer test agent allows operators to simulate triple-play traffic patterns with configurable codec, IP precedence, VLAN, jitter buffer and other key parameters to accurately replicate services and provide user-perceived quality of service (QoS) metrics.

http://www.minacom.com

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Boise State Adds SIP Trunking to VoIP Network

By Erik Linask

Time Warner Telecom (news - alert) has announced the successful installation of its SIP IP connections for VoIP services to Boise State University — what TWT says is the largest VoIP deployment in an American educational environment.

The installation will entail running VoIP technology over Boise State University's existing campus-wide metro Ethernet service — a Cisco Powered Network that encompasses more than 14,000 telephone numbers and 4,000 handsets.

Time Warner Telecom's 20 Mbps-capable SIP trunk service replaces the University's existing T1s to boost bandwidth by nearly 20%. The scalable SIP installation will allow IT managers to connect directly to a VoIP PBX, thereby allowing the school to reduce the number of gateways it needs by six — all of which were required previously to convert digital voice signals into IP.

TWT's SIP trunking solution overlays the existing Ethernet LAN platform and is designed to seamlessly connect with existing Cisco VoIP applications. It is also highly scalable, which means that, as needs change, the school can expand the system again with relative ease and no disruption.

SIP trunking entails using Session Initiation Protocol (SIP) for call control and routing, which enables a single IP-only connection to the carrier. Voice simply becomes another application across the IP network. In theory, this results in a pure IP to IP call if both end users are using SIP trunking.

http://www.tetelecom.com

Verso Announces Availability of I-Master Application Server

Verso Technologies, (news - alert) a global provider of next generation network solutions, announced the introduction and general availability of the I-Master Application Server, an enhancement to the I-Master application.

The I-Master application adds value to both vertical and horizontal markets, offering a standards-based interface using SIP to third-party softswitches and RADIUS interface upstream for AAA (Authorization, Authentication and Administration) functions.

The offering is a real time call control and service logic solution with embedded IVR functionality supporting a wide array of call flows. A user-friendly Graphical User Interface (GUI) enables operators to easily create and manage customized call flows with multi-language support.

The combined deployment of Verso's I-Master Revenue Assurance application and the application server enables real-time authentication, rating and call control for calling card, fixed line ADSL, dial-up, WiFi, and enhanced IVR services, resulting in increased customer loyalty and revenue generating ability for any next generation network (NGN) environment. Additionally, the I-Master Application server supports account balance preauthorization to enable simultaneous concurrent service usage on one account when deployed with the I-Master Revenue Assurance Platform.

http://www.verso.com

Fusion Files Patent Application for its VoIP

Fusion Telecommunications (news - alert) announced that it filed a patent application with the United States Patent Office for its Directed SIP Peer-to-Peer ("DSP") technology, acquired by Fusion in February 2006. The patent application describes a system that Fusion plans to utilize to provide its free service between SIP devices.

Fusion is incorporating its DSP technology into the Company's international network for Internet voice calls between any combination of computers, Internet connected telephones, wireless devices, and other SIP-enabled hardware. Fusion believes its new "efonica" branded softphone and uniquely configured VoIP network will provide significant advantages over most VoIP peer-to-peer networks. Fusion's technology eliminates the method of routing utilized by many VoIP peer-to-peer networks, in which many users' Internet bandwidth and/or PCs are utilized as part of the carrier's larger network to set up calls for thousands of other users.

"Our DSP technology should be of particular interest to security conscious individuals and businesses, a fast-growing segment as SIP is quickly becoming the de facto VoIP standard for communicating between VoIP hardware devices. We believe our entire communications package should be particularly advantageous for Fusion's primary target of the emerging markets of the Middle East, Asia, Latin America, Africa, and the Caribbean, and their related communities of interests," said Matthew Rosen, Fusion's President and CEO.

http://www.fusiontel.com

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InterTel's New 7000 Communications System

By Laura Stotler

The new Inter-Tel (news - alert) 7000 system can scale up to 2,500 users per site and is designed as a pure, standards-based communications platform. Using SIP technology at its core, the system is redundant and secure and offers an easy-to-use interface for remote management and configuration.

The new platform will also offer full PBX-style features as well as a number of enhanced features like embedded presence management and mobility. It is designed to support Inter-Tel's advanced IP-powered applications like contact center and messaging solutions, collaboration and the company's existing lines of multi-protocol IP and SIP endpoints.

"Inter-Tel has done a masterful job in developing a platform that is being designed to successfully leverage advanced IP technology to provide a rich, intuitive feature set that can be an asset to any business," said Mark Ricca, a partner with Intellicom Analytics.

"Standards-based software solutions, with SIP at the core, will enable organizations to avoid obsolescence by leveraging the communication options that Internet standards make possible," noted Allan Sulkin, president of TEQConsult Group.

"The fact that the Inter-Tel 7000 delivers presence capabilities as part of its core feature set differentiates it from other solutions in the market," said Rob Arnold, an industry analyst with Current Analysis. "This is a real incentive for businesses looking to leverage these tools without having to add servers or additional software."

http://www.inter-tel.com

3 Rivers Chooses Pannaway's Convergence Network Solution

By Patrick Barnard

Pannaway Technologies (news - alert) has won a major deal with 3 Rivers Communications, which will be using Pannaway's broadband access system for the delivery of new next-generation services — including IPTV — to its rural subscribers.

The cooperative will use Pannaway's ADSL2+ and Active Ethernet FTTH solution to deliver a robust set of triple play features, along with the necessary bandwidth capacity to add future serives. Pannaway's copper and fiber-based access solution will deliver SIP-enabled triple play, including Primary Line VoIP for guaranteed Lifeline calling and E 911.

Broadband Aggregation Routers (BARTM) will be deployed in the cooperative's central office (CO) for scaleable 1 to 10 Gigabit Ethernet transport, while Broadband Access Switches (BASTM) will reside in remote terminals (RT) for high-performance last mile voice, video and data delivery.

The ILEC will also use Pannaway's Broadband Access Manager (BAM) to simplify the deployment and provisioning of its new copper and fiber-based network.

The use of IP-Ethernet and SIP allows Pannaway to deliver improved video quality with enhanced rate/reach benefits. Enhanced network performance coupled with 10Gbps transport scalability ensures that telcos' networks are future-proofed, that they will have the performance and capacity to support emerging bandwidth-intensive services.

http://www.pannaway.com

Siemens HiPath ProCenter Helps Improve First Contact Resolution

Siemens Communications (news - alert) announced HiPath ProCenter Enterprise Version 7.0, a new Internet protocol-ready contact center solution designed to help enterprises improve first contact resolution, drive up productivity and increase customer satisfaction. The solution includes pre-built integrations with front-office customer relationship management (CRM) applications from Microsoft Corp., SAP AG and Siebel Systems.

The Siemens HiPath ProCenter Enterprise solution helps enterprises improve the efficiencies of multiple customer interaction channels — including voice, e-mail, and live Web interactions — with Siemens' award-winning presence and collaboration tools. With presence-driven applications, front-line agents can get real-time information about the availability of subject matter experts and connect to them across various media types throughout the enterprise.

"Contact center industry research continues to demonstrate that first contact resolution is a key driver of customer loyalty, revenue growth and operating cost effectiveness," said Al Baker, vice president, Enterprise Division, Siemens Communications Inc. "This presence-driven solution makes first-contact resolution possible even for highly complex or sensitive customer interactions."

The solution's presence and collaboration tools help drive first contact resolution via Team List and Team Bar features that enable agents to view real-time presence and availability states of peers, managers, and experts outside the contact center. Available users can be included in call transfers, conferences, or consultations with just a mouse click.

http://www.siemens.com

CRM Solution Helps Target Interact Increase Customer Satisfaction

By Erik Linask

Management consulting firm Target Interact (news - alert) recently chose UCN Inc.'s (news - alert) solution for enhancing the reliability and efficiency of its call center operations. UCN provides on-demand contact handling software and telecommunication services over its national VoIP network. InContact is a hosted solution set designed to significantly enhance caller satisfaction, boost agent productivity, and improve overall profitability. Features include interactive voice response (IVR), skills-based ACD routing, computer/telephony integration (CTI), inbound/ outbound call blending, remote agent or multi-site support, and much more.

InControl, part of the inContact suite, is an application development tool that simplifies customizing solutions to a drag and drop level. The ability to quickly create a call script allows contact management models to be tailored to each client's individual needs.

InContact identifies and routes a repeat customer to an appropriate agent using Caller ID, where caller information pop-up screens appear simultaneously with the call for the agent, whether they are working at home or in the office.

http://www.targetinteract.com
http://www.ucn.net

Salesnet's On-Demand CRM Solution Takes Flight

By Erik Linask

During its six-year existence, JetBlue Airways has endeavored to be more than the typical airline — but for less. But as more and more airlines find it difficult to keep profits soaring, JetBlue also had to formulate a flight plan to keep its customers satisfied without cutting into margins or eliminating amenities.

It began by selecting Salesnet's (news - alert) patent pending on-demand CRM software to increase the effectiveness of its sales force. Salesnet's is a cost-effective alternative to expensive, complex, packaged, or premise-based CRM software.

Salesnet's Guided Performance Selling (GPS) strategy essentially turns software, configuration, integration, and administration into service offerings designed to drive increased ROI — higher than purchased in-house solutions, anyway.

The solution increases performance by defining best practices, guiding salespeople to use those best practices, and tracking ongoing success using those best practices. It also assists in defining the path between business goals and realistic solutions. On demand software is nothing new, but the ease of administration, adaptability, and use and the single, low monthly subscription cost make Salesnet's an intriguing proposition.

Ultimately, Salesnet was the most intuitive solution, including the sophisticated Salesnet Dashboard, which was able to display information significantly more clearly than other vendors' solutions.

http://www.salesnet.com

FrontRange Launches Service Management Software Upgrade

By Michelle Pasquerello

In an effort to support to its growing customer base, FrontRange Solutions (news - alert) introduced the latest installment to its IT Service Management (ITSM), the IT Service Management 5.0.4 Service Pack 1.

The software is designed to assist IT service managers and support staff with new features, such as Inventory Management enhancements, including CI Comparison Utility, Inventory Identity, Discovery Enhancements and Scheduled Jobs. ITSM 5.0.4 Service Pack 1 can be installed on top of ITSM 5.0.4.

"These innovations increase ITSM's enterprise class functionality," said Lori Samolyk, FrontRange Senior Product Marketing Manager in a statement. "The FrontRange ITSM suite is helping small and large enterprises manage their IT systems and processes in accordance with ITIL. The ground-breaking ITSM suite is a mirror image of the ITIL best practices, which are quickly becoming accepted as the business model for IT."

FrontRange's products are designed specifically for small-to-medium-enterprise (SME) and distributed enterprise organizations.

http://www.frontrange.com

Nice's Contact Center Shopping Spree: Performix and IEX

By David Sims

Nice Systems (news - alert) has announced two acquisitions. The company signed a definitive agreement to acquire IEX, a vendor of contact center workforce management products. The company also signed a second definitive agreement to acquire Performix, a vendor of contact center performance management.

IEX sells workforce management, strategic planning and performance management products for the contact center market. IEX's flagship product TotalView is billed as providing "a high-end centralized product that compiles data seamlessly across the enterprise, enabling more accurate and effective forecasting, planning and scheduling."

Perrformix sells contact center performance management, an emerging trend in the contact center market.

The combined product will, in the words of company officials, offer "the first truly holistic view of contact center operation, addressing contact center and enterprise stakeholders at all levels — agents, customers, supervisors, management, and decision makers in the enterprise."

It will also have contact center business performance and analytics products, including: interactions capture, quality monitoring, interaction analytics, coaching, forecasting, strategic planning and performance management.

http://www.nice.com

Aspect Software Breaks Down Last Barrier

Aspect Software (news - alert) announced it will provide and support the Digium open source IP PBX, the Asterisk Business Edition — a professional-grade version of the industry's first open source IP PBX — for customers of its Unified and Signature product lines.

The increased adoption of SIP and standards-based technology points to open source as an increasingly viable option. The early adopters of this technology have been drawn by the low cost, as well as the greater control and flexibility that open source telephony offers to companies.

The Asterisk Business Edition IP PBX provides tested reliability of critical functions and features and includes support and full documentation. Based on the Asterisk open source PBX, the product offers companies the same call handling capabilities expected of closed PBX systems, at a substantially reduced cost, including features such as switched or packet data and voice mail.

"Industry experts have acknowledged that the biggest obstacle to widespread adoption of open source applications has been installation and ongoing support," said Gary Barnett, chief technology officer and executive vice president of technical services at Aspect Software. "Now, there is nothing to stand in the way of companies being able to leverage the benefits that open source provides, including inexpensive voice transport."

http://www.aspect.com

Avtex Achieves Microsoft Gold Certified Partner Status

By Stefania Viscusi

Provider of contact center solutions, Avtex Inc., (news - alert) which provides contact centers with applications and systems integration solutions for enhancing customer contact, recently announced its Microsoft (news - alert) Gold Certified Partner status. Gold status is the highest level of the Microsoft Partner Program and is only given to those companies that meet standards and requirements set specifically by Microsoft.

As a part of the partnership, Avtex will have exclusive access to resources, software training and support as well as Microsoft's "stamp of approval" for its products and services.

"Today, Microsoft recognizes Avtex, Inc. as a new Microsoft Gold Certified Partner for demonstrating its expertise in providing customer satisfaction with Microsoft products and technology." said Allison Watson, Vice President of the Worldwide Partner Sales and Marketing Group at Microsoft Corporation.

http://www.avtex.com
http://www.microsoft.com

Jacada Leads the Way to Next-Gen Contact Center Productivity

By Erik Linask

Jacada Ltd., (news - alert) which develops contact center productivity solutions, has released version 3.0 of its unified desktop solution, Jacada WorkSpace, in a model created to meet the demands of the growing number of contact centers seeking a modernized agent desktop, while optimizing processes and maximizing CSR productivity.

Jacada WorkSpace 3.0, which became popular under it former brand, Jacada Fusion Agent Portal, now represents the next generation of contact center desktops. It is a single agent console that unifies customer interaction tools with a single access point to all the mission-critical applications that enable the agent to effectively service customers.

Enhancements in version 3.0 include universal agent capabilities, support for multiple, simultaneous call sessions, support for Linux servers, and Asynchronous JavaScript Technology and XML (AJAX) controls and features found in the new Web 2.0 rich client foundation.

This release is only the beginning of greater changes within the industry, according to the firm. "Market experts and industry visionaries continue to place significant importance on the adoption of a unified CSR desktop," said Jacada CEO Gideon Hollander. "But while desktop unification is a top priority in many contact centers, what this next generation desktop should actually be has, until now, remained undefined."

http://www.jacada.com

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Primus and TalkSwitch Establish Interoperability in the U.S.

By Johanne Torres

VoIP service provider Primus Telecommunications (news - alert) and voice systems designer TalkSwitch (news - alert) joined forces to announce full interoperability between Primus' calling services and TalkSwitch's 48-CVA PBX.

The ability to bundle Primus' VoIP calling service and TalkSwitch's 48-CVA PBX will offer customers VoIP features, follow-me functionality, and conferencing, while maintaining a connection to the traditional telephone network.

Joint deployments of Primus and the TalkSwitch PBX will provide call processing at the customer premises, while the user is free to select from a set of voice, data, and access options provided by Primus. Each TalkSwitch telephone system comes loaded with a host of features, including multi-level auto attendants, call cascade options, voicemail and will feature Primus' VoIP calling plans.

http://www.talkswitch.com
http://www.primustel.com

Nero Partners with Digital Rapids

Nero, (news - alert) a provider digital media technology, announced a partnership with Digital Rapids, (news - alert) a developer of professional hardware and software solutions for post production, broadcast, IPTV, Video on Demand (VOD) and other advanced media applications.

The first collaboration between the two companies is the integration of Nero's AVC/H.264 and High Efficiency AAC, which are core components of Nero's ISO-standard Nero Digital(TM) technologies. The Nero Digital technology family includes MPEG-4 and AAC, AVC/H.264, and High Efficiency AAC for the ultimate viewing experience from mobile phones to High Definition video screens.

Moving forward, Nero and Digital Rapids will continue working on a number of market-facing solutions, combining Nero's technologies with Digital Rapids' software and hardware solutions for professional video ingest, playout, encoding, transcoding, and delivery.

http://www.nero.com
http://www.digital-rapids.com

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VOICE • DATA • VIDEO

Sangoma has just announced its new FXO/FXS analog telephony solution that brings new levels of Telco grade voice quality, value and serviceability. For Asterisk™ applications they are your best choice in their class for reliability, price, support and ease of installation. Sangoma solutions include support for software-based PBX and IVR voice systems from traditional legacy protocols to the latest IP-based voice and data technologies. The Sangoma/Octasic partnership ensures the best Telco grade echo cancellation performance.

Talk to us. Make The Call to Sangoma at 1-800-388-2475 or visit us at Booth #522.

Sangoma's AA series analog cards have the following benefits:

» They use the same PCI interface, architecture and digital path as Sangoma's T1/E1 cards meaning no motherboard or compatibility issues and ultra-reliable interrupt handling.

» They have full line protection, making them legal to connect to the telephone network – this includes FCC Part 15, FCC Part 68 and CE certification with other certification to follow.

» Sangoma's AA architecture supports up to 24 analog interfaces both FXO and FXS, all operating through one FPGA and one PCI slot using one IRQ. This avoids the problems of multiple asynchronous DMA accesses and interrupts that would occur with multiple PCI cards.

Sangoma's "D" series cards with hardware digital signal processing also have a full range of features:

Powered by octasic semiconductor

» Octasic's internationally deployed carrier-grade echo cancellation solutions deliver unprecedented voice quality. With Octasic's advanced voice enhancement features, you can enjoy the highest standard of quality on all calls. *Because you know quality when you hear it.* Visit www.octasic.com

» Sangoma's Echo Cancellation hardware supports 1024 taps (128ms) of echo tail handling on each channel to take care of the most demanding echo problems

» Noise reduction and voice enhancement technology provides better-than-toll grade voice quality

» On board DTMF decoding and conferencing will be available as a software upgrade to reduce your system load even further

» Sangoma's patented EDAC technology saves you money by allowing you to purchase only the echo cancellation you require.

SANGOMA™ DELIVERING OPEN SOURCE MEDIA GATEWAYS WWW.SANGOMA.COM

Available from: voipsupply.com
1-800-398-8647 x30, Booth #852

Earthlink, Linksys Team to Provide VoIP Solution

By Erik Linask

Internet service provider (ISP) EarthLink (news - alert) and Linksys, (news - alert) a division of Cisco Systems, have announced a co-marketed Voice over Internet Protocol (VoIP) hardware and service solution that provides everything customers need to make phone calls over their Internet connection.

The new co-marketed VoIP solution features Earthlink's trueVoice telephony service along with the customer's choice of a phone adapter (SPA2002-ER) or wireless-G router (WRTP54G-ER) from Linksys.

EarthLink's trueVoice is compatible with any high speed connection and is a plug and play solution that can be installed in minutes to work on any touchtone phone. With the service, customers can achieve the cost savings and feature benefits of calling via the Internet, without having to invest in expensive new handsets.

The Linksys phone adapter works with a standard wired or wireless router, while the wireless-G has a built-in phone adapter for home networking. Both products provide two phone ports for connection of two standard home phones or fax machines.

http://www.earthlink.net
http://www.linksys.com

VoIP Logic to use Highdeals Transactive Pricing and Rating Solution

By Patrick Barnard

VoIP Logic (news - alert) has selected Highdeal's (news - alert) Transactive pricing and rating solution to improve the delivery and management of its next generation IP services.

Highdeal's Transactive modular software suite can price and rate thousands of transactions per second, giving carriers the functionality needed for fast and cost-effective delivery of today's emerging services.

"By integrating Highdeal's billing and rating engine into our on demand delivery platform, VoIP Logic enables service providers to deploy a carrier-grade billing solution quickly and with few in-house resources," said Kevin Burke, COO and CMO of VoIP Logic.

Highdeal's pricing and rating solutions solve the billing challenges brought about by the emergence of next generation services. By delivering unconstrained pricing and packaging flexibility, Highdeal enables the rapid implementation of convergent services with diverse payments models and multiple partners.

http://www.highdeal.com
http://www.voiplogic.com

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Vodavi and Converged Access Ink VoIP Reseller Deal

By Johanne Torres

Vodavi (news - alert) and Converged Access (news - alert) announced that they partnered to provide traffic management and quality of service (QoS) systems for VoIP networks. The new partnership calls for Vodavi to resell Converged Access' Converged Traffic Manager (CTM) and Converged Access Point (CAP) systems to Vodavi's authorized dealers.

Enterprise customers deploying the Converged Access and Vodavi's XTS-IP and TeleniumIP communications system bundle can now employ traffic management, and other WAN optimization and application control techniques. The Converged Access systems enable network managers to set priority levels for voice and data applications so that voice traffic is protected with Quality of Service (QoS) over the private network.

The Converged Traffic Manager enables toll-quality VoIP, jitter-free IP video, and business-IP response times. The product integrates traffic monitoring, traffic management, compression and caching with advanced application layer visibility and control to assure WAN efficiency and application performance for converging IP voice, data and video applications.

http://www.convergedaccess.com
http://www.vodavi.com

Interlink Global Taps XO to Deliver VoIP Offering

By Johanne Torres

Interlink Global (news - alert) tapped XO Communications (news - alert) to support the delivery of VoIP services to its US customer base.

The newly inked deal will enable Interlink Global to integrate the XO infrastructure and IP transport services to deliver VoIP services to business and residential customers in markets across the U.S., previously not served by InterLink Global. XO's wholesale services will also be used to support NetTalk, Interlink Global's recently launched point to point and multiparty video phone service for home and business customers.

"With XO in our camp we can continue to deliver the highest QOS as our client rolls increase due to the diversity and fast rollout of new products, such as NetTalk and Streaming Video," said Anastasios Kyriakides, chairman and CEO of Interlink in a statement.

http://www.xo.com
http://www.interlink-global.com

CompUSA Retails 8x8 Business VoIP Offering

By Johanne Torres

VoIP and video service provider 8x8 (news - alert) announced that CompUSA will now sell Packet8 Virtual Office in each of its 226 retail outlets nationwide

Packet8 Virtual Office is a VoIP-hosted PBX phone service that allows small and medium sized businesses (SMBs) to implement a business-class phone system with unlimited local and long distance calling in the U.S. and Canada for a flat-rate of $39.99 a month, as well as lower-priced metered plans and international calling.

The new agreement calls for CompUSA to offer a Packet8 Virtual Office starter kit comprised of three business-class telephones and terminal adapters, priced at $239.97. A single unit offering for expansion beyond the three-phone minimum is also available for $79.99. Virtual Office includes auto attendants, conference bridges, extension-to-extension dialing, business class voicemail and ring groups.

http://www.8x8.com

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VoIP SMB Market Ripe for Growth

By Greg Galitzine
Internet Telephony

Recent reports from the analyst community all speak to the potential for growth of the Hosted VoIP market, particularly when targeted at small to medium businesses (SMB). An AMI-Partners report, *The Emergence of Hosted Voice Services: Enabling Service Providers to Transition from Commoditized to Value-Added Services*, indicated that:

- Hosted voice offerings continue to make strong inroads into the SMB sector and the market, while still hesitant, shows strong interest and adoption plans.
- The SMB market for hosted voice services will grow at a 55% CAGR over the next three years.

The AMI study warns that hosted voice services are yet to become mainstream in the market, and some hesitation lingers as many businesses still desire an on-premise solution, but the news for hosted providers is not all bad. AMI-Partners believes that the following factors will all work in tandem to create a tremendous market opportunity for hosted VoIP:

- Cost advantages;
- Predictability of monthly service charges;
- Easy-to-use interfaces; and
- Feature-rich, productivity-enhancing services.

It is precisely these benefits that will drive many SMB decision makers to adopt hosted VoIP services.

According to Michael Lauricella, vice president of AMI's Telecommunications Practice, "Any firm focused on cutting costs and increasing productivity should seriously consider the benefits of a hosted solution. The fact is that hosted solutions present a reliable and viable solution to many of this nation's 6.3 million small and medium businesses as well as enterprise customers."
Research from Yankee Group supports this.

According to the December 2005 report, *Assessing the SMB VoIP Market*, hosted VoIP solutions are finding particular favor within SMBs, as 70 percent of those surveyed indicate they would prefer a hosted VoIP solution to a premises-based solution.

Yankee Group believes that SMBs would be attracted to hosted VoIP solutions, "because it enables them to focus on their core business rather than allocate resources to implement and manage the system."

And, the folks at Research & Markets believe that, "…the Centrex/Hosted IP market is primed for substantial growth over the next four years," citing growing demand that should "continue to perform at a year-over-year growth rate of 90% through 2009."

An article in *Business Solutions* concurs, pointing to even more analysis from organizations such as InfoTech and Analysis Consulting. The commentary, written by Jay McCall, offered the following stats:

- Small business VoIP adoption (companies with around 50 employees) is predicted to grow from 100,000 in 2004 to 800,000 VoIP subscriptions by 2008. (Analysis Consulting)
- By 2008, SMBs will spend $5 billion worldwide on VoIP, compared with just $41 million in 2003. (InfoTech)

What's Tempting SMBs to Adopt VoIP?

So why the interest in Hosted VoIP? What's driving SMB decision makers to consider and ultimately adopt this technology? The main drivers are financial (potential cost savings from cheaper minutes and the ability to reduce overall networking costs), and the creation of an efficient, effective workforce by giving them the tools they need to succeed. Hosted VoIP solutions offer enhanced features such as mobility and integrated/unified messaging options, ease of use, the ability to collaborate among multiple branch offices, lower operational expenditures as a result of simplified management schemes, and possible integration of the voice system with more traditional business applications.

Cost Savings

Price is still the number one factor driving companies to consider VoIP, but it's the promise of next-generation productivity applications and services that are the reason many people keep the service once deployed. With Hosted VoIP, the SMB pays a predictable monthly fee, obviating the steep up-front cost of purchasing a premise-based solution outright.

Tremendous cost savings can certainly come in the form of lower telephone bills. By converting analog voice into packets and transporting these packets over an IP network, corporations are able to avoid the PSTN and the tolls associated with that.

Perhaps one of the greatest benefits of a Hosted VoIP solution is that small business owners are free to focus on their business, not on the operation, maintenance, and management of their phone system, nor do they have to hire a specialist for that job.

Features
SMBs who subscribe to a Hosted VoIP solution to multiple locations can enjoy all the benefits of a distributed phone system, while maintaining a single voice mail store, extension dialing between remote offices, unified messaging, ad hoc conferencing, and more. Security features have increasingly become a major consideration in today's telecom decision.

Another major area of consideration concerns disaster recovery and business continuity. In the wake of such high-profile disasters as Hurricane Katrina, it has become evident that a distributed, fully redundant system is a must. For SMBs who have been displaced due to some calamitous event, a Hosted VoIP provider offers the ability to get back up and running in a matter of hours, versus days. And in an era where downtime costs businesses real money, it's an important consideration when choosing a provider. Displaced employees need only log in from a different location, and provided they have the suitable bandwidth, they'll be back in business in no time.

Yankee Group warned in their aforementioned report that only five percent of SMBs had adopted VoIP as their primary means of communications. Service providers have a lot of work ahead of them to inform and educate the market, but the time most certainly is now. Hosted VoIP providers are faced with a tremendous market opportunity, and they need to move quickly to capture their share. IT

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

For More Information on Covad's solutions
Contact John Grady, Director of Product Management, Covad Communications:
jgrady@covad.com

Covad: Serving the SMB via Hosted VoIP and Beyond

Covad (news - alert) has made significant strides over the past two years in its move to become a household name as a provider of integrated voice and data services. Among the key technological achievements in the company's evolution was the announcement of their Voice-Optimized Access (VOA) technology, which was designed to enable a fully-managed, business-class offering on T1 and SDSL to over six million small businesses in cities from coast to coast.

Oftentimes, consumer-class VoIP products require the customer to 'bring their own access' and thus do not control the quality of the broadband underlying their voice offering.

Covad's focus on the SMB (which includes such elements as service-level agreements or SLAs, as well as business-class features) allows them to differentiate themselves from those 'bring your own access' consumer VoIP providers. Covad's field technicians and dedicated customer care are able to provide the immediate support that an SMB requires, especially if they don't have their own IT staff.

Covad offers a whole portfolio of VoIP solutions powered by the innovative VOA technology, which was designed to manage the quality of both voice and data services simultaneously, allowing customers to make multiple phone calls and access large data files at the same time, without sacrificing voice quality.

Covad's executive vice president of sales and marketing, David McMorrow, said "Voice Optimized Access unlocks the value of our facilities-based network and enables Covad to manage the voice quality over both the 'last mile' and our entire nationwide network.

"This strategic development demonstrates Covad's continued commitment to creating solutions that provide small-to-medium businesses with competitively priced, innovative communications services."

Covad VoIP provides the following features and applications:

- Find Me/Follow Me - Instant Call Forwarding.
- Click To Call - Speed Dial with the Click of a Mouse.
- Visual Voicemail - View, prioritize, and control voice and fax messages.
- Fax Mail - Receive and forward faxes just like e-mail.

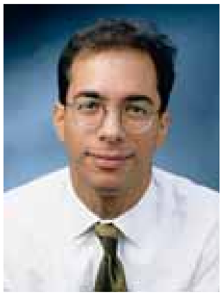
All of these features and benefits are easily accessible through a user interface known as a dashboard, which sits on the end user's desktop and makes it easy to increase efficiency and save administrative costs.

Covad has enjoyed success by offering their services over DSL, and in fact has evolved to become the nation's largest independent broadband provider. But as the market evolves and competitive threats arise from different quarters, it becomes necessary to innovate in order to maintain a competitive edge. To that end, Covad made a strategic decision to acquire NextWeb Wireless, a broadband wireless provider. The move enabled Covad to offer broadband voice and data services to businesses without the need to serve customers via DSL.

According to Charles Hoffman, Covad president and chief executive officer, "NextWeb expands our service area and offers us new, higher-speed broadband services than Covad can currently offer, and provides an alternative to the last-mile copper for delivering data and voice services to business customers.

"Wireless broadband is a perfect complement, not a replacement; to our nationwide DSL network," he added.

It should be noted that although Covad has made a big push with their hosted service, the company still offers PBXi, an alternative for the SMB that feels the need to exert some control and wants to retain their premises equipment. IT



By Marc Robins

Investors See the Light, as the IP Communications Industry Shines Bright...

The IP Communications industry is finally getting the respect it deserves from the investment and financial services community. What we in the industry have known for years — namely that the wholesale migration from TDM, ATM, and other legacy communications network infrastructures to one based on IP (coupled with the integration of the Web) will fundamentally, completely and forever change the communications industry landscape — is now finally becoming hot news among the "masters of the universe," including investment bankers, VCs, corporate fund managers, angel investors, and other prospective investors. It also doesn't hurt to have a few large deals open people's eyes...

Out of this disruption, of course, is emerging a whole new set of business opportunities and new companies entering the marketplace to take advantage of them. I've said in my recent writings that I fully expect the pace of new company launches, investment, mergers, and acquisitions to increase significantly in this year and beyond.

A number of factors are helping to create this attraction. According to TeleGeography, a global telecom industry research firm, the bandwidth glut is officially over. In a recent announcement by the company for its Global Bandwidth Research Service that provides a range of market analysis, forecasts, and essential statistics on long-haul bandwidth supply, demand, prices, costs, and competition around the world, "the global bandwidth market is showing signs of improved health: supply equilibrium, price stability, and competitor consolidation. Persistent international bandwidth demand growth has depleted inventories of unsold circuits on many submarine cables and on some segments of terrestrial networks. This has led many network operators to light additional wavelengths and fiber pairs on an as-needed basis. This incremental approach to managing spare circuit inventories means that lit bandwidth supply and bandwidth demand are coming into balance."

TeleGeography projects that, by year-end 2010, VoIP will have attracted over 21 million subscribers — nearly one in five of all U.S. households.

In addition, the extension of VoIP (define - news - alert) price wars to include the incumbent providers — most recently illustrated by Verizon's move to slash the monthly price of its VoiceWing VoIP service from $34.95 to $24.95 to compete more effectively with upstarts such as Vonage and far below the offerings of some of the cable MSOs — means that the incumbent telcos are starting to really feel the heat from rival VoIP service providers. According to TeleGeography's latest VoIP market survey, 5.4 million U.S. households now subscribe to a VoIP service — up from just 2.7 million one year ago. Even more troubling, 2.8 million of those households have defected to cable MSOs' VoIP services and have cancelled their local phone lines altogether.

By year-end 2005, Verizon (news - alert) had lost more than eight percent of its residential phone subscribers. According to the results from an April 2006 survey conducted by TeleGeography, the number of customers jumping to VoIP will only accelerate over the next year. TeleGeography projects that, by year-end 2010, VoIP will have attracted over 21 million subscribers — nearly one in five of all U.S. households. These numbers spell trouble for traditional phone companies. Subscriber migration to VoIP translates to $13.9 billion in lost long distance revenues over the course of the next five years, and $17.4 billion in lost local phone service revenues.

For more information about TeleGeography's U.S. VoIP Research Service and their ongoing analysis of the US consumer VoIP market, visit http://www.telegeography.com/products/us_voip/ IT

Marc is Chief Evangelism Officer of RCG (Robins Consulting Group), a leading marketing, communications and management-consulting firm dedicated to the IP Communications industry. For more information, call 718-548-7245 or email robinsconsult@optonline.net.

Announcing VoIP Demo and the IP Communications Business Summit

We're so confident that the market is primed for new, accelerated investment activity that TMC and RCG (my company) are putting our time and money where our mouths are. Recently, our two companies teamed up to create two unique events that will allow investors and prospective customers to research leading industry trends, and evaluate IP Communications products and services by viewing live demonstrations from a number of leading companies.

A first of its kind event, VoIP Demo allows attendees to research new VoIP and IP Communications products and services by taking in live demonstrations and interactive presentations. VoIP Demo will take place August 8-10, 2006 at the Santa Clara Hyatt Regency, located in Santa Clara, California.

Developed as an integral part of VoIP Demo, *The IP Communications Business Summit* is an associated conference and networking event aimed at bringing together vendors and service providers with members of the financial and investment community — including venture capitalists, investment bankers, financial industry analysts, M&A specialists, and angel investors. The IP Communications Business Summit will provide a unique opportunity for attendees to meet and network with top executives from IP communications companies seeking investment, and gain invaluable insight from leading independent industry researchers, analysts, and consultants.

Unlike any other events, VoIP Demo and the IP Communications Business Summit were created to provide value to companies seeking to implement VoIP and IP communications technologies in their organizations, to members of the financial industry interested in evaluating companies and products for investment, and to vendors looking for new customers and funding in this increasingly competitive industry.

Vendors interested in participating at VoIP Demo should contact Dave Rodriguez at (203) 852-6800, ext. 146; or e-mail: drodriguez@tmcnet.com. For more information, please visit http://www.voip-demo.com.

For more information about the IP Communications Business Summit, e-mail inquiries to Marc Robins at summitinfo@robinsconsult.com, or call 718-548-7245.

By Tony Rybczynski

Leveraging Converged Networks To Optimize IT Budgets

The global IT spend across enterprises, governments, and SMBs was estimated (from various sources) to be a whopping $1.4 trillion on a global basis in 2004. Only three percent of this amount was for networking equipment and yet, the strategic importance of networking goes well beyond this. Networking is central to CIO priorities of doing more with less across IT, enhancing employee productivity, and strengthening customer engagement.

The enterprise segment is diversified, ranging from single site businesses with a few employees to multinational corporations and national governments. So there's no such thing as a typical IT budget across these diverse environments, but there is value in looking at the distribution of global IT spend as a surrogate for an enterprise's IT budget, with the intent of better understanding how IT costs can be reduced.

Attacking The Monthly Bill

Roughly one-third of the global IT spend is paid to service providers. Two-thirds of this is for traditional services, such as inter-site and public phone calls, including 1-800 number and calling card, Internet access, and inter-site connectivity frame relay and leased lines. The other third and growing portion is for cell phone charges. There are three major implications.

Firstly, converging all forms of traffic onto a properly engineered and designed converged IP network can shift the task of finely engineering multiple networks to minimize the bandwidth costs to optimizing the price/performance of a single network. Even if you are paying a penny a minute for voice today, saving half a cent per minute can put money in your pocket if you have a significant amount of monthly call minutes.

Secondly, further efficiencies can be achieving by performing WAN optimization through the intelligent deployment of QoS mechanisms, application-aware dynamic routing, and compression at the link, network, or application levels. Saving 10 percent of your monthly bill by investing in WAN optimization in your edge device (e.g., a Secure Router) can drive a fairly straightforward business case through reduction in bandwidth costs or deferral of new bandwidth procurement. There is an added bonus, particularly if such WAN optimization results in increased WAN reliability and faster recovery from network failures.

A properly engineered and designed converged IP network can shift the task of finely engineering multiple networks to minimize the bandwidth costs to optimizing the price/performance of a single network.

Thirdly, cell phones have become indispensable to business, and will continue to grow in importance with the introduction of third-generation broadband mobility (e.g., EV-DO, CDMA wideband, WiMAX, metro WiFi). As a result, there is tremendous pressure on IT to gain control over cellular costs whether paid for directly by IT or personally expensed by employees. Given that it is estimated that over 50 percent of cell calls are made or received from inside buildings, the marriage of voice and enterprise WLANs and so-called dual mode devices have created an opportunity to substantially lower these cell costs. Dual mode mobile devices allow you to use the public network when on the road, and then to use the enterprise wireless LAN network when within buildings. This will result in better coverage within buildings than available from cell networks, seamless roaming, and an opportunity for more consistent user experience (e.g., single mailbox and enterprise feature operation) across public and private environments).

Server Centralization Over Converged Networks

Approximately twenty five percent of the global IT spend is spent on software and hardware, including PCs, servers, and storage. Not only does this expense drive traffic and rely on secure, reliable connectivity, but there are two significant opportunities to lower the investment in these elements through networking.

Firstly, with convergence, the enterprise has one network to build, secure, and make reliable in support of voice, emerging real-time converged communications and mission critical data (including storage) applications. Having a single higher speed network with QoS creates opportunities for centralization of data centers and storage facilities. Worried about application performance, business continuity and regulatory compliance? The answer lies in integrated server/optimization solutions that accelerate applications including e-mail and file sharing, provide in-flight encryption for enhanced regulatory compliance, and deliver business continuity via effective desktop backup and recovery capabilities. Centralization delivers economies of scale by requiring fewer

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servers, better management of software licensing and more effective operation.

Secondly, the distributed nature of IP Telephony also creates an opportunity to follow a similar model for voice, eliminating the traditional nodal deployment of voice systems. Contact centers can also be virtualized with servers located in data centers and agents distributed through IP telephony. Centralization of IP Telephony is consistent with the trends to mobility and enables disaster recovery/business continuity.

Even if you are paying a penny a minute for voice today, saving half a cent per minute can put money in your pocket if you have a significant amount of monthly call minutes.

Networking: A Utility for Users and a Strategic Asset for the Business

Budgets are flat, while overall complexity, traffic, applications and security threats keep growing. While the acquisition of networking technology may represent a small part of the overall IT budget, this investment can help reduce overall bandwidth and computing costs, while improving performance. These freed-up resources allow enterprises to address strategic imperatives that transform the business through real-time converged communications and stronger customer engagement. IT

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

By Hunter Newby

What's In A Name?

As the fittest survive during this technological evolution it's no wonder the applications that make it are ones that people need and find easy to use. This same common sense can be flipped into logic that is the basis for determining what will be successful. There are certain components to the bigger picture that can be found in everyday life now and there are others that will be created and developed into mainstream when the time is right.

The key to success lies in knowing the entire value chain, from dark fiber to transport to IP networks... all the way up to the application layer. Understanding where everything is provides the ability to know where a new product, or service will sit in the chain, how well it will do, and how it might impact the other pieces.

Take ENUM, for example. Number mapping makes sense for many reasons, not the least of which is on-net calling at zero cost per call. ENUM sits at layer 5 and enables the voice application to work more efficiently. Today it uses E.164 numbers, but, in reality, SRV records work just as well, or better, with e-mail addresses.

SRV stands for Service which makes it an abbreviation, I guess, and not an acronym. As ENUM (define - news - alert) is to number mapping, SRV is to name mapping. Names seem to be quite logical to map for the purposes of establishing sessions given their current and growing acceptance in the marketplace.

Consider URLs in this process. Do we type an IP address in our browser, or a domain name? Do we remember everyone's telephone number, or do we just select their name on our mobile phone or PDA?

This actually brings up two separate points about the future of VoIP: Network Intelligence and Human Intelligence. In both, names may ultimately be the top level routing identifier because of simplicity. With Network Intelligence it is really the layer 5 piece doing the thinking (searching) for you. It's not really the "network" that is smart, but rather the database logic in it and signaling from your switch, IPBX, or that of your carrier. The Human Intelligence is when you want to initiate a session of some type with another person. You "look-up" the person by name on your device, or PC. When selected, the device asks you how you would like to interact — MMS, SMS, e-mail, voice, video. Your intelligence chooses the type of application and the network will work off of your queue and search for the other person by name, most likely an e-mail address. The reason for that is simply that it currently exists and it works. Keeping "look-ups" as names eliminates a step and decreases signaling and session set-up times.

Names will ultimately dominate the domains and that last remnant of the PSTN, the telephone number, will be relegated to the Smithsonian.

There are some that believe that names will ultimately dominate the domains and that last remnant of the PSTN, the telephone number, will be relegated to the Smithsonian. There are others that know nothing of this concept and are trying to build businesses and revenue models on numbers through ENUM. Telling the future is tricky business, so look to the past to see what has worked. Are we all just a Social Security Number, or are we really known by our name?

This issue and many more were raised at the recent Voice Peering Forum held at the Wyndham in Miami Beach. The participants included several experts in the field that openly share and exchange information for the benefit of all in the industry. Without that interaction there wouldn't be much in the way of progress. Among those I would like to thank are Rich Tehrani, Gary Kim, Eric Dean and Randy Waters. Thank you all for your exceptional insight. I look forward to future events and the future of VoIPeering! IT

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


SURVIVING
THE
IP
JUNGLE
PAINFUL TRANSITION
UNEXPECTED COSTS
POOR QUALITY OF SERVICE
LACK OF FEATURES


Tadiran Telecom
CORAL IPX OFFICE

VoIP: Targeting the Small/Medium Market

By Greg Galitzine

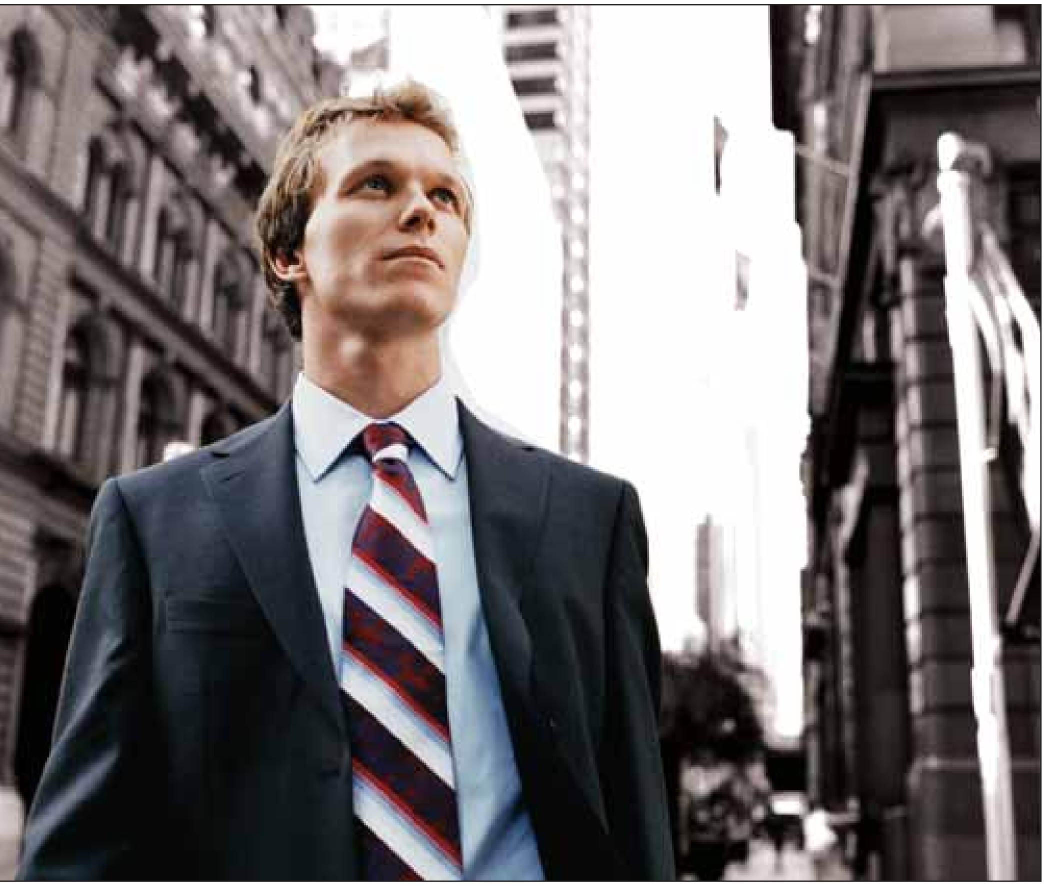
The VoIP (define - news - alert) market has seen numerous shifts over the past several years. From enterprise solutions to a strong service provider focus, and back to enterprise, the pendulum has swung from one extreme to the other. These days, arguably the hottest target market for VoIP services is the small to medium-sized business (SMB).

A dizzying array of solutions is increasingly available to the SMB market, and while choice is a wonderful thing, how are businesses to discern which solution is right for them? The average SMB does not have an IT department with an unlimited budget and the time to dedicate to testing and evaluating and navigating the vast number of choices of phone systems available to them.

Therefore, SMBs are looking for solutions that are neither complex nor time-consuming to install and operate.

Further muddying the landscape for SMB decision makers is the dreaded build versus buy question. Do I go with a hosted VoIP provider or do I purchase the equipment and deploy it in-house? There are a number of factors that go into this decision, but in the end, it's less a debate about technology and more a question of what SMBs feel more comfortable with and frankly how each type of solution addresses the specific needs of the business.

And let's not forget the issue of price. Total cost of ownership (TCO) and the return on investment (ROI) remain key considerations for the SMB owner.



For the sake of this article, let us assume however that the business has the resources to deploy and manage an IP telephony solution in-house.

First off, the SMB owner is looking for a phone system that will deliver quality voice at a reasonable price point. It's often been said however, that while price may be one of the leading factors driving SMBs to consider VoIP, it is the promise of enhanced features and functionality that really drives home the benefits of deploying this type of phone system.

These benefits can include bundled applications such as unified communications and integrated conferencing and collaboration capabilities, as well as applications such as Web-based system management, which are designed to simplify moves, adds, and changes.

Today, people use multiple devices to communicate (cell phones, softphones, Blackberries/PDAs, PCs, etc.) be it via voice or some other form of communications such as Instant messaging/chat, or even ad hoc video conferencing.

The long-sought promise of unified communications brings together these various modes of communicating, providing access to messages of all types (e-mail, voice, fax, IM, etc.) via a single message store, available to the user regardless of which device they prefer to use. Thus, one can check e-mails via the phone, listen to voice mails played back as .wav files via an e-mail client, send and receive faxes through a PC interface, and so on.

Unified communications also offers the key advantage of being able to tie together the aforementioned communications modes with various business processes using IP. Now you can combine enterprise applications with your communications infrastructure, thus enabling your employees to be more productive and more efficient as they go about their daily business regardless of location and what device they're using.

Another application is conferencing/collaboration. This application enables SMBs to save money on business travel and adds new levels of interaction and collaborative abilities for communicating with colleagues and clients alike, without the need to be there in person.

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Today's conferencing/collaboration tools enable workers in one location to work on projects with remote workers. This removes geographic barriers to hiring and housing employees. Collaboration software, tied to your communications system, enables ad hoc conferencing for real-time collaboration between far flung colleagues, affording them the same access to opinions and advice from a coworker as employees who share the same location take for granted. Lastly, these solutions increase the efficiency and effectiveness of communications across the business.

"SMB decision makers need to consider from whom they're purchasing their solutions."

When it comes to deployment, today's IP phone systems also need to integrate with the existing networking infrastructure. Pre-deployment analysis will dictate what level of network upgrade is necessary, and certainly the cost of bringing your LAN up to speed to handle voice is something to contend with. Other considerations include system redundancy and survivability, security, remote office connectivity, and mobility.

Perhaps most importantly, SMB decision makers need to consider from whom they're purchasing their solutions. For all the talk of products and features and benefits, the level of support and the level of comfort with the reseller bidding for the business is often the deal maker. When the issue is something as important as replacing a company's phone system, there is simply no substitute for a good reseller or systems integrator, with a good service and support record, who just happens to sell good products from a reputable vendor. After all, at the end of the day, communications is all about people. IT

Greg Galitzine is the editorial director of Internet Telephony magazine.

NEC: Serving Up Innovation for the SMB

By Gail Fisher

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

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

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

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

In response to customer demand and an increase in SMB interest in VoIP technologies, NEC's IPK II telephony solution was developed to provide the needed flexibility to tackle the most complex organizational needs.

The IPK II provides a unique set of advantages to those businesses seeking an advanced information system that is both flexible and dependable. With the ability to serve as a stand-alone IP telephony node, and scale from four-456 ports, IPK II allows for future growth as business requirements change. With the SMB in mind, NEC developed the IPK II to work with an organization's existing hardware and application infrastructure in order to provide remote users with the same features available to those in the home office location. Additionally, remote users may use an organization's communication solutions, such as softphones, Call Center applications, and other tools, to ensure that standard business procedures are not only completed, but are done so more efficiently. IPK II's feature-rich HTML-based function allows for easy adds, moves, and changes within the solution, meaning your staff can make adjustments to the system as needed without having to call a technician, which increases efficiencies and lowers the total cost of ownership.

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

Gail Fisher is product manager, SMB solutions, at NEC. (news - alert)

By Steve Timmerman

Doing Key System Behavior Better Than Key Systems

Many branch offices and small businesses are clinging to their trusty key systems, and no wonder: IP-based key system solutions are perceived as awkward, overpriced compromises that fail to exploit the power of the new technology or retain the full and familiar functionality of the old one. This is ironic, because a properly architected IP telephony system can deliver better key system behavior than traditional key systems themselves, plus a whole lot more.

Traditional key systems have been a mainstay of business voice platforms for decades, providing a cost-effective and reliable turnkey solution for small companies and branch offices. They are simple and straightforward, offering a focused subset of telephony features and functions without all the complexity of an enterprise voice switch loaded with features and functions that smaller locations and companies don't need or want.

At issue is whether key systems can accommodate the rising expectations of business users. Increasingly, even small outfits want voice systems that support unified messaging, desktop call control, contact center functionality, workgroup collaboration, hoteling, teleworking, and mobility.

And enterprises with multiple small locations don't want as many standalone phone systems to manage as they have sites. Even individually, key systems are inflexible and difficult to program, often requiring cryptic coding through the telephone key pad, or needing outsourced management altogether.

IP telephony offers the opportunity to eliminate these shortcomings, mimicking key system functionality while supporting advanced applications, provisioning a single feature set across all locations, and offering remote GUI-based management. In practice, many VoIP platforms have a centralized architecture that makes dial tone dependent on central servers and strips them of their standalone reliability. In some cases, dial tone is dependent on software and hardware such that, when the central servers or WAN links go down, the remote sites lose most or all of their voice capabilities unless very expensive 1:1 redundancy has been implemented throughout the infrastructure.

IP-based key system solutions can be limited by the lack of specialized key system phones. Users are expected to move from their familiar key system devices to IP phones originally designed for more general purposes — enterprise users. Some users may find this confusing.

Instead, consider the level of key system functionality that can be delivered by a fully distributed IP telephony architecture. While there is a single system image for the entire enterprise, full call control capabilities are embedded in the switches at each site. Remote offices can go on offering full key system functionality even after being cut off from the WAN or the corporate data center.

Done right, IP telephony can be anything you want it to be.

In short, you get IP telephony's ease of administration and management without sacrificing key system autonomy and reliability. You also get unlimited scalability, with no need to deploy and support different platforms in small, medium, and large offices. Imagine the possibilities when key system behavior can be enhanced by all manner of advanced applications and is supported by state-of-the-art IP key system phones, which retain the simplicity of traditional key system phones.

One of the main benefits is the ability to set up bridged call appearances, creating virtual line appearances for inbound calls. This feature enables very fast call handling among users in environments requiring shared call answering. It also works across multiple sites so the employees in the answering pool don't have to be at one location.

Specialized IP key system phones with lots of buttons can offer the best of both worlds, providing one-touch access to hard-wired and programmable features. These include traditional telephony functions such as transfer, hold, conference, intercom, redial, voice mail, and more advanced features such as directory access and group and whisper paging.

In fact, pretty much any feature that can be added to an IP telephony system can be accessed from a button on the IP key system phone. If more buttons are needed for more functions, just daisy-chain some IP button boxes to the phone. More than 100 buttons can be configured in a single key system solution this way. And all the buttons can be self-labeling, making inefficient paper labels a thing of the past.

IP telephony can enable new and better ways of doing business, but it doesn't have to force compromises on business users as they relinquish their key systems. Done right, IP telephony can be anything you want it to be, and deliver a full complement of key system features and functionality. When the IP telephony architecture is fully distributed, a single VoIP system can scale from 10 to 10,000 users across hundreds of locations and include a rich set of key system behavior wherever it is needed. IT

Steve Timmerman is a board member of the Enterprise Communications Association (news - alert) (http://www.encomm.org) as well as the vice president of ShoreTel, Inc. (news - alert) For more information, please visit the company online at http://www.shoretel.com.

The Voice Peering Fabric™

Enhancing VoIP with Voice Peering

The Voice Peering Fabric is a service of Stealth Communications®.

White Paper - *Published July 2005 V.2006050101*

Abstract

From the invention of the telegraph to the emergence of the Internet, the world has evolved and reinvented itself over and again. Technology has always created efficiencies and opportunities. Despite skepticism from some, history shows that every time there was a change for the better, investors and the public realized and followed. The drive for a better life has invariably and inevitably shifted workforces and profits across different industries. Investments were made and revenues collected directly or indirectly with the evolution. The latest trend in the technology world is voice peering, whether between carriers, enterprises or anyone joining to form this new community.

The Voice Peering Fabric ("VPF") was launched in October 2003 to accelerate transmission of digitized voice traffic. Built as a distributed Layer 2 Ethernet network, the VPF has solved many of the uncertainties engineers have had about both security and quality of digitized voice traffic. The VPF is a large and secure private network that enables carriers and enterprises to trade minutes as well as to distribute and acquire access to different applications that are necessary or useful for efficient communications among branch offices and with national and international business clients and partners. It is a global interconnection mechanism, a unified transport infrastructure, and a private grid for voice and telephony communication.

This white paper is brought to you by the following members of the Voice Peering Community™

Introduction

Voice peering is a method for the exchange of digitized voice traffic. A typical example is a company with two branch offices, linked via a data connection. When a telephone call is made from one office to another office on this connection, the call transverses the data connection without interacting with the Public Switched Telephone Network (”PSTN”). The phone systems at both offices are configured to categorize and route calls internally. These telephone calls do not incur any incremental per minute charges. The only cost is the cost of the data connection itself. This method is often referred to as “toll bypass.”

For dealing with the outside world, large organizations and service providers often have complex designs involving multiple locations and multiple interfaces into the PSTN. Phone switches and systems usually require intelligent-routing capability such as Least Cost Routing (“LCR”) and Electronic Number Mapping (“ENUM”). LCR and ENUM are actual methods of voice peering. LCR allows a particular call to be routed to a particular connection based on cost, time-of-day and other parameters. ENUM simplifies the routing of a telephone call based on the database number look-up architecture.

Controlling costs in addition to ensuring quality and security on a voice network are similar to the challenges faced in managing a data network. In this information age, business processes have accelerated in all applications, not least in the demand for efficient communications. Voice over Internet Protocol (”VoIP”) is an important factor and a revolutionary technology supporting an organization’s ability to constantly reinvent itself. VoIP requires careful planning and engineering to ensure current network infrastructures are properly utilized and potential savings realized from the new implementation.

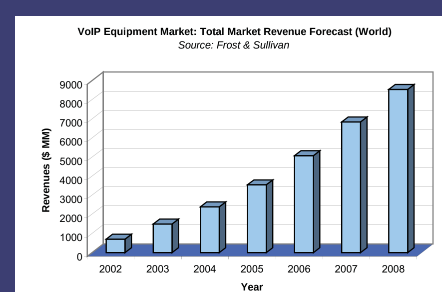
This paper discusses the value of voice peering through the use of the Voice Peering Fabric (”VPF”) coupled with VoIP implementations & deployment models for businesses with different backgrounds and needs. The paper will also provide a detailed introduction to the three major components of the VPF that enhance voice peering.

The Growth of VoIP

VoIP: It’s a method of transferring voice over an Internet Protocol (“IP”) network, which was originally referred as IP Telephony.

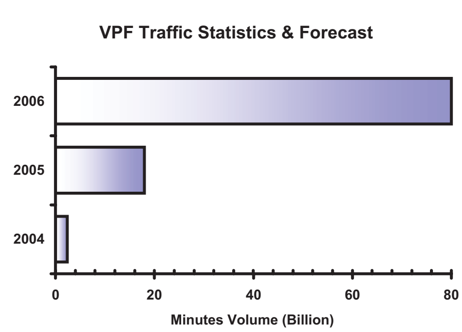
IP Telephony gained popularity when Yahoo, MSN, AOL and other messaging programs embedded voice-to-data conversion on their online chat programs. Since this implementation is over the public Internet on a desktop computer or handheld device, business entities are reluctant to employ these functions because of security and quality issues. Running VoIP over IP “VPN” (Virtual Private Network) or “MPLS” (Multiprotocol Label Switching) connections have somewhat resolved the security issue, but since some VPN sessions are over the public Internet, latency is difficult to control. Unless a physical dedicated connection is in place between branch offices, organizations with sensitive data such as those in the financial and healthcare industries may not deploy VoIP fully throughout their networks.

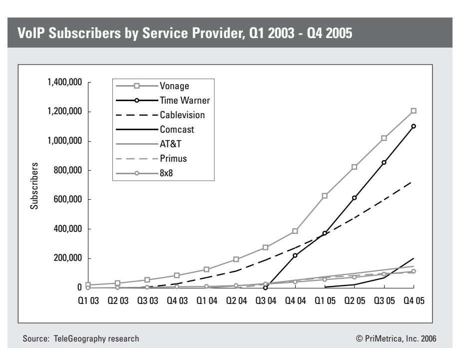
When Napster was launched, people were excited that digitization allowed them to download music for free. Similarly, VoIP (which does not have the copyright issues that have been so difficult in the music area) caught many people’s attention since it may significantly reduce expenses. The fact that VoIP is here to stay is proven by the growth of sales on VoIP technologies, as well as by the efforts of traditional telephone companies (like Verizon and SBC Communications) to offer their own VoIP products. The number of choices for VoIP phones, switches and gateways in the market seem to outnumber the choices for music gadgets. VoIP technology is important not only for cost reduction but also because it has enabled the world to communicate freely over PCs, mobile devices and IP phones and to break free of the limitations inherent in wireline, circuit communications that have changed little since the days of Alexander Graham Bell. With growing adoption of VoIP in Europe and Asia, more enterprises in North America have begun to deploy new IP PBX systems, IP gateways and IP phones within their businesses.

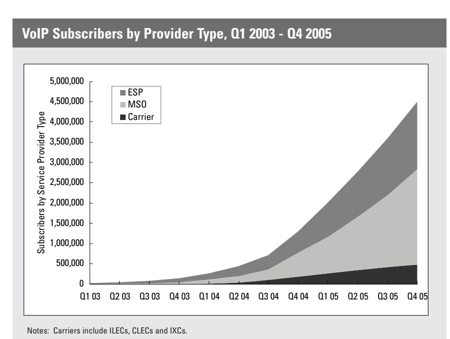


Stats on VoIP Growth

The growth of VoIP can also be seen through the exchange of minutes via the Voice Peering Fabric and acceptance of VoIP service in the consumer markets:







Defining Voice Peering

The Internet provides enterprises with improved communication, functionality and productivity with fewer resources. With the near total and universal adoption of the Internet, companies have taken steps to "localize traffic" to better control their internal communications and expenses. Given the means of keeping data traffic within a region, Internet service providers have long been "off-loading" Internet traffic at different "meet-points" around the world. In recent years, they have been able to establish more regional "meet-points" in ever smaller regions and to set-up "meet-points" for the private exchange of information among schools and universities. In the technology world, these "meet-points" are often referred to as "peering exchanges."

The same concept is being applied to voice communications. More importantly, separate direct connections are required for voice traffic due to the sensitivity of voice packets. Depending on the type of voice technology deployed by its members, today there are a few different methods of voice peering:

"TDM Peering": is typically a bilateral agreement between two carriers, to route telephone calls using "TDM" (Time-Division Multiplexing) technology, to and from the PSTN at a cost negotiated by the parties. This has been the traditional method and model in today's voice industry.

"Bilateral VoIP Peering": is the same as TDM Peering except instead of using TDM technology, calls are routed using VoIP technology. Major long distance carriers such as WorldCom, Sprint, and AT&T have been connecting using Bilateral VoIP Peering at different sites around the world to manage call volume and quality. In fact, billions of minutes are routed via VoIP for transport nationally and globally.

"Multilateral VoIP Peering": is a service within a peering exchange where all of its members agree to a set of rules for the exchange of VoIP traffic. It allows its members to send and receive telephone calls at no cost across the peering exchange.

In recent years, more VoIP networks have interconnected to exchange VoIP termination without tapping into the PSTN. Currently there are a few types of voice exchanges. Though these voice exchanges share a goal to localize traffic and bypass the PSTN -- they have their differences and limitations. Some voice exchanges connect their members by a voice switch and make their profit on commission fees; other voice exchanges are setup for the purpose of reselling voice terminations. Thus far, the VPF is the first to maintain a peering exchange that allows its members to trade minutes freely in an open marketplace while simplifying their business functions.

The Extended Value of Voice Peering

The role of the VPF is to build a community, of businesses and service providers for the exchange of telephony services. Rather than pursuing dedicated connections and spending resources searching for partners for particular types of routes of transport, a physical connection into VPF's distributed Ethernet fabric allows an organization to interact with hundreds of businesses and service providers located around the world.

The Voice Peering Community

Since our first user event held in late 2003, our community continues to grow at a rapid pace. New members and partners have brought with them more then products and services they offer. They have become part of the community where they share experience, knowledge and help businesses accelerate.

In our last two Voice Peering Forum held in New York and Miami, we had over three hundred industry professionals that met face-to-face, of which over a hundred of them joined us in both occasions. This business environment created an open marketplace that allowed buyers and sellers to come together – during which business deals were initiated and many were completed.

In addition to helping members of the community to connect with each other, we are on a continued mission to bring educational workshops to help our members on multiple aspects of their business.

VPF Members now include enterprises and service providers who are international PTT's, voice wholesalers, VoBB's to domestic ILEC's, CLEC's, and MSO's.

Partners to the VPF include Ethernet providers, carrier hotels, co-location operators, real estate owners, hardware & software companies and ASP providers.

VoIP is a new technology to many businesses. In the mist of technology convergence and net neutrality, we need direction and guidance. It is our privilege to work with the following publications, who have helped create awareness about Voice Peering and whose editorials have been comprehensive, informative and insightful.

VPF Minutes Market

Similar to stocks traded on financial market exchanges (NYSE, NASDAQ, etc.), voice traffic exchanged on a voice exchange is measured by minutes passed to each party, thus referred to as "minutes trading."

Traditional Minutes Trading, also known as Clearinghouses: The number and type of participants, the locations of the clearinghouses, rates and commission fees limit the growth and scale of minutes trading. Running long-haul circuits to a mutually agreed meet-point by the parties can be expensive and un-scalable. Connections via IP VPN over the public Internet are unsecured and low quality.

Minutes trading on the VPF is simple and secure. VPF PoP's (points of presence) are established at major fiber-dense connection points to meet businesses and service providers locally in their markets. Trading within the VPF Minutes Market provides its members:

- Direct access to multiple carriers;
- The ability to buy and sell origination (DID) and termination (DOD) services;
- The freedom to negotiate direct bilateral relationships;
- The choice of industry standard VoIP protocols and codec's;
- The opportunity to customize their LCR;
- The option of eliminating dedicated connections to each trading partner; and
- Access to VPF Minutes Market Request for Proposal ("RFP") Engine.

Featured VPF Members:

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DVVCom is a facilities based, VoIP Provider (located at 60 Hudson St. NY, NY) specializing in VoIP Origination / Termination (Domestic & International). We are also a Master Broker / VAR for multiple companies including: Bell South, ATT, Tier 1 / Tier 2 carriers, other RBOCs and many award winning equipment vendors. Our nation-wide portfolio includes a full range of telecom products from analog lines to T-1, DS-3, OC, GigE connections, equipment, etc.

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Telecom New Zealand International is a provider of premium quality wholesale voice and data services, offering fixed line, mobile and cable MSO customers a range of services from quality voice termination, mobile roaming and wireless content distribution, signaling services to subsea capacity, delivered through a global softswitch-enabled IP network. The VPF is key to our 2006/2007 business plan, and will serve as both the interconnection medium to access new partners, as well as the vehicle upon which we deploy managed voice-related application services.

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XO Communications is a leading provider of national and local telecommunications services to businesses, large enterprises and telecommunications companies. XO offers a complete portfolio of services, including local and long distance voice, dedicated Internet access, private networking, data transport, and Web hosting services, as well as bundled voice and Internet solutions. XO provides these services over an advanced, national facilities-based IP network and serves more than 70 metropolitan markets across the U.S. For more information visit www.xo.com.

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CEO
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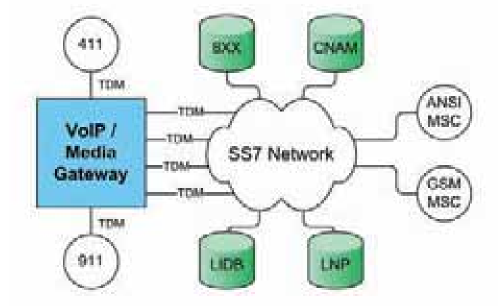
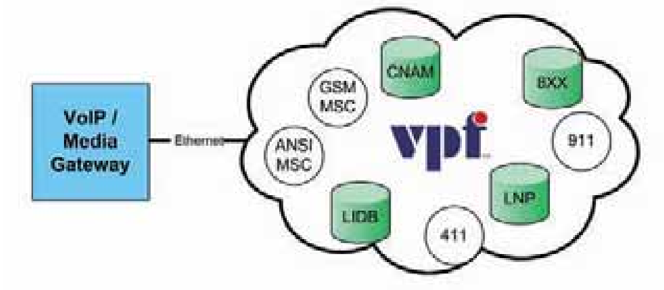
Waveleap Communications, LLC, founded in 2002, provides high quality VoIP origination and termination services to domestic and foreign carriers and enterprise customers. Our strategic partnerships combine the advantages of our IP network with the networks of multiple top IP and PSTN carriers to assure our customers of the best possible connectivity and quality. We also understand the difficulties facing new VoIP carriers and are glad to provide these companies with DIDs and other high quality services at reasonable rates.

Transition of SS7 to IP

Traditional method of accessing SS7 and other ASP applications through dedicated SS7/TDM circuits:



New method of accessing SS7 and other ASP applications via an Ethernet connection to the VPF:



Featured VPF ASP Partners:

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Telecom Services Division
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TNS Transaction Network Services®

TNS is a provider of network services designed to meet the needs of carriers operating current generation IP networks or traditional circuit-switched networks. TNS offers SS7 services including ISUP, CNAM, 800 and LNP, VoIP/PSTN signaling mediation, VoIP peering and route discovery services, managed network services, and custom applications.

Terri Dory
Marketing, Next Generation Services
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VeriSign®

VeriSign, Inc. operates intelligent infrastructure services that enable and protect billions of interactions every day across the world's voice and data networks. VeriSign runs the world's largest private SS7 network, and offers a full spectrum of solutions for intelligent communications, commerce and content, such as connectivity and interoperability services, intelligent database services, ENUM directory services, and content and applications services. VeriSign is your single source - providing secure, fully-managed solutions - from nationwide number acquisition and activation to VoIP peering to fixed-mobile integration. Ask about services available via the VPF at VPF@verisign.com, or visit our website at www.VeriSign.com.

VPF ASP Market

During consultation with our members to simplify their VoIP businesses, we found that much of their time and effort was devoted to acquiring and accessing third-party applications to support call routing/setup functions. A handful of connections were installed to access different databases; some members have dedicated people or departments to ensure the accuracy of telephone data on each telephone number. After completing surveys with members and partners, it became apparent that Application Service Providers ("ASP") services are a vital component to the VoIP industry. With the FCC's ruling on E-911 in May 2005, the development for the VPF ASP Market became imperative to better service our members.

An ASP is an entity that operates applications, databases and gateway services that are telephony related. Databases and gateway examples include:

- Toll Free Gateway ("8XX") – Enables providers to route toll free numbers (800, 888, etc.)
- Directory Assistance ("411") Service
- 911 Gateway – Maps telephone number to physical address and routes telephone call to the nearest 911 / E911 center.
- Caller Name ("CNAM") – Displays first and last name of a calling party.
- Local Number Portability ("LNP") – Enables providers to move and route telephone numbers.
- System Signaling 7 ("SS7") – Ability for service providers to access the SS7 network over a VPF Ethernet connection using SIGTRAN protocol.

While many of the services offered in the VPF ASP Market are geared toward service providers, businesses themselves can also utilize services such as 411, 911 and CNAM services. Benefits of the VPF ASP Market include:

- *An open framework and marketplace for buyers and sellers;*
- *Direct bilateral relationship with the VPF ASP Partners;*
- *Direct and easy access to the ASP services over an existing VPF connection.*

The VPF has simplified and made available a marketplace for ASP's and their customers. Their integration removes a layer of complexity and expense that too often is little more than interfacing with legacy networks and technologies.

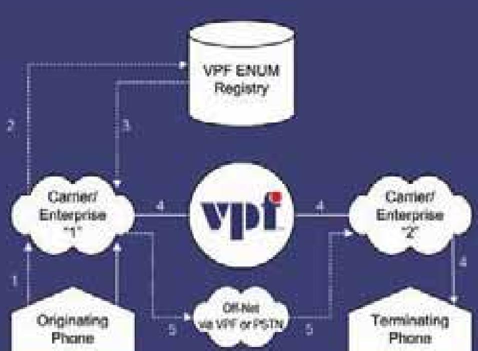
VPF ENUM Registry

ENUM: It is a network protocol that takes a telephone number and resolves it to an Internet address [URL], as a traditional Domain Name Server ("DNS") takes a URL (like www.google.com) and converts it into a numeric IP address. With ENUM, a telephone number is sent to the DNS server, which then replies back with the appropriate URL, if the URL/telephone number has been registered. This allows VoIP networks to send and receive telephone calls within the IP domain.

The VPF ENUM Registry is a multilateral peering enabling service that allows members to send and receive telephone calls to one another directly, free of charge across the VPF. It is toll bypass at its best.

Call volumes on the VPF ENUM Registry has been increasing. Launched in April 2004, the registry houses over 11 million unique telephone numbers, with no charge for the registration, lookup and calls. Current members of the VPF ENUM Registry include communities of universities, telecom companies and financial institutions. As the number of telephone numbers increases on this registry, more calls will be routed within the private networks.

VPF ENUM Registry Call Flow Diagram



The diagram above illustrates a call flow when using the VPF ENUM Registry.

1. User initiates phone call
2. Query sent to ENUM Registry
3. Routing information returned
4. If true, call established between the organizations through the VPF
5. If false, call sent to user's selected VoIP Carrier via the VPF Minutes Market or PSTN

Featured Technology Manufacturers:

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Cantata Technology's IMG 1010(tm) Integrated Media and Signaling Gateway offers service providers and enterprises unparalleled performance, reliability and flexibility to introduce services across fixed and mobile networks worldwide. The IMG 1010 supports wireline and wireless codecs which make it the ideal platform for transcoding in the next generation network. With its compact 1U package, integrated SS7 and flexible architecture, the IMG 1010 is a true carrier-grade VoIP gateway and/or VoIP transcoder that enables service providers to reduce costs while improving service quality.

Carrius Sales Department
T: +1-866-CARRIUS
E: sales@carriustech.com
www.carriustech.com

The Carrius *Compleat*™-200 Service Delivery Gateway (SDG) combines media gateway, and softswitch call control functions with powerful application control and signaling protocol translation, in a scalable platform. It supports interconnection of a full set of signaling protocols including SIP, H.323, ISDN-PRI, SS7, WIN, CAMEL, and CAS, and uses CCXML/VoiceXML, SIP, or the Carrius API to present applications with an abstracted network view. Through the SDG, service providers and solution developers can deliver services over a large collection of disparate networks.

Force10 Sales Department
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E: sales.americas@force10networks.com
www.force10networks.com

Force10 Networks is the pioneer in high performance switching and routing. Based on a revolutionary system architecture that delivers best-in-class resiliency and massive scalability, Force10's TeraScale E-Series switch/routers deliver predictable application performance and latency, increase network availability, and reduce operating costs in VoIP networks. For additional information, please visit the company's website at www.force10networks.com.

NexTone Sales Department
T: +1-240-912-1300
E: sales@nextone.com
www.nextone.com

NexTone develops carrier-grade products for delivering scalable control of real-time IP services, such as Voice over IP (VoIP) and other digital media. NexTone's products and technology give IP networks a common way to exchange, monitor, secure, and bill for sessions flowing through them. Over 400 service providers and enterprises worldwide use NexTone's solutions to dramatically lower capital expenditures and deliver ongoing operational efficiencies such as reduced interconnect "turn-up" time and simplified network operations.

Andy Voss
President & CEO
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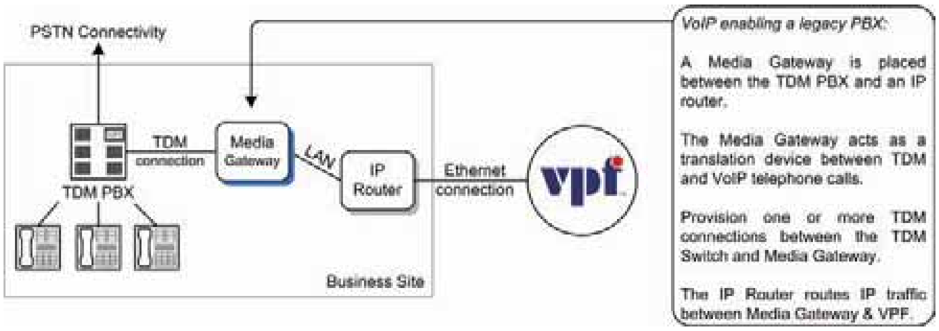
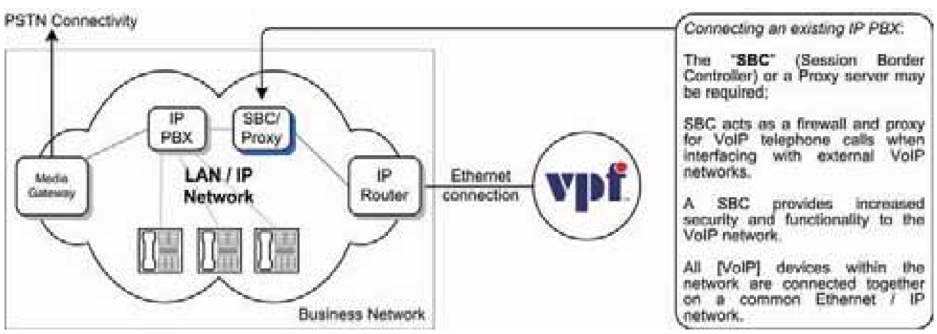
E: avoss@sansay.com

Sansay is the fastest-growing session controller company in the business. Sansay systems serve as a new generation of core infrastructure in high-demand VoIP call routing/switching applications. The Sansay VSX VoIP Session Controller brings VoIP carriers a new level of flexibility and control over their VoIP network through Sansay's advanced routing capability. Sansay's Least-Cost-Routing, ENUM and SIP/H323/MGCP interworking features can help reduce operating costs and increase profitability for organizations.

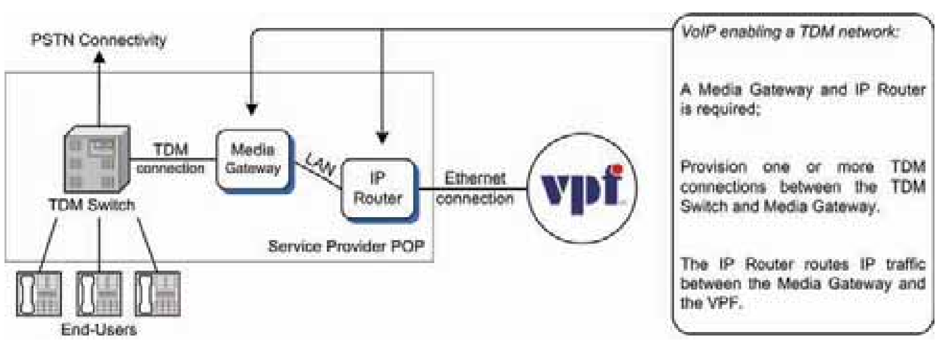
Large Businesses: Legacy PBX



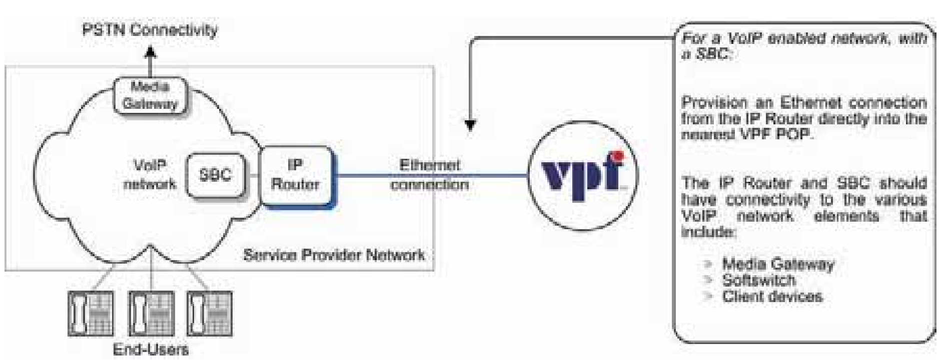
Large Businesses: IP PBX



Service Providers: TDM only



Service Providers: VoIP enabled



VPF Member Implementation Models

Since the VPF telephony marketplace introduces a new method for the exchange of minutes, members connect to us with minds open to the idea of trading minutes on terms that they negotiate easily and freely and the idea that they can now access applications on the same physical connection. The VPF simplifies both business plans and network configurations. To help our members and partners benefit, and to help others, the diagrams on the left are implementation models that map out a simple setup for each type of businesses connected to the VPF.

Did you know?

Utilizing a Media Gateway can protect existing infrastructure investments and enable you to exchange VoIP traffic immediately.

For most Enterprises, this solution allows them to realize savings between 50 to 90% and achieve an ROI within 3 months.

Service providers can now break free of the traditional footprints; this approach facilitates the reach into new market segments while controlling capital cost.

For more design tips and case studies, visit: www.thevpf.com

VPF Access Locations

The network reach of the VPF continues to expand. We are now accessible from most major carrier-hotels. In addition, the VPF Carrier Alliance extends the VPF to hundreds of buildings in various markets throughout North America and Europe. For a complete list of our ever-growing locations and partners, visit: www.thevpf.com.

- **New York City**
  60 Hudson Street - 9th Floor - tel^{X}
- **Atlanta**
  56 Marietta Street - 2nd Floor - tel^{X}
- **Boston**
  1 Summer Street - RCN
- **Chicago**
  600 South Federal Street - RCN
- **Miami**
  NAP of the Americas
- **Dallas**
  2323 Bryan Street - MMR - Digital Realty Trust
- **Los Angeles**
  1 Wilshire - 19th Floor - CRG West
- **San Jose**
  55 South Market Street - CRG West
- **St. Louis**
  Bandwidth Exchange Buildings
- **London, UK**
  Coriander Avenue - 1st Floor - Telehouse Europe

Featured Carrier-Hotel Operators:

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An operating partner of The Carlyle Group, CRG West provides the telecommunications industry with a robust offering of colocation cabinets, cages, and fully conditioned private telecom suites in the world's richest Meet-Me Room environments. CRG West owns, operates, and provides all interconnection at the One Wilshire Building in Los Angeles, Market Post Tower in San Jose, and 1275 K Street in Washington, DC. The CRG West Any2 Packet Exchange provides Internet peering as a utility for One Wilshire and Market Post Tower tenants, and is also home to the Voice Peering Fabric.

John Bonczek
VP of Sales
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tel^{X} is a premier operator of telecom "meet-me" network interconnection facilities. More than 250 networks physically converge within tel^{X}'s NYC facility, and more than 90 networks physically converge within its Atlanta facility. Known as a "marketplace" for network services, tel^{X} actively facilitates business opportunities between its carrier and enterprise customers. tel^{X} customers report higher revenue, greater profit margins, and lower costs from their tel^{X} based network operations. Internationally recognized for its facilities and services, tel^{X} continues to enjoy industry growth and success.

Summary

Literally, "One Connection to the World!"

The Internet has revolutionized business practices and enhanced the value of our lives. It is a tool that enables us to explore our interests and expand our dreams and imagination. The VPF is an evolution of the Internet's ability, via VoIP, to make voice communications faster, cheaper, and more comprehensive. Unlike the Internet, an organic community with no controls, the VPF has its implementation focused on security and quality.

The VPF has become the world's largest telephony marketplace. Working closely with its alliance partners, the VPF has expanded, and will continue to expand into businesses of different industries and markets. The growth of its VPF Minutes Market and VPF ENUM Registry over the last year has changed many businesses and benefited millions of users. With the recent introduction of the VPF ASP Market, the VPF provides an open architecture allowing third party applications to be developed and incorporated within the fabric - for instant accessibility to a global audience.

For information about the Voice Peering Fabric, visit the VPF web site at http://www.thevpf.com.

About the Author

Jinci Liu is co-founder and Managing Director of Stealth Communications. She is responsible for corporate strategy and business operations. Her functions at the company also include marketing and product development. She has a BS in Computer Science from Pace University and has been in the telecommunications and computer industry since 1995.

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STEALTH® COMMUNICATIONS

thevpf.com

The **Voice Peering Community**™ is personal.

We believe in putting faces to each business transaction. Whether you are looking to buy or sell telephony services, establishing direct business relationships can help increase margins and revenues. Our community is here to help you grow and maintain a personal network of contacts.

Peering @ the Personal Level

Through our **Voice Peering Forum**™ and **Voice Peering Workshop**™, we bring buyers and sellers together in an open market environment where no broker or counter-party exists in a transaction.

Members of our community include industry professionals who are C-level financial and technology executives. The people you meet here can recommend you to others and help you to grow your business. Personal referrals are still the best form of advertising.

Our goal is to combine a robust schedule of programs and events throughout the year to help members to develop strategic relationships within the business community, acquire new information and expand marketing opportunities. We encourage you to join our community to participate and take advantage of the opportunities that await.

Establish

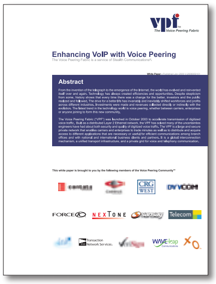
direct voice peering relationships to enhance your least-cost-routing system and reduce termination cost.

Access

SS7 and TCAP services via the VPF from leading North American SS7 providers to eliminate the cost and complexity associated with SS7 TDM connections.

Utilize

the VPF ENUM Registry, enable your networks to send and receive calls toll-free.


Enhancing VoIP with Voice Peering

Abstract

Download our free white paper
“Enhancing VoIP with Voice Peering”
from www.thevpf.com

The Voice Peering Fabric ™

SIP-ify the Internet E-mail Base

We are building borders that we will have to destroy.

By Jon R. Doyle

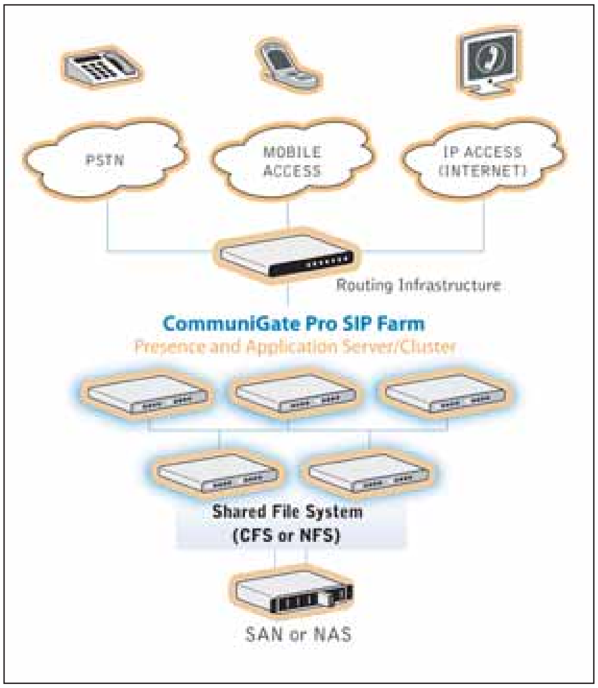
Widespread adoption of open standards-based VoIP is expected to take years as most companies begin to deploy new solutions for communications. There continue to be islands of users in proprietary systems, and we are creating more all the time. Think back ten years ago to the explosive growth of e-mail. There were many closed systems, but the advent of standards broke them wide open. Similar to the phone system, e-mail today is based on a set of standards and when you say "drop me an e-mail," you do not need to tell somebody to use a specific service.

What would happen if you handed somebody your business card, which lists your Skype address or your Google address? Essentially, you are forcing your colleague to use a specific technology in order to communicate with you or your business. This has serious and fatal flaws, and as was the case in the early days of e-mail, these models cannot be tolerated in today's business environment.

On my business card, jdoyle@communigate.com is my IP Communications address. Since it is based on the open SIP standard, if you were to type it into Windows Messenger, you would be able to IM me, see my presence, click to call me, and even start up a video conference. Inevitably, this is the natural evolution of true IP Communications: multiple media types, one account, and true portability of "address."

If you believe the promise of IP is to break the old business models, what then are we to make of new companies using new technology to mimic outdated practices? When I think about the models that are cropping up today, I chuckle. Let's look at VoIP providers and peer to peer schemas. Both Skype and Vonage charge you a fee or toll to access the telcos' closed networks; SkypeOut is a prime example of this. Calls within their closed networks, however, are free. So, what happens if these companies are successful in ridding the world of the PSTN someday? What will they charge then, and what will they charge to go from Skype to Vonage then? What changes from what we have today? Why do we continue to build more islands of closed networks?

Let's look at e-mail. It is open, it works through DNS (Domain Name Services) just like Web servers, and there are more than a billion e-mail accounts around the world. What would happen if we just "SIP-ified" all these accounts, meaning that every e-mail address was now also capable of receiving an invite for an IM session, or a direct, end-to-end phone call? Now you have a network on which roughly one quarter of the world's population can communicate without paying tolls. Think about that. The technology is in place now. The DNS servers and the SIP protocol all are designed to enable just such a scenario.



The SIP Advantage

E-mail users today have a common format and an open shared standard with which to communicate: name@domain.com. With SIP communications, this same address is used for all forms of communication, including instant messaging, voice, video, and, of course, e-mail. The address can have "aliases," associating phone numbers, extensions, and dial plans directly with addresses, allowing inbound and outbound calling with standard numeric PSTN and IP phones as well as alphanumeric smart phones and softclients. As an example, you can define a numerical number to jdoyle@communigate.com as you wish and not be defined or confined to what the telco tells you is the address. My 415 area code thus becomes meaningless. But since all forms of communication share this one address, a user can pass out a business card with one common address for all contact methods.

Just like e-mail, VoIP should be open, and SIP is the standard to do this.

So what is missing and why is this not happening as fast as we would like? There are several reasons. For starters, VoIP solutions are often based on closed standards and companies that do deploy open SIP-based solutions do not register their domain with DNS SIP and XMPP SRV records and ENUM entries that are exposed directly for end-to-end Internet calling. Also, many of the technologies on the market today do not carry an Internet pedigree, but are solutions from legacy telephony companies embracing IP, so they do not even have true hooks or solutions for e-mail — an essential ingredient in the tiramisu of unified communications: Multimedia In The Inbox (MITI).

Another issue just like in the early days of e-mail is that a number of VoIP solutions and presence-based technologies do not have the ability to scale because of flawed architectures. Keep in mind, presence alone, for 100 million people, with 10 buddies each on their buddy list, constitutes a massive load. Now imagine each user has a phone, a mobile device, an IM client, and a softphone, all running simultaneously. These numbers are very real indeed when you consider Vodafone's network or SBC's. What will it take to turn all those subscribers on to SIP?

CommuniGate Pro

CommuniGate Pro (CGP) is one of the key technologies that can make the transition to open standards-based IP Communications feasible. With nearly 10 years of product history and the world record for e-mail scalability and

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performance (as measured by the SPECmail standardized benchmark for e-mail), CommuniGate Pro recently successfully completed a telco-scale POC (Proof of Concept) for VoIP on an HP Superdome Integrity server, to demonstrate massive signaling for global carriers. And yet, the product also equally scales down from 10-million-subscriber, multi-node clusters to single servers or small clusters for SMBs using exactly the same application, putting CommuniGate Pro in the class of "truly unified communications servers" with only one member, and without any direct competition.

CommuniGate Pro has proven what others find extremely difficult: scaling down a carrier product to run on a laptop or home entertainment box. This opens up some interesting possibilities. Imagine placing CGP in any home or business and creating a SIP backbone with nodes all around the world, acting along with other standards-based SIP servers to create a global, Internet-based VoIP network which completely bypasses the PSTN, or at least only transfers to the higher-cost PSTN when absolutely necessary. The product's architecture also allows it to create a SIP Computing Grid, called SIP Farm. Simply put, CGP can run on a laptop or home server to provide all IP Communications, home PBX/answering machine functions, and even provide multi-party conferencing for use with family and friends, while simultaneously serving SIP-based VoIP subscriber bases of 100 million accounts within telecommunications data centers.

A carrier using CommuniGate Pro SIP Farm is fully redundant and can expand their service capacity on-the-fly by adding nodes to the cluster, while the entire cluster continues to act as a single organism. Regional SIP Farms can be used to keep most calls local to busy calling areas, such as metro regions. If one of those areas experiences an event that knocks a node or cluster offline, traffic can be re-routed to other SIP Farm PAKs in other regions, which are all part of one large cluster with consolidated identity management (all domains and all accounts within one administration menu). This allows any changes, like upgrades or equipment failures, to occur without impact on users. We are accustomed to Internet outages, and scheduled service downtimes for software upgrades and the like; but you never hear your phone company tell you they will be down Saturday for upgrades. That is the elegance of CGP's SIP Farm: One global grid, self healing, and expandable.

Trade In & Trade Up

Due to commoditization of e-mail, large scale providers today do not make a large revenue stream for that service. Therefore, these large e-mail systems are a huge operational expense. Moving in SIP-based IP Communications or replacing these systems with modern platforms is simply cost prohibitive. Add to this burden the ongoing operational expense of managing and maintaining years-old technology that is often overly complex and possessed of a flawed or outdated system design.

CommuniGate Systems is planning to offer these providers a move to IP Communications based on the maintenance of the software and waiving the cost associated with licensing for systems built on legacy Critical Path, Openwave, and Sun's old iPlanet server. With collectively more than 40 percent of the world's users using these legacy systems in tier1 and tier2 carriers around the globe, the first step is making SIP a widespread and open-use network of subscribers. Getting the technology in the hands of the providers, consumers, and enterprises at little to no cost will quickly make the IP Communications network useful.

Conclusion

We will witness a fundamental change in the communications landscape over the next five years, just as we saw in the early 90's with e-mail becoming the standard communication medium for business. Holding users to a location with their phone numbers and then charging them for roaming to other locations is preposterous and will soon be replaced with the mobility and portability of VoIP — where one address can find and follow me no matter where I am, regardless of the network, and irrespective of the access device. IT

Jon Doyle is Vice President of business development at CommuniGate Systems.

What's Happening at CommuniGate?

By Greg Galitzine

In a bold move, CommuniGate is making available a free edition of CommuniGate Pro that will serve up to five users. The CGP Community Edition is designed for small companies and home users. The product will offer a full e-mail server, SIP & Presence Server, IM Server, voice mail, PBX, and conferencing server.

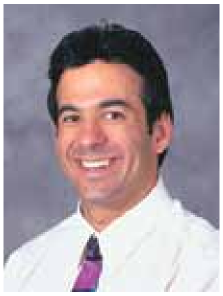
Any person can install this on their home computer, with a domain of their choosing, and become SIP enabled with access to their IP Communications anywhere in the world. That means a small company, or home user can flip open their laptop and connect to a WiFi network at the airport, read e-mail, IM, and receive phone calls, all with their one SIP-based e-mail address.

Enabling true mobility and number portability, CommuniGate's Community Edition allows a user to call a colleague in Paris and ring their laptop soft client in San Francisco.

Small business users will have a wide choice of clients (SIP phones, soft clients, IM clients, browsers, etc.) and they will be able to send and receive all IP Communications via a single account, which is identified by their e-mail address. Communications will be open to every other SIP-based application, and will remain vendor agnostic. The Community Edition will ship with a Flash-based user interface that can do e-mail, IM, and audio calls, a softphone, and will offer out-of-the-package compatibility with many SIP phones like Polycom, Linksys, and others.

CommuniGate Pro will be available to run on home computers, laptops, and eventually could be adopted into the home entertainment environment on devices such as cable and DSL modems as these devices begin to empower the home or family domain for all IP Communications.

Greg Galitzine is the editorial director of Internet Telephony.



By Rich Tehrani & Max Schroeder

Continuity Planning 101

"Déjà vu All Over Again"

In this month's column, Rich and Max interview some members of the Disaster Planning Communications Forum (DPCF) to better understand the role managed services plays in the business continuity market. They spoke with the following:

- Doug Brown, Principal, Miami Data Vault
- Lawrence Bothwell, Sales Analyst/IT, Standley Systems, Inc. and
- Rick Davis, Chief Sales Officer, Skyport International, Inc.

Historical Perspective

In the 1960s and 1970s, the computer industry was dominated by mainframe and mini-computer systems, but offsite access was available on a limited basis using simple timeshare terminals. Communications were slow and the performance was limited but the customer required little in the way of in-house equipment or expertise. A plus factor was that business continuity was built-in, as the computer center was off-site. The big trend of the 80s was the PC revolution followed by networking. The 90s brought the Internet and high-speed communications, setting the stage for one of the hottest service products of the year 2000 — the Application Service Provider (ASP). The next year brought the .com crash of 2001. The crash also brought down many of the newly launched ASP companies and the term ASP was no longer in favor.

RT: *Max, you are a big fan of managed services. How did the model rise like a Phoenix from the ashes and become so popular in today's market?*

MS: The Yogi Berra quote in the headline is appropriate in that the terms ASP and Managed Services are very similar and this business model is hot once again. It was always a sound model but the launch of so many ASPs in 2000 was just bad luck and poor timing. The model allows companies to reduce overhead, outsource services plus protect their organizations from business interruptions. Managed Services covers all of the bases and today's converged IP communications solutions, combined with high-speed access, make a great bundled package for the enterprise.

Managed Services covers all of the bases and today's converged IP communications solutions, combined with high-speed access, make a great bundled package for the enterprise.

RT: *It is also a perfect model for any company looking to prepare for disaster by implementing a business continuity plan and many resellers have seized the opportunity to expand into this quickly growing market.*

MS: Exactly, and with all of the choices available, many enterprises need the advice of an experienced reseller to guide them though the wide selection of services. Concurrently many resellers need to partner with companies that provide secure facilities to offer an all-inclusive continuity plan to their customers.

RT: *Doug, how does Miami Data Vault (MDV) fit into this equation and what type of customers are you looking to serve?*

DB: MDV is a unique co-location center which also offers disaster recovery space within a 100,000-square-foot secure facility. Our fully redundant carrier fiber-optic connections are carrier neutral and we house 18 ISPs. We offer various platforms and flexible business terms and relationships. Our target market runs the gamut of enterprises looking to self-manage their business continuity deployments to resellers and other managed services providers looking to collocate in a secure facility. We are also open to partnering with other co-locations and data vaults to provide maximum geographic security for their managed services customers.

RT: *Lawrence, Standley's has a somewhat different approach that targets more specific markets that often require compliance with government regulations, like Sarbanes-Oxley and HIPPA.*

LB: Standley Systems realized that today's business environment demands new thinking about the management of electronic and printed communications. As a company, we have evolved from a traditional imaging company to a provider of services and applications covering a full range of document management and backup. Our partnership with Savin allows us to provide customers with the DocumentMall service, which is particularly suited to legal, medical, and financial organizations or divisions thereof that are document-centric. This service provides for managing and securing documents

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via a Citrix interface and Java "push" technology. It also allows companies to back up documents, yet retain immediate (and sharable) access. For customers that need full system backup, we partner with CoreVault Managed Services to provide disk-to-disk backup and recovery software for protecting an entire company's business-critical data, Server to Laptop. CoreVault maintains locations in Oklahoma City and Cheyenne, Oklahoma for maximum redundancy. Large file applications such as CAD/CAM require a large amount of bandwidth when transferring files; using the DocumentMall service with the Citrix interface allows employees to work from remote offices efficiently. Both of the aforementioned solutions comply with both Sarbox and HIPPA regulations. In addition, CoreVault uses delta coding technology for date-compliant restoration — which is a critical requirement for some enterprises

MS: *The one link critical to having an enterprise maintain continuity or failover to a managed facility is a secure communications link. Of course, the ultimate communication service is satellite technology. Rick, could you provide a quick overview of how SkyPort provides this critical link?*

RD: SkyPort is a licensed global broadband communication services provider for converged voice, video, and data via both satellite and terrestrial networks. SkyPort believes that secure data transmissions begin with a secure facility. Our Houston teleport and Global Network Operations Center are located at Ellington Field that is currently home to multiple Texas National Guard units, as well as the U.S. Coast Guard's Air Station Houston. The site is entirely fenced with multiple protection devices, and the area is patrolled around the clock by airport police. Our customer base, which includes FEMA, the National Guard, and major energy companies with drilling rigs deployed in the Gulf all demand a very high level of security. One of the services we provide is managed video conferencing and SatCom services for organizations like the Montgomery County Emergency Communications District.

RT: *Didn't SkyPort recently receive a prestigious award?*

RD: Yes, The National Guard Bureau presented the "Minuteman" Award to SkyPort International for outstanding and invaluable support to the National Guard's relief efforts following the devastation wrought in Texas, Louisiana, Mississippi, and Florida by Hurricanes Katrina, Rita, and Wilma. It was the first time ever that an external service provider received this award.

Rich and Max admonition: The major disasters of the past year tend to obfuscate the smaller and more common events that account for the largest percentage of business down time. Your company may be at a fork in the road where you are deciding if a business continuity plan is critical to your operations. The companies included in this article are only a sampling of the DPCF member companies and do not even include all of the members that provide managed services. If your company is interested in business continuity planning please visit http://www.tmcnet.com/channels/disaster-preparedness/ to view additional information provided by DPCF members, TMC, and the ECA.

And remember, as Yogi Berra once said, "When you come to a fork in the road, take it." IT

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

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

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

Current Members of the Disaster Planning Communications Forum

(as of 5/1/06)

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MSI Services
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By Mike Katz

The Next Wave — IMS Hype and Reality

The preceding article in this series covered the current battleground between the walled garden carriers and what the market is referring to as the virtual telecom operators (Skype, Google, Yahoo! and AOL). This month we take a look at the IMS value proposition in the market and some of the early adopter feedback from actual operators who are testing and deploying it. The reviews are mixed, at best.

IMS Hype Defined

Anywhere you turn these days in the telecom industry you'll hear someone mentioning IMS. (define - news - alert) It has become one of the hottest topics in telecommunications today. It has been collectively embraced by operators, network equipment providers, and analysts alike. Each group focuses on its own conclusions about why IMS is great. An example from In-Stat's Web site, *"IMS will deliver the 'Holy Grail' of convergence of access to multimedia services/applications across any end user device that all service providers are seeking to offer their customers in the future," says Henry Goldberg, In-Stat analyst. "But providers have a long list of challenges facing them that must be overcome to fully migrate to a converged network architecture for their entire wireline and wireless businesses."*

The important message here is that IMS, at its core, provides two immensely valued concepts — *ubiquity of access to applications via IP and reduced applications delivery cost through common billing and customer information control (via the HSS), no more silos of information!* This is great news for carriers that deploy IMS and network equipment vendors that sell it, but is it great news for consumers? That depends on how these benefits are exposed to end customers in applications that provide unique value and whether or not there exist insurmountable business and/or technical problems.

Intra-Operator, Interoperability is What Counts

Typical IMS solutions today are provided from a single vendor, top to bottom. This means that the various IMS layers, transport, session and control, and application services will most likely work seamlessly and provide the deploying carrier a solution to their needs. This is in effect creating just a "bigger" stove pipe, one that can serve a carrier reasonably well. However, this is not the originally hyped intent of IMS, which was to enable "best of breed" solutions to be integrated by an operator to meet their needs. Unfortunately for the near term and, perhaps, forever, single vendor solutions will likely be the rule not the exception. But this is not the biggest issue with IMS. Early test reports from operators like Cingular indicate that IMS — as a delivery vehicle for applications that need to follow users as they span multiple operators — has many real world challenges built in.

Remember the operator is purchasing IMS from one vendor to eliminate interoperability issues, so what happens when one operator, say Cingular, fields an IMS application to its subscribers, which they purchase and then proceed to try and access in a T-Mobile region. Well, it just doesn't work. There are both technical and business issues to solve here. For one, it's not likely that two different operators' open service access gateways (OSA-G/W) will interoperate and that their home subscriber server (HSS) content will be shared. Remember that each operator's HSS holds subscriber profile data, which they will guard with every fiber in their corporate body. So, for example if you're a Cingular customer and happen to subscribe to an IMS-based application service, it's likely not to work outside of your home network. Will consumers care? The answer is yes.

Examples include combined presence, location-based services, or being unable to authenticate the purchase of Digital Rights Management encoded music outside of your network. So intra-operator interoperability is a big technical issue and a big business issue. What two operators would like to sit down and give a competitor their most important private asset: customer data? It's not going to happen. What's needed is some sort of third-party clearinghouse, not unlike the services that interconnect GSM operators for roaming and billing today. So the least common denominator (a voice call) will always go through, but an application will not traverse the service layer of a foreign operator. Can this be fixed? It's possible, but many, many issues need to be thought through and agreed upon in the industry.

It's really about how well developed the technology is and how quickly the business realities of an all-IMS world can be resolved.

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IMS' Use of Session Initiation Protocol (SIP) is Great, Right?

SIP is an ASCII text-implemented signaling protocol that is easy to modify and extend to absorb new service models, which is part of the reason it was chosen to be used in IMS. It also has a lot of unique protocol methods that were originally developed for services in the VoIP world and are now being adapted to the IMS application space. There is a notion in SIP of the ability to register a "notify request" that reports back to the originator/requestor when some state has changed on the client's end — for example, when a mobile user is off the phone and available to take a call. Another lab report from Cingular clearly shows that the protocol overhead alone in conventional SIP methods like "notify" — if made available to a moderate percentage of users — would swamp the data carrying capability of most mobile IMS networks. That's just one example of how much more work needs to be done to define the reality in IMS.

Intra-operator interoperability is a big technical issue and a big business issue.

Making IMS Real

For operators and end users, IMS can bring positive values. It's really about how well developed the technology is and how quickly the business realities of an all-IMS world can be resolved. To be clear, there is an ominous message from outside the walled garden: solve these types of problems or alternatives will be brought in their place. An example of this is the recent MVNO announcement from Disney on top of the Sprint mobile network. It proves that it is possible to successfully deploy targeted mobile advanced applications without the need to wait for IMS. IT

Mike Katz is director of product marketing for NMS Communications. (news - alert) For more information, please visit the company online at http://www.nmscommunications.com.

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The New Rules of Enterprise IP Telephony

Say Goodbye to Your PBX, and Hello to Networked Applications and Open Standards Adaptability.

The only constant… is change.

No doubt you've heard that sentiment before when the topic is communications systems and the technology behind them. And it's never been truer than for business tools like the telephone and all the transformations PBX hardware has undergone over the last 40 years.

In their infant stages, before Private Branch eXchange systems became fully automatic, they were, as Newton's Telecom Dictionary refers to them, "electro-mechanical step-by-step monsters" that required callers to dial the phone company's operator to make an external call. Thanks to its Carterfone decision in 1968, however, the FCC broke up the local phone company monopolies, opened the door for new PBX development, and vendors including AT&T and GTE launched the first of roughly six legacy generations that have led to the digital PBX and auto attendant systems many businesses use today.

So where does PBX technology stand now? With the emergence of IP telephony and the move from traditional communications hardware boxes to networked software applications and open communications standards such as the Session Initiation Protocol (SIP) for VoIP, let's just say PBX systems have run their course. In fact, the downfall of the PBX has been on industry analysts' projection boards for some time. As far back as February 2003, for instance, Gartner, Inc.'s "Next-Generation: Enterprise IP Telephony" report stated rather inauspiciously that by year-end 2007, "traditional enterprise telephony system manufacturers will have ceased development of TDM-based PBX systems, and will have announced their intention to discontinue support within five years."

Life beyond the PBX, the "New Rules"

To many business owners and IT decision-makers, life without their trustworthy PBX phone system is a scary proposition — as is the thought of migrating to new technologies like IP PBX phone systems and SIP for IP telephony. Yet, the IP-based convergence of voice and data on a single network has opened a whole new world of communications and is increasingly making businesses and their employees more effective at collaborating, serving customers, and generating revenues than any previous generation of technology.

In other words, the rules of business communications have changed, primarily in that IP telephony and its open standards approach make the enterprise itself more readily adaptable to constantly-changing customer and market requirements. That said, your enterprise can either continue to conduct business as usual with a PBX that's soon to be extinct, or it can play by the New Rules of IP Telephony to set groundbreaking trends for how your business, employees and customers interact. Here are four such New Rules to think about if you haven't already said goodbye to your organization's PBX and made the move to IP communications.

OLD RULE: Dial-tone is all you need.
NEW RULE: The value is in the applications.

Some businesses and their workforce still need little more than a telephone to open the pipeline to customers. But now that e-mail, Web chat, and instant messaging have joined the list of multimedia options consumers insist on, it's safe to categorize such businesses as the minority. Which is where the New Rule comes in for value-adding applications. Rather than one hardware box after another for a PBX, ACD, automated attendant, Web server, chat server, IVR system, and on and on, pre-integrated application bundles allow an enterprise to replace costly, inflexible multi-box equipment that doesn't always cooperate across media channels.

Moreover with data running closer than ever alongside voice interactions to speed business processes and transactions, the new breed of "information worker" who interacts with customers (as well as partners and suppliers) must be able to access information quickly while on the phone or in a Web chat. Product info, pricing, CRM data, a customer's account record, supply chain inventories — that's where the New Rule comes in again both for data and voice applications.

On the voice side, several new IP PBX phone and communications systems such as the Enterprise Interaction Center (EIC) from Vonexus, offer pre-integrated client applications to manage queued calls and Web chats from the desktop, including client options for selected Microsoft Dynamics applications and Outlook. Choose an IP PBX system developed on an open standards software architecture and integrating the business applications your employees use most is also fairly seamless, which brings us to the next New Rule.

OLD RULE: Applications are what they are: Disconnected.
NEW RULE: Pre-integrated functionality… out of the box.

Imagine you're a customer contacting a business. All you care about is getting consistent quality service no matter what contact channel you choose and what transaction you

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want to perform. Because a PBX and other communications boxes aren't always fully "connected" themselves, an enterprise can't realistically leverage hardware to connect things like desktop business applications and customer databases.

> "IP telephony gives your enterprise the advantage of rapid adaptability."

Go back to many of the IP PBXs now hitting the market. Most enable an organization to unify data as well voice applications on a data network. In EIC's case, the EIC Server comes fully pre-integrated out of the box and fits directly on your network, where it's easily managed with other existing data and communications servers. Also reaching to the end-user level, pre-integrating to applications for IP telephony — as well as for things like CRM, ERP, accounting packages, screen pop, unified messaging, conferencing, etc. — greatly minimizes any chance of application disconnects when also integrating enterprise business processes.

Consider, too, that pre-integrated call queuing and routing in today's IP PBXs allow a business to more quickly and accurately queue calls, chats, and Web callbacks, and do so with no expensive external devices.

OLD RULE: "IP-enabled" is close enough.
NEW RULE: Buy an IP system, not "close enough."

If a proprietary vendor tells you their "new" PBX is IP-enabled to accommodate SIP and VoIP, run away as fast as you can. Besides PBXs being on the way out, the truth is they were never designed, and have never been redesigned, for open standards like SIP without having to bolt on more hardware. And even that approach isn't close enough for an effective IP solution.

If your enterprise is committed to finding a true IP communications system, look closely at its back-end architecture with regards to a SIP carrier environment for IP telephony. Does the system converge data and voice networks into a single IP-based network to reduce access? Does it sufficiently compress IP voice packets on the network to send more calls over a single circuit? Must you purchase gateways to convert VoIP traffic to TDM just to hand it to your telco? If the answer is yes to any of these questions, it simply isn't an IP system.

OLD RULE: To get more, you have to pay more.
NEW RULE: Greater functionality, cost effectively.

Ah, the age-old rule of proprietary vendor lock-in. "You want an IVR system and a CRM connector to supplement your customer service processes? No problem. But you have to buy a few more boxes and maybe some middleware, which only we can provide..." When it comes to adding features and functionality to a communications system, wouldn't it be much easier to simply activate a pre-integrated application via licensing and be done?

Perhaps more than any other benefit of a pure software IP communications system is the ability to activate applications virtually on-demand with a license for users. Again in EIC's case as a complete, pre-integrated IP PBX phone and communications system (phones included), licensing is all that's required to activate features such as desktop faxing via the system's fax server application. Same thing for EIC's IVR application if that's what you need.

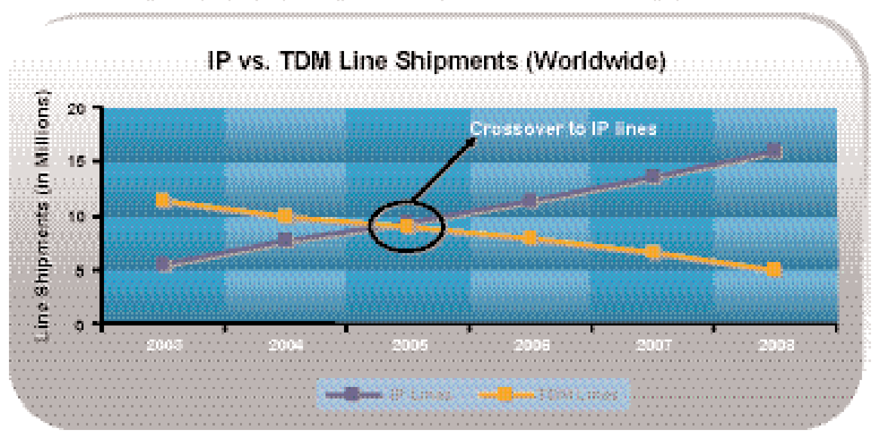
And if expanded functionality via licensing isn't cost-effective enough, just think about how much your enterprise can save with a software-only approach to IP communications. Centralized station device administration. In-house moves, adds, and changes, often in minutes and with no need for vendor maintenance contracts. And a single application server on the network instead of a bunch of high-priced hardware and proprietary devices. Most of all, your enterprise gets the cost-saving ability to purchase media services as you need them, without some proprietary vendor dictating your spending decision.

Business as usual? Or the "New Rules" of IP Telephony?

While the New Rules of Enterprise IP Telephony aren't set in stone, they do provide a good gauge of where the telecom industry is headed and how business communications technologies have changed. Whether your business decides to follow them, of course, is up to you. But one thought as a final word of wisdom: Business will always be about gaining an advantage, and IP telephony gives your enterprise the advantage of rapid adaptability for constantly changing market and customer requirements. Conduct business as usual with your age-old PBX, however, and you can kiss that advantage goodbye. IT

Joseph A. Staples is Senior Vice President of Worldwide Marketing for Interactive Intelligence Inc. For more on their suite of enterprise IP telephony and IP contact center solutions, contact Interactive Intelligence (news - alert) at 317.872.3000 (http://www.inin.com) and Vonexus (news - alert) at 888-817-5904 (http://www.vonexus.com).

Crossover from TDM to IP



Source: In-Stat, "IP PBXs Hitting the Tipping Point" May 2004

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

In this issue, we excerpt three of Rich's recent conversations with several industry leaders, including Inter-Tel's (news - alert) Craig Rauchle, Packet 8's (news - alert) Bryan Martin, and Verizon Business' (news - alert) Nancy Gofus.

For the complete interviews, please log on to our Web site at http://tmcnet.com/296.1

Craig Rauchle, President & Chief Operating Officer, Inter-Tel

RT: *Inter-Tel has caused a lot of buzz in the industry by announcing the Inter-Tel 7000, an IP-based communications system that can scale to 2,500 users. Why is Inter-Tel looking to enter a crowded enterprise field, which already features the likes of Cisco, Avaya, and Nortel?*

CR: Well, we already successfully compete against these companies in the small and mid-size market. Inter-Tel has earned a sterling reputation in this segment for its ability to tangibly improve business performance through advanced communications technology. Typically, companies choose to do business with us because of our ability to provide systems and applications that can help them address their most fundamental needs, such as increasing revenue, streamlining operations, and controlling costs.

Inter-Tel sees that many larger businesses face a similar set of circumstances. Unfortunately, most of the vendors currently occupying the enterprise still perceive their position as technology providers. Our philosophy — and our reputation — are built upon an ability to deliver tools that can positively impact these fundamental business processes.

And when we combine our applications focus with an innovative Managed Services program, we are comfortable that we can deliver a number of very compelling reasons for larger businesses to work with us.

RT: *Describe Inter-Tel's experience in selling VoIP solutions? Has the market growth met your expectations?*

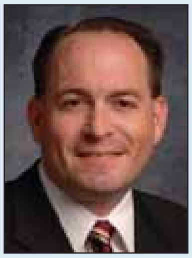
CR: I don't think anyone who follows our industry can contend the point that VoIP solution sales are noticeably increasing. Whether they are "dramatic" or "substantial" is open to interpretation. Inter-Tel's view is that VoIP sales are growing the way we initially expected: slowly, at first, and then escalating at a much quicker pace.

RT: *Is good technology a compelling enough reason to sell solutions, or do customers expect more?*

CR: Having an outstanding product portfolio is certainly very important, but there are a number of factors that come into play when we meet with customers.

Today's businesses are looking for a lot more than just hardware and software. They need consultative services to ensure their systems are configured and performing properly, and they want an ongoing relationship with a vendor that will address any issues that may come up. In addition, many businesses look for help with a migration strategy to leverage new products as they become available. But virtually every business we speak with wants a company that can demonstrate expertise in both voice and data environments.

That's why it is imperative that vendors have proficiency in assessing a network to ensure it has the proper bandwidth to carry not just voice traffic, but also video and other rich media applications. The demands on the network will change exponentially over the next few years, so it's critical that customers plan accordingly.



Bryan Martin, Chief Executive Officer, 8x8, Inc.

RT: ***Packet8 has its roots in video and has a slew of patents in the space. In fact, before many of today's VoIP companies existed, Packet8 was in the VoIP business. Having visited the company's offices over the years, I was consistently blown away at how many patents adorned corporate headquarters walls.***

The parent company, 8x8, was selling products to enable service providers to get into VoIP and, when the telecom meltdown took place, they decided to become a service provider instead.

I spoke with the company's Chairman and CEO Brian Martin recently about what his company is up to and what the future may bring.

How is the uptake of your service?

BM: The biggest concern is that people are suspicious. They perhaps heard a Cisco IP phone two years ago that sounded funny. This is one of the reasons we offer a 30-day guarantee. Customers need to order a minimum of three extensions, as less than that and you can't take advantage of all of our features.

Many companies take us up on this offer, and 14–30 days later they order the amount of phones they really need. We guarantee a better VoIP experience today and we let people try it and see for themselves.

The majority of people keep the service — 9 out of 10, to be precise.

RT: ***What is your biggest competitive threat?***

BM: In the near term it is Vonage (quote - news- alert) — as it has a recognized brand and many customers. Analysts will say that the cable companies are the biggest threat.

I think that, as mobility takes hold, the cable companies will have issues as well. In the long term I have much respect for SBC. This is where the battle lines will be drawn.

We need to steal some of their business customers. The longer war will be VoIP versus Ma Bell.

RT: ***What do you think of Kevin Martin?***

BM: We have done everything we can to comply with recent 911 mandates — we and others were waiting to see what the response was late last year. Nothing has happened; we have no guidance. The silence has him and others confused. We aren't getting feedback.

RT: ***Where will we be in five years?***

BM: Two major trends will dominate as we go into the future.

Video will become prolific as bandwidth is getting cheaper and more ubiquitous. In addition, wireless will continue to gain traction. I won't profess to know whether WiFi or WiMAX will win the war. I hope in five years the wireless issue gets worked out. We should all have gigabits of access and application providers will go nuts. Business productivity will improve and new entertainment applications will

Nancy Gofus, Vice President of Product Management, Verizon Business

come onto the scene.

RT: ***How are businesses benefiting from VoIP?***

NG: VoIP (define - news - alert) is reducing actual telecom cost by allowing for flat rate unlimited local and long distance. In addition, moves, adds, and changes are easier, as is wiring and management. Companies are using the power of feature functionality in very interesting ways, such as rolling out unified messaging and more collaboration. In addition, companies are using call blasting to multiple devices to ensure calls get through in a timely manner.

Enterprises are embracing what VoIP can do. They are getting creative and figuring out how to run their businesses differently.

RT: ***How will the telecom landscape look in the US in the next five years?***

NG: There will be two dimensions — as well as more and more management choices. Customers can chose to have service providers manage different parts of their network.

Customers are also looking for increased visibility on network as well as more hands-on control. Service providers will invest in portals. Companies will demand dynamic bandwidth to flex networks as they choose

In addition, wireless and wireline networks will come together, allowing us to talk on a particular device in a building and use the same device when we leave the office. This will happen with secure access to business tools such as CRM, SAP, etc.

Ethernet will continue to grow in importance, allowing connectivity from the WAN to the LAN. Verizon Business will introduce a virtual private LAN later this year, which will be Layer 2 end-to-end. All locations will look as if they are on a LAN nationwide. Everything will become IP and VoIP will grow. The same thing will happen in the toll-free market.

Finally, later this year Verizon Business will offer IP toll-free service, IP trunking, and IP routing. This will allow a customer to terminate across an IP trunk to a customer's IP ACD.

RT: ***One of the takeaways from this interview is the emerging trend towards wireless and wired integration. One of the biggest challenges corporations of all sizes have is managing wireless devices with disparate voicemail systems. Companies want to consolidate message stores and, perhaps most importantly, the greetings their customers hear. Individual cell phone numbers with non-uniform greetings are certainly not the right image any company wants to portray. The integration of the wireless and wired worlds will allow companies to have this seamless connectivity between these once disparate worlds. In addition dual mode devices will be more likely to be sold in the U.S. if an operator owns the wired and wireless networks, so industry consolidation seems to favor this trend.*** IT

IP PBX

Selecting the right IP PBX is an important business decision. There are many companies that market even more PBXs; the truth is that some are true IP PBXs and some are hybrid solutions, and some may incorporate most of the features and benefits currently available via IP communications, while others may not. Some solutions may require more additional hardware than others, and some are more scalable.

What matters is that any individual or company looking to make the switch to IP communications should be aware that there are countless viable options available and should do the appropriate research to ensure the end choice provides the maximum benefit for the situation. What follows is a listing of many IP PBX providers, with a brief summary of the offerings. We encourage you to use this as a starting point for your exploration of IP communications opportunities. Next month's round-up will feature an alternative to these on-premise PBXs — the hosted IP PBX.

3Com

http://www.3com.com

The 3Com (news - alert) NBX IP Telephony module delivers the intelligence, power, and flexibility for managing the most demanding communications needs of organizations in one or many locations.

The NBX V3000 is affordable with sophisticated features for SMBs with four to 200 stations. It is a complete telephony solution with a design that can scale to 1,000 users. The system includes four analog central office ports, one analog terminal port for a device such as a fax machine, four ports for Auto Attendant/Voice Mail (expandable to 72), 400 hours of message storage, Automatic Call Distribution, graphical Call Detail Recording, and Desktop Call Assistant.

The NBX V5000 provides advanced communications for up to 10 sites or 400 users. With all of the software features and scalability of the V3000 solution, the V5000 system adds the reassurance of redundant power supplies, hard drives, and uplink ports.

The NBX 100 delivers VoIP communications for stand-alone small businesses with minimal growth requirements, existing NBX 100 systems at other sites, or who require BRI support. It ships with four ports (expandable to 12) and 30 minutes of Auto Attendant/Voice Mail (expandable to 80 hours), Automatic Call Distribution, graphical Call Detail Recording, and Desktop Call Assistant.

AdTran

http://www.adtran.com

ADTRAN's (news - alert) NetVanta 7100 is a converged IP PBX switch/router that combines all of the networking functionality needed for voice, data, and Internet communications in a single, easy-to-use platform. The NetVanta 7100 sets a new price point for IP PBX systems that support up to 50 users at costs of up to 40% less than traditional multi-product VoIP solutions.

ADTRAN designed the NetVanta 7100 for SMBs that want the features larger systems offer, but that don't have the IT staff or budget to support a large, complex system. The SIP-based PBX includes voicemail, auto attendant, a 24-port Power over Ethernet (PoE) switch, full-featured IP Router, IPSec Virtual Private Network (VPN), stateful inspection firewall, and Quality of Service (QoS) to prioritize voice traffic. It also includes an intuitive Graphical User Interface (GUI) that allows users to easily make changes and add new phones. ADTRAN's optional n-Command network productivity software enables authorized partners to quickly and easily perform remote system backups and firmware updates for end users.

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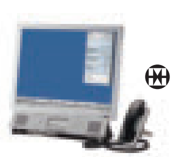

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Alcatel
http://www.alcatel.com

The Alcatel (news - alert) OmniPCX Enterprise is an integrated, interactive communications solution for medium-sized businesses and large corporations. It combines the best of the old (legacy TDM phone connectivity) with the new (a native IP platform and support for Session Initiation Protocol, or SIP) to provide an effective and complete communications solution for cost-conscious companies on the cutting edge.

Alcatel's solution is designed to improve productivity and enhance customer care, while reducing capital expenses and operations costs. It provides a high-availability platform (under UNIX and Linux), which delivers powerful communication tools, including a customer care center, business applications and tools that simplify daily administrative tasks.

The Alcatel OmniPCX Enterprise also features an embedded mobility solution, Web-enabled soft phone, feature-rich networking over all types of media, and resiliency mechanisms.

The Alcatel OmniPCX Enterprise is built around six value propositions:

- Architectural flexibility
- Intelligent networking
- Highest reliability
- Simplified management
- Agile workplace
- Superior customer interaction

Allworx
http://www.allworx.com

Designed for companies up to 100 users, the Allworx (news - alert) 10x system is a state-of-the art communications system that integrates three essential business operations into one simple system. It is a feature-rich phone system, a robust data network system, and a message center that substantially improves productivity.

The Allworx 10x provides enterprise-class features, such as custom call routing, presence management, call center control, remote user access and site-to-site connectivity. It is a phone system that integrates VoIP with a fully featured PBX and Key system, providing amazing flexibility and power to the small business. The system also supports remote users and you can connect multiple Allworx systems to create one phone system across multiple sites.

The Allworx 10x is flexible, scalable, and offers LAN and WAN access. It easily integrates into your existing network, or it can function as your complete network server. You can even create a VPN that provides secure Internet access for remote employees or directly connects two office locations.

The Allworx 6x delivers the same level of enterprise-class service and functionality for companies with up to 30 users.

AltiGen
http://www.altigen.com

Out of the box, AltiGen provides a high level of capability for the money. Every AltiServ system is prepackaged with software and licensing for the operator, extensions, and AltiView's call control software.

The system includes all standard PBX functionality, a comprehensive voice mail system, call detail reporting and an advanced auto attendant. The system can support all Voice over IP, analog, or a combination of both. The system is easily scaled in both size and capability. This inherent flexibility will allow your company to easily adapt the phone system to the way you would like to do business.

Based on your current needs and future plans, there are a selection of AltiServ base systems to fit every business. AltiServ phone systems are "All in One" products that can be easily administered on site or remotely.

Anta Systems
http://www.antasystems.com

Anta Systems (news - alert) has developed a turnkey VoIP solution that reduces the cost and complexity of VoIP deployment, enabling service providers and SMEs to reap the benefits of VoIP without unnecessarily large capital investments. Anta's Simplicity VoIP System is an end-to-end SIP-based VoIP solution.

The Simplicity SE-160 is a plug and play IP PBX designed for small businesses that has all of the traditional PBX features as well as a suite of enhanced applications such as auto-attendant, visual voicemail, 9-way conferencing and remote access capability. It is designed to be deployed in multi-site SMEs or as a Managed IP PBX solution by service providers.

The Simplicity ME-1000 brings to SMEs integrated functionalities of a media gateway, Softswitch, media server, application server, SBC. It is designed to enable VoIP service providers to quickly and cost-effectively launch basic residential and business VoIP services, as well as enhanced services such as IP Centrex, Visual Voicemail and VoIP Conferencing.

The ME-2000 for SMEs integrates the

functionalities of a media gateway, Softswitch, media server, application server, and SBC. It is designed to enable VoIP service providers to quickly and cost-effectively launch basic residential and business VoIP services as well as enhanced services such as IP Centrex, Visual Voicemail and VoIP Conferencing.

Asterisk

http://www.asterisk.org

Asterisk (news - alert) is a complete PBX in software that runs on Linux, BSD, and MacOSX, providing all of the features you would expect from a PBX — and more. Asterisk does VoIP in many protocols and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware

Asterisk needs no additional hardware for VoIP. For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsors, Digium.

Asterisk supports a wide range of TDM protocols for the handling and transmission of voice over traditional telephony interfaces. Asterisk supports US and European standard signaling types used in standard business phone systems, allowing it to bridge between next generation voice/data integrated networks and existing infrastructure.

Asterisk provides a central switching core, with four APIs for modular loading of telephony applications, hardware interfaces, file format handling, and codecs. It allows for transparent switching between all supported interfaces, allowing it to tie together a diverse mixture of telephony systems into a single switching network.

Avaya

http://www.avaya.com

Avaya (news - alert) IP Office is an all-in-one solution specially designed to meet the communications challenges facing SMBs. Due to its modular design, the solution can scale from 2 to 360 extensions to meet the needs of home offices, stand-alone businesses, and networked branch and head offices. IP Office supports a wide range of telephones, but the 5400 series Digital phones and the 5600 series IP phones have been specifically designed to work with IP Office and provide SMBs with a choice of solutions to meet business efficiency and customer service requirements.

IP Office is a highly integrated voice and data communication solution that aims to deliver within a single product solution the complete communication requirements of the SME customer — including many of the benefits enjoyed by the larger enterprises.

Avaya IP Office can easily scale up to 360 endpoints with more than 200 analog and digital trunks (up to 96 trunks; 192 analog trunks), giving small and medium size businesses room to grow.

Multiple Avaya IP Office systems can be linked together using a standard data network, providing limited rich transparency and advanced applications, such as centralized voice mail and call center. Avaya IP Office allows users access from desktop computers, laptop PCs, with individual firewall security and access control.

Aztech Systems

http://www.aztech.com

With Aztech (news - alert) IP PBX system IPX1050, calling an overseas office within the same organization is virtually free as it uses SIP to establish connections among users. A web browser-based tool will configure and assign telephone extension number to employees within an organization. Aztech IPX1050 is PoE enabled and does not require any power adaptor, as the Ethernet cable will provide the necessary power

The IPX1050 will induce significant cost savings by allowing customers to manage a single system for voice and data and the sharing of a single broadband connection. It scalable to 50 extensions and 8 PSTN lines, so customers can start with a system based on their current needs and then, as they grow, scale accordingly. It supports 50 voice mailboxes and 100 hours of recorded messages.

The system is also softphone and WiFi compatible, leaving room for today's convergent technology and comes with a convenient, easy to use Web-based installation tool. It also includes PoE support.

BlueNote

http://www.bluenotenetworks.com

(news - alert) SessionSuite IP Telephony, modular standards-based IP communications software, delivers voice/video as distributed communication services that can be easily incorporated into enterprises' business process applications.

SessionSuite IP Telephony offers real-time voice/video services as standalone IP applications that can be accessed over any network by any application or user. It leverages enterprises' exist-

ing data center infrastructure, integrating voice with the same back-office directory, authentication, authorization and accounting services employed for data applications. Optimizing the Internet, it economically extends the reach of services outside of enterprises' traditional "four walls." By combining industry standard SIP with Web Services, enterprises can easily integrate voice/video services into their business processes and applications within a Service Oriented Architecture (SOA).

Enterprises benefit from SessionSuite IP Telephony's ability to provide a forward-looking solution where voice/video can be tightly integrated into business processes, while leveraging existing infrastructure investment to offer high-value, applications-based communication services to users, customers, partners and employees.

Brekeke
http://www.brekeke.com

(news - alert) OnDO PBX is a software-based telephony system. It is SIP-compliant and Web-based for easy installation and management. The PBX includes all the features of traditional PBX systems, such as call transfer, call conference, call forwarding, voicemail, and much more.

Our OnDO PBX, SmallOffice and Standard Editions, which have multi-platform support, provide fully functional telephone systems often referred to as PBX systems. They are easily managed through a web-based administrative tool, and scalable. Both editions feature voicemail, call forwarding, call conference, call monitoring, call recording, and much more. You can install and use a fully functional IP-PBX, with OnDO PBX, quickly and easily.

OnDO PBX Version 1.5, Standard Edition, now supports Automatic Route Selection (ARS) failover. With ARS failover, OnDO PBX seeks an alternate route if the specified route is unavailable, and makes outgoing calls on the best route depending on the situation.

Cisco
http://www.cisco.com

Cisco (news - alert) Unified CallManager is a scalable, distributable, and highly available enterprise IP telephony call processing solution. It extends enterprise telephony features and capabilities to packet telephony network devices, such as IP phones, media processing devices, VoIP gateways, and multimedia applications. Additional services, such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems are made possible through Cisco Unified CallManager open telephony APIs. Cisco Unified CallManager is installed on the Cisco Media Convergence Server 7800 Series of server platforms and selected third-party servers.

Cisco Unified CallManager Version 5.0 enhances application delivery through the support of line-side SIP and SIP trunk-side enhancements. These enhancements facilitate increased interoperability with third-party applications and devices, and provide the foundation for supporting innovative new presence-based applications. In addition, Cisco Unified CallManager 5.0 simplifies deployment and management by supporting a Linux-based appliance model implementation. This version also features networking and administration enhancements, including support for Cisco RSVP Agent which enables more efficient use of networks.

Citel
http://www.citel.com

Rather than supporting multiple PBX systems and remote connections, Citel's (news - alert) EXTender products allow enterprises to connect remote call centers, home workers, and branch offices to a central digital PBX over an IP network, significantly reducing telecom operating costs and simultaneously improving business operations by providing single voice mail and call center applications, central reception, and four-digit dialing throughout the enterprise.

As enterprise customers become ready to complete the migration to SIP, the EXTender IP6000 can be software upgraded to accommodate an on-premise or service provider hosted IP PBX, leveraging the existing handset and wiring infrastructure at each location. This phased migration path allows enterprises to immediately realize the advantages of a central PBX platform, then complete the full migration to SIP telephony in the future.

The EXTender IP6000 expands the EXTender product line with a new lower price point and an assured upgrade path to SIP-based hosted or premise IP telephony in the future. The EXTender

IP6000 is available in a 12-Port configuration, which can be scaled to accommodate the number of stations at each remote site. The EXTender IP6000 is compatible with many leading PBX platforms, including Avaya/Lucent Definity and Magix, Nortel Meridian and Norstar, Alcatel, Ericsson, Iwatsu, Toshiba, and Panasonic

Colt Telecom

http://www.colt.net

Colt Telecom's (news - alert) IP PBX service can either be hosted in Colt's data centers or at the customer premises. It includes several hundred traditional PBX telephony features, in addition to new IP-enabled functionality such as softphone support and number portability. Additional services in development include video telephony, hosted voice recording and fixed/mobile convergence services.

The advantage to customers is they can realize the benefits of an IP PBX, while not having to deal with system management. Colt's pricing model is a flat-rate one, which includes an installation fee and a monthly charge for service.

Colt's service brings together the reliability, security, and features needed for business. The enhanced COLT IP Voice is now available as a stand-alone service or as part of the COLT Total service, which extends the power of IP telephony to a wider group of businesses, from small businesses with as few as 20 users to large organizations with as many as 1000 users.

This launch marks the second phase of Colt's "Total" converged voice and data service for SMBs. Additional rollouts will include video telephony, voice recording, and mobile services.

Dialexia

http://www.dialexia.com

Dialexia's (news - alert) line of enterprise solutions is a suite of products that address small business communications needs. For companies that have an established Local Area Network (LAN) and want to converge data and voice over that network, Dial-Office is the ideal IP-based PBX solution.

Dial-Office is an Internet alternative to the traditional circuit-switched enterprise phone system. It lets service providers deliver all of the features that companies are accustomed to in full-featured PBX or Centrex systems without the associated high costs. It handles internal calls, connects users to the worldwide phone network and sets up and manages their users and resources while offering a seamless integration of voice, data and video using a simple configuration and management tool.

Dial-Office is also suitable for multi-office connection, connecting branches which are geographically distant from each other. Built on Multi-domains Architecture, Dial-Office offers the flexibility of distributed location connections; unlike legacy PBXs, it offers employees a wide availability IP-Phone use in any location served by the company's Wide Area Network

Ericsson

http://www.ericsson.com

Ericsson's (news - alert) multi convergence communication system is the MD110 Convergence Communication System, a system that well and truly integrates fixed and mobile telephony, IP phones, PC softphones, cordless phones, mobile/cellular phones, digital phones and IP Gateways.

MD110 Convergence Communication System supports cost-effective, seamless communication across corporate voice networks, intranet, LANs, WANs, and public networks. Through convergence, enterprises will have the power to be productive by capitalizing on real-time, mission-critical business and communication applications. Now full IP Networking between MD110 and BusinessPhone enable convergence on existing IP connections, making for highly flexible work methods and better cost efficiency.

Dynamic Network Administration, D.N.A., gives IT managers the choice of managing the network from a single point or from multiple points. You can manage a network of any size simply by adding D.N.A. servers and linking these together over a WAN. And you can distribute management responsibility to designated individuals anywhere within the network, each with different levels of access.

FacetPhone

http://www.facetcorp.com

FacetPhone (news - alert) is an IP PBX telephone system that transforms the business phone system into software running on industry standard servers operating industry standard VoIP gateways. The system will never become obsolete, since individual components can be replaced or added as needed with the latest components.

FacetPhone comes with all the features you expect in a phone system, such as voice mail, auto-attendant, conferencing, callerID support, call detail recording, and call center features like automatic call distribution, call monitoring, call recording and call barge-in. In addition, with its graphical computer user interface, FacetPhone provides visual voice mail management, enterprise instant messaging, computer telephony integration with UNIX or Linux or Windows applications, presence and availability management, roaming extensions, and branch office and telecommuter support for toll bypass and true remote employee integration.

The FacetPhone architecture is based on a Linux or UNIX server, external media gateways, and standard analog telephones which make the phone system inherently flexible, reliable and cost effective.

Fonality

http://www.fonality.com

Fonality's (news - alert) PBXtra product line provides small businesses with enterprise-class phone systems for 40 to 80% less than the cost of traditional PBX systems. PBXtra is a complete PBX application for small business customers who want a phone system with enterprise-class capabilities.

PBXtra's enterprise-class features include telecommuting, branch office support, voicemail-to-email, click-to-call, VoIP, softphones, support for IP and analog phones, and advanced call center functionality. Fonality has streamlined and simplified the complex tasks of PBX setup, administration and management to make PBXtra the world's first enterprise-class phone system that can be installed and administered remotely using a Web browser, without specialized training. The PBXtra software runs on standard PC hardware, Digium hardware cards, and uses layers of Open Source Linux and Asterisk software to provide the least expensive option for small businesses deploying IP PBX phone systems.

Inter-Tel

http://www.inter-tel.com

Whether you need to connect several phones in an office, hundreds of phones in a building or on a campus, remote and telecommuting associates, or even geographically dispersed offices, Inter-Tel (news - alert) unifies your communications into a single enterprise that interoperates harmoniously. The Axxess Converged Communications Platform offers a flexible, feature-rich solution that is migration-friendly and Internet ready. The converged platform combines IP, wireless, digital, and analog into a single platform — enabling you to deploy traditional or IP telephony or a combination of both when and where it's right for your organization.

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

IPBX Systems

http://www.ipbxs.com

(news - alert) The LanPBX has all the features you would expect from a modern PBX, with a simple to use Web interface for both the users and management. Built on industry standards, it offers flexibility, ease of use, and all the features you expect. A new phone system needs to take advantage of modern communications methods in order to help increase productivity, save money, and communications.

The LanPBX offers unrivaled flexibility in the way it can be deployed, which allows for the ideal integration into your office environment, supporting multiple offices and remote workers with ease. The connection to the phone network and the physical configuration can be specified to integrate into your current IT practices.

LanPBX conforms to SIP standards, ensuring that new SIP-compliant products can be added to the LanPBX through its Web-based management interface. The VoiceXML scriptable LanMS can be used to introduce new services from traditional phone-based features through to new features based on the IP architecture.

Iwatsu

http://www.iwatsu.com

(news - alert) The Enterprise 2.0 Communications Server utilizes QuadFusion Technology to marry the four dominant communication protocols onto one platform. SIP, VoIP, TDM, and H.323 can be used alone or in tandem, making the Enterprise 2.0 a truly versatile system.

It converges voice and data traffic for higher cost savings, fewer hardware requirements, and more flexible bandwidth usage. Its reliable modular design allows small companies to grow up to 1024 ports with add-on features and applications. It can integrates various applications, including transparent networking, unified communications, contact center solutions, in-building wireless and more.

Web integration provides convenient browser-based system administration and reduced maintenance costs and flash-based software allows system updates from a remote maintenance console, eliminating the need to modify or replace hardware to support new software revisions.

The systems also peer-to-peer communication enabling IP phones to "talk" to each other directly and rely less on system resources.

Linksys

http://www.linksys.com

(news - alert) The SPA9000 marries the rich feature set of high-end PBX telephone systems with the convenience and cost advantages of VoIP. It

has common voice system features, such as an auto-attendant, shared line appearances, three way call conferencing, intercom, music on hold, call forwarding, and much more. The SPA9000 opens up access to the benefits of VoIP, including low-cost long distance service, telephone number portability, and one network for both voice and data.

The SPA9000 is so easy to configure that a fully working system can be set up in minutes. New telephones are automatically detected and registered when they are connected to the SPA9000. The SPA9000 has an integrated Web server that allows features to be configured using a Web browser. The Web server has multiple levels of password protected access to user- and service-level features.

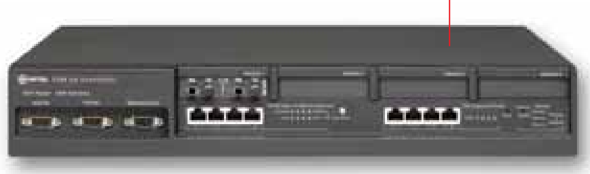
While the SPA9000 will work with any SIP compatible IP Phone, it is the ideal host for Linksys IP Phones. Powerful configuration capabilities enable the SPA9000 to support a greater set of advanced features with these IP Phones, such as shared line appearances, hunt groups, call transfer, call parking lot, and group paging. With its two FXS ports, the SPA9000 can support traditional analog devices such as telephones, answering machines, FAX machines, and media adapters.

Mitel

http://www.mitel.com

The Mitel (news - alert) 3300 IP Communications Platform (ICP) provides enterprises with a highly scalable, feature-rich communications system designed to support businesses from 10-65,000 users. The 3300 ICP provides enterprise IP PBX capability plus a range of embedded applications including standard unified messaging, auto-attendant, ACD and wireless. Operating across virtually any LAN/WAN infrastructure, the 3300 ICP provides seamless IP networking allowing for full feature transparency within distributed environments by supporting networking standards such as Q.SIG, DPNSS, and MSDN. The 3300 ICP provides organizations with the opportunity to IP-enable their legacy PBXs, protecting existing investments while delivering all the advantages of a converged infrastructure.

The 3300 ICP supports the industry's largest range of desktop devices including entry-level IP phones, Web-enabled IP devices, wireless handsets, and full duplex IP audio conference units. Mitel's Navigator integrates two of the most used business tools, the PC and the phone, to deliver real benefits to the user. The 3300 ICP also supports a powerful suite of applications including multimedia collaboration, customer relationship management and unified messaging. Industry standard Application Programming Interfaces (APIs) are supported for extensive third-party applications through Mitel Solutions Network (MiSN).



Motorola

http://www.motorola.com

(news - alert) The Centellis 1000 Series is a MicroTCA Open Application-Enabling Platform from Motorola. The Centellis 1000 series platform provides highly integrated and verified hardware and software components, reducing development costs and accelerating time-to-market. This allows telecommunications equipment manufacturers (TEMs), defense primes, and original equipment makers (OEMs) in a broad range of market segments and applications, to focus their development efforts on critical, differentiating features that provide a competitive advantage.

The Centellis 1000 series is designed to the draft specification of the MicroTCA open standard, making it physically smaller, with finer-grained scalability than Motorola's initial communications servers that are based on the AdvancedTCA industry standard. This fine-grained scalability enables MicroTCA platforms to support a pay-as-you-grow business model that allows customers to realize solutions with less capital expenditure and expand the computing platform capabilities in small, low-cost increments as demand for the new service increases. This advantage is particularly relevant to some of the new point-of-access applications such as WiMAX and IP PBX.

The Centellis 1000 family will be used in a wide range of applications, such as WiMAX access points, VoIP access gateways, and cellular base stations where reducing the capital cost of installing or extending next-generation network elements is very important. Small physical size, low power consumption, and enhanced serviceability also make these new communication servers ideal for a variety of applications in defense/aerospace, federal, medical, and industrial market segments.

NEC
http://www.necunified.com

NEC's (news - alert) UNIVERGE NEAX family of voice systems has been designed for a lifeline rather than a lifecycle. Relatively simple upgrades have kept its systems current in technology and functionality, while preserving 80% or more of the customer's existing investment.

The UNIVERGE NEAX 2400 Internet Protocol eXchange (IPX) fuses existing NEC technologies with dynamic advancements in hardware and software to satisfy the most stringent system requirements. All of NEC's networking services and Dterm Series E digital telephone features are provided when deployed over an IP network. Peer-to-peer switching is also introduced in the NEAX 2400, directly connecting all stations participating in a call to each other.

The SV7000 provides over 780 service features that enhance productivity, reduce operating costs, and improve communications efficiently. Innovative hardware and software design allows it to serve efficiently and grow incrementally over its entire size spectrum, ranging from 50 ports to 16,000 ports. Expanding from its minimum configuration to its maximum capacity with virtually no loss of existing hardware, The SV 7000 can grow in a cost-effective manner along with the user's requirements.

Nero
http://www.nerodigital.com

(news - alert) The SIPPSTAR IP PBX is an innovative, cost-effective, modern, software-based alternative to expensive conventional telecommunications systems that makes it possible for you to use innovative VoIP technology in your business.

You have the option of operating the SIPPSTAR IP PBX as an extension of your conventional system. To do this, you simply have to insert an ISDN card into the server on which your SIPPSTAR IP PBX runs, and then connect it via the S0-bus using a conventional ISDN cable. This operating mode provides you with significantly higher user scaling. You can also convert to the new technology gradually, without doing away with your old system.

Another option is to operate SIPPSTAR as an independent telephone system. If you wish to make your telephone calls via a regular ISDN connection, you need an ISDN card in the server on which SIPPSTAR is operated. In this way, the SIPPSTAR IP PBX functions as a gateway server and feeds calls via an ISDN card directly into your ISDN multiple device, system or primary multiplexer (E1) connection on the local telephone network. Your company can then also be contacted from the public telephone network at the previous number.

Nortel
http://www.nortel.com

(news - alert) Communication Server 1000 is a server-based, full-featured IP PBX, providing the benefits of a converged network plus advanced applications and over 450 world class telephony features. Fully distributed over IP LAN and WAN infrastructure with built-in reliability and survivability, Communication Server 1000 supports business-critical applications, including unified messaging, customer contact center, IVR, wireless VoIP and IP phones.

CS 1000 is designed to scale to meet growing enterprise requirements up to 15,000 IP clients per call server, multiple Call Servers networked with transparent IP networking. Built-in reliability based on VXWorks operating system and proven feature set with multiple resiliency mechanisms, including survivable Call Servers, Signaling Server redundancy configurations and survivable WAN gateways.

Extensive desktop portfolio includes IP phones, software phones, 802.11 Wireless VoIP phones, as well as digital and analog phones to meet diverse end-user requirements supports business-critical applications, including IP Contact Center, CallPilot unified messaging, and integrated services such as conferencing, one-number-follow-me Personal Call Director, recorded announcement, network-wide attendant and messaging

Pandora Networks
http://www.pandoranetworks.com

Pandora Network's (news - alert) provides Worksmart "on demand" PBX services, uniquely targeted at small to mid-sized businesses (SMBs). For the same price as a PBX-only solution, Worksmart provides an enterprise-class IP PBX, integrated with a comprehensive set of IP communications services.

Pandora's features include:

- Integration — Offers PBX integrated with VoIP, a desktop client, call routing, private IM network, public IM access, video services, web co-browsing, group collaboration, web contact center (integrated with Salesforce.com), ACD, ACD recording, ACD management and flat rate calling;
- On-Demand — Offered as a managed service vs. forcing SMBs to buy, integrate and manage expen-

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sive hardware/software;
- Low Risk — There are no changes to the SMB's existing systems, since Pandora can remain on top of the customer's legacy system. SMBs can simply unsubscribe to the service and return to their legacy solutions if they are unsatisfied.

pbxnsip

http://www.pbxnsip.com

The pbxnsip (news - alert) PBX is the perfect product for the small to medium-size enterprise. It offers all popular features of the PBX and maximizes the interoperability with existing vendors of SIP equipment. It works with most ITSP vendors that offer SIP services. pbxnsip customers enjoy:
- The look & feel of a traditional PBX,
- Connectivity to an ITSP or to a customer premises PSTN gateway,
- The ability to run infrastructure solely on IP,
- Vendor independence,
- Built-in security using the sips, srtp, and sdes standards,
- Built-in SBC for solving NAT problems,
- Call recording and paging functionality,
- Trunking to gateways and service providers,

The PBX offers two ways to connect to external devices. First, you may register SIP user agents (UA), for example hard or soft phones, FAX or other SIP-compliant devices like conferences systems. Second, you may connect to the outside world via trunks.

The PBX supports several kinds of trunks. You may connect a regular SIP PSTN gateway to the PBX and terminate your calls to the PSTN through this gateway. The gateway does not need to support advanced features like transfer or dialog stage. See the list of PSTN gateways that we were able to test already.

PBXpress

http://www.pbxpress.com

(news - alert) If your phone system is based on a legacy PBX using old style telephone lines, PBXpress can offer a range of affordable products that will enhance your corporate telephony infrastructure, while dramatically reducing overall capital and operating costs. PBXpress products are at the forefront of the next generation of VoIP-based PBX replacements that offer tremendous benefits by consolidating your organization's data and telephony capabilities to a single network.

PBXpress uses a reliable VoIP connection if it is available or it will fall back to a PSTN connection. In a PBXpress VoIP network each user is immediately recognized by handset or log on. Minimum investment is required to get started, since the PBXpress VoIP technology is embedded in your existing corporate data network. Subsequent telephony system expansion is as simple as plugging extra handsets into your existing LAN connections. PBXpress is software based and so upgrades can be performed remotely via the internet.

The wealth of enhanced telephony user features that arrive with PBXpress will increase employee productivity while the superb array of built-in call recording and archiving features will facilitate control of telephony traffic.

Samsung

http://www.samsung.com

Samsung's (news - alert) OfficeServe 7200 communication system is a completely converged platform supporting both voice and data communication with powerful IP-based wired and wireless flexibility. The Samsung OfficeServe 7200 will support traditional voice communication, VoIP, IP-based data, and wireless solutions through a wireless LAN

Businesses can deploy OfficeServ 7200 to build sophisticated telephony applications, secure data-communications infrastructure, and policy-driven networks. OfficeServ 7200 with its comprehensive range of features and functionality, offers an effective, affordable solution for any organization. Whether you are a small office, a main office, or a branch of a larger organization with a need to take advantage of cutting-edge solutions, the OfficeServ 7200 can be at the heart of your communications network.

The OfficeServ 7200 includes a firewall, intrusion detection/prevention, policy manager and packet shaper. By deploying both a firewall and an intrusion detection/prevention system (IDS) the OfficeServ7200 is particularly strong on network security. The IDS not only monitors network traffic but reports attempted security breaches to the network manager by way of an alarm.

The included technology ensures that voice will always be automatically prioritized. The OfficeServ 7200 also provides all of the telephony features and rich functionality necessary for mobility solutions and home working.

ShoreTel
http://www.shoretel.com

The ShoreTel (news - alert) system is a completely integrated IP phone system that scales seamlessly from one to 10,000 users including PBX, voice mail, and automated attendant functions.

The ShoreTel system is built from the ground up to be easy to use and manage full-featured IP PBX system. Its distributed architecture is ideal for multi-site companies that span multiple locations because their phone system appears and behaves as one, unified system.

ShoreTel 6 is the sixth generation of its distributed IP PBX voice solution, incorporating ShoreTel's management and integration capabilities and integrated software distribution, media encryption, on-net dialing, and increased support of international operations. With native SIP support, ShoreTel 6 provides increased flexibility supporting connectivity to third party devices such as WiFi phones, conference room phones, and gateways.

Office Anywhere extends the power of the ShoreTel system to remote workers without relying on the internet for voice quality. Remote users have all the power and productivity of all their Personal Call Manager delivered over their internet connection yet have the confidence of toll quality voice since the phone call uses the PSTN

ShoreTel 6 allows you to mix and match multi-vendor equipment with SIP support per the IETF's RFC 3261.

Siemens
http://www.siemens.com

(news - alert)
HiPath is a modular portfolio of multifunctional communication and security solutions. It makes it possible to set up a modern real-time communications landscape and ensures that employees are always accessible and that corporate resources can be accessed from any location.

HiPath solutions are based on open interfaces and compliance with common standards, so companies need only invest in those components they truly require. Existing systems can be migrated, new applications installed, and existing telephones adapted to current requirements through software updates. HiPath convergence solutions integrate the wireline and wireless worlds offering comprehensive options combined with an attractive return on investment and low Total Cost of Ownership

HiPath applications integrate seamlessly into existing business applications and deliver the necessary presence and status information. HiPath offers secure access to all resources, accelerating communications and decision-making processes.

The basis for a reliable and resilient communications environment, HiPath Real Time IP Systems transmits voice, data and video worldwide and over any type of network. This means service is never interrupted and productivity is enhanced. This goes for very small business networks, as well as huge global networks with 100,000 or more users:

- HiPath 1100 and HiPath 1220 for small enterprises
- HiPath 3000 for small to medium enterprises
- HiPath 3000 and HiPath 5000 for medium to large enterprises
- HiPath 4000 for medium to very large enterprises
- HiPath 8000 for very large global enterprises

SIP Foundry
http://www.sipfoundry.com

The sipXpbx (news - alert) solution is an open source enterprise SIP PBX, complete with voice mail, auto-attendant, and a host of features. It is suited for large and small enterprise use, as it supports fully redundant call control (HA) as well as a SOAP-based Web services interface. sipXpbx offers plug and play Web-based management, including integrated management and configuration of attached phones and gateways. sipXpbx is a modular server based solution that runs on standard Linux. sipXpbx does not require any additional hardware as it interoperates with any SIP compliant gateway, phone or application.

sipXpbx is a native SIP communications solution strictly following and implementing all the relevant SIP IETF standards.

sipX is an IP PBX, but it's much more than that — it is an ECS (Enterprise Communications Server). It handles voice and video, but its vision goes far beyond that. Real-time communications should be like email — a global interoperable system that allows for the exchange of real-time information of any kind (voice, video, IM, collaboration). A system based on presence throughout. A system based on SIP URI addresses that over time will replace PSTN phone numbers.

snom
http://www.snom.com

With the snom (news - alert) VoIP Box IP PBX solution there's no more need for expensive hardware components, such as gateways or servers — snom is breaking new ground with the VoIP Box.

The snom VoIP Box is a 9x8x4cm small complete IP PBX solution that's easy to attach to your LAN. It doesn't contain any moveable parts, like hard disks or ventilators, so it's durable and

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extremely low maintenance. It's also completely silent.

The snom VoIP Box can be completely configured through Web interfaces, by the customer or one of our partners. The configured software sits fully on the VoIP Box's built-in Compact Flash (CF).

Then simply connect the snom VoIP Box to the local LAN, using an Ethernet cable, plug in the Compact Flash, and switch on the power. After less than two minutes lyour whole telephone system is up and running — with all the performance features of a telecommunications private branch exchange which, in addition, can be extended over Web interfaces.

Sphere
http://www.spherecom.com

(news - alert) Sphericall IP PBX is a business communications application that runs on industry-standard servers across your existing data network without expensive proprietary hardware.

It is an open system solution that supports standards-based business telephony devices, gateways, and other communications endpoints, giving you the greatest flexibility and choice to create a customized enterprise communications solution to fit your unique needs.

Through its unique distributed software architecture, Sphericall IP PBX scales to 30,000 ports and achieves 99.999% reliability with no single point of failure. Rather than simply clustering servers to achieve redundancy, Sphericall IP PBX is built upon a distributed software architecture that enables you to achieve 99.999% reliability in every location.

Sphericall IP PBX offers Sphericall Web Services, providing a rich set of communications services for integration with other enterprise-class business applications. With Sphericall Web Services, the IP PBX becomes an enterprise softswitch that is utilized by various business applications to integrate communications into your business processes.

The result is an advanced, proven enterprise communications platform that seamlessly integrates with your business. Now you can stop thinking about your phone system and start leveraging integrated communications to create value for your business.

Sphericall IP PBX is the world's most flexible, scalable, reliable and cost effective IP PBX.

Switchvox
http://www.switchvox.com

Switchvox (news - alert) is everything that you don't expect from a PBX. It's truly affordable, easy to set up, simple to configure, and a breeze to maintain. It has features that let your business run more effectively and with fewer hassles. And it does all of this for a fraction of the cost of the PBX dinosaurs of the past.

Switchvox is VoIP enabled, can use basic telephone equipment, and almost any sized business will find it's a perfect fit. The Switchvox PBX interoperates seamlessly with traditional standards-based telephony systems and VoIP systems, so you can call anyone in the world regardless of their phone, PBX, or phone company.

Setting up your Switchvox couldn't be any easier when you use VoIP because no additional hardware is needed. You can set up your PBX to use VoIP in a matter of minutes.

Your Switchvox PBX supports SIP, the technology choice for real-time communication session control throughout the Internet, corporate networks, and within next generation wireless networks.

Tadiran
http://www.tadiran-us.com

Tadiran's (news - alert) Coral IP-enabled servers allow you to add functionality and capacity as your business needs change and grow, helping to minimize your total infrastructure cost over time. Designed to provide unparalleled scalability, Coral's voice and IP-enabled features are completely compatible from the smallest system to the largest. Whatever the size of your business, there is a Coral system to deliver the bottom-line benefits you expect from advanced networking: lower networking costs, improved customer service, and increased employee productivity.

- Coral6000 — High traffic duplicate processors for large businesses needing 6,000 ports and critical component redundancy;
- Coral 5000 — Duplicate processors for medium to large businesses needing 5,000 ports and critical component redundancy;
- Coral 400 — Full-featured communications in a mid-sized platform supporting up to 384 ports;
- Coral 200 — Powerful, state-of-the-art communications for the small business needing up to 200 ports;
- Coral FlexSets — Digital telephones to complement your Coral system;
- Coral QNet — Build a converged network with the QSIG global standard incorporated in your Coral system;
- Coral FlexAttendant — Make every customer's call a smooth, professional experience by delivering information directly to telephone attendant PCs.

TalkSwitch
http://www.talkswitch.com

TalkSwitch (news - alert) is a new-breed, all-in-one phone system designed specifically for businesses with 1 to 32 phone users per location. Its big-company features and intelligent capabilities work right out of the box and are a snap to customize.

It integrates standard analog telephones, fax machines, cellular and off-site phones with the traditional telephone network and the Internet for VoIP. TalkSwitch delivers a world of connections, easy configurability and unprecedented prices.

TalkSwitch comes with features that many other companies charge extra for and can be up and running in an hour. There also is no need for specialized telephones; TalkSwitch is compatible with standard analog telephones, so it works with existing phone sets — corded or cordless.

TalkSwitch is a SIP-based hybrid phone system that can be connected to both the traditional telephone network and the Internet. With broadband access, and the optional VoIP module, businesses can place branch-to-branch calls over the Internet and access SIP-based service provider networks.

Built with an upgradeable, modular architecture, expansion with entails snapping in an expansion board or connecting additional units over the LAN. TalkSwitch does the rest.

Toshiba
www.toshiba.com

The Strata CIX family includes modular, scalable, and networkable IP systems that provide sophisticated business communication features for businesses of all sizes. These systems deliver on the promise of IP telephony by providing all the features and benefits of our traditional business communications systems on a converged IP platform.

Because Strata CIX systems support all types of telephones, they provide the configuration flexibility to build the communications system you need, in addition to the investment protection from re-using devices from other Strata systems. That's why the Strata CIX is much more than just an IP system. It is a unified communications environment that supports all types of client devices.

Toshiba's Strata Media Application Server supports voice processing and all value-added applications integrated within one platform that connects to the Strata CIX via Ethernet. Applications include:

- Auto Attendant
- Voice Mail
- Automated Speech Recognition (ASR)
- Text-to-Speech
- Unified Messaging
- Interactive Voice Response (IVR)
- Automatic Call Distribution (ACD)
- ACD Reporting
- Toshiba-approved 3rd party CTI applications
- Info Manager Web-based telephone applications
- FeatureFlex adaptability tools
- eManager browser-based system administration.

Vertical
www.vertical.com

(news - alert) InstantOffice delivers the features most in demand in an IP PBX, including voicemail, conferencing, auto-attendant, and integrated fax. Equally important, it accommodates the budget and staffing constraints of smaller organizations, allowing you to dramatically enhance the quality and efficiency of voice communication while keeping a tight lid on costs.

InstantOffice allows your organization to consolidate voice and data communications on a unified platform that is easy to use, easy to manage and economical. The system provides VoIP capabilities, integration of voicemail, e-mail, and fax, and integrated networking — all delivered using a single, cost-effective T1 line.

The InstantOffice PBX provides a comprehensive set of features for handling calls promptly, efficiently and flexibly. Easy-to-use functions such as call transfer, call waiting, off-site call forwarding and paging through speakerphones and overhead systems help ensure that calls reach the intended recipient, even if that individual is away from their desk or out of the office.

InstantOffice supports a wide variety of analog and digital endpoint phones, giving you the flexibility to deploy the best and most cost-effective device for the purpose at hand, all using a single infrastructure.

With an extensive set of customizable features, the system can be tailored to meet the unique and fast-changing needs of your organization.

Vodavi
http://www.vodavi.com

(news - alert) Available for businesses seeking cost-effective mobile applications for remotely dispersed workforces ranging in size from 10 to 250 station users, the TeleniumIP system with Generation III software is based

on distributed architecture that's completely LAN-based with Web-based system administration and maintenance.

TeleniumIP offers a full suite of user applications that improve communications and deliver mobility required to network multiple offices, home offices and remote buildings together for a cohesive communications solution. Enhanced capabilities include the IP7000 Series of Desktop Telephones and Soft Phone IP Endpoints that improve mobility and deliver voice and video capability on a laptop PC or PDA. Enhanced calling features, integrated ACD and enhanced system administration features are included. A wireless IP (WiFi) handset that operates via wireless access points will be available in 2006.

Vonexus

http://www.vonexus.com

(news - alert) Traditional PBX phone equipment was never made for VoIP and open standards like SIP. Enterprise Interaction Center, however, is a pre-integrated IP PBX application suite and phone system designed for VoIP and SIP out of the box. More than that, EIC gives your enterprise everything it needs for "new-age" business communications.

With EIC, customers get an integrated SIP-based media server that offloads the processing of call recording and other media operations, allowing enterprises to leverage SIP's network scalability. They also get a bundled software suite for IP technologies that favor software over hardware, and that incorporates a cost-effective disaster recovery solution.

Vonexus developed the EIC IP communications and phone system software exclusively for businesses using Microsoft's product families. But EIC is truly designed for any company looking to lower costs and gain long-term investment protection by deploying VoIP and Microsoft-based standards using the SIP-architected EIC IP communications solution.

And whether your enterprise is centralized in one location or looking to consolidate distributed branch offices, regional offices and remote and mobile workers on a single IP communications platform, the EIC solution is powerful enough to accommodate your entire organization.

Whaleback Systems

http://www.whalebacksystems.com

Whaleback Systems (news - alert) has leveraged cutting-edge technologies and the latest industry standards to engineer a software-driven PBX that transports voice signals over a broadband connection instead of a traditional phone line.

Our software-driven solution seamlessly integrates with your IT network to simplify system management, streamline communications, save money, and provide the advanced features your company needs, including desktop messaging, video calling and our exclusive Road Warrior Functionality. Best of all, Whaleback's all-inclusive service package makes the most advanced technology immediately affordable for small and mid-sized businesses.

The Whaleback SMB 1500 solution was built from the ground up specifically for broadband. It also is 100% premise-based and software-driven to simplify system management and make advanced features easy to use. There's no equipment to purchase and upgrades are free so you do not have to worry about obsolescence.

Whaleback's exclusive Road Warrior Functionality feature connects customers with employees, regardless of their location, and gives mobile employees full access to the system, which streamlines communications and improves productivity.

Zultys

http://www.zultys.com

The three (news - alert) Zultys systems — MX250, MX1200, and MX30 — are fully integrated IP PBX systems based on open standards. Each system can autonomously provide full PBX functionality enhanced with real-time collaboration applications. Customers can cost-effectively deploy an enterprise class VoIP platform for 5 to 10,000 users.

The MX30 is a powerful system that enables multimedia communications for small offices. By integrating the functions of many devices into a compact box, the MX30 simplifies the VoIP network. It is a comprehensive solution that is easy to install, use, and maintain.

The MX250 makes VoIP affordable for SMBs or branch offices. It scales from 5 to 250 users without additional hardware, combining the functionality of a PBX, voice mail server, voice gateway, and Internet gateway. Built with the same technology and the same productivity tools as the MX1200, it provides smaller sites with premium features at attractive prices.

The MX1200 is a truly converged product that makes VoIP viable for the enterprise. This single 2U box brings together the functionality of a PBX, voice mail server, switch router, and Internet gateway. With a host of productivity applications, such as IM, presence, and PC voicemail management, the MX1200 can streamline communications for facilities with 25 to 1200 users.The system is 100% based on open standards, embedding technologies such as SIP, TAPI, VoiceXML, and Linux. IT

It's not Voice over IP! It's Voice in the PC!

By David Mandelstam

Some weeks ago I attended at a conference session where the topic was Real World VoIP Applications.

A large manufacturer of telephony equipment gave as an example of such an application the case of an airport installation in which his equipment had been used. One problem at this airport had been that gates were rigidly assigned to airlines. This meant that should the flight mix change, Airline A could not make use of an idle slot in the case of congestion, because it "belonged" to Airline B. All the signs, electronic notice boards, airport control information, and airline systems had each gate rigidly designated, and the assignments could not readily be reassigned.

Enter VoIP (define - news - alert) technology as the magic technology to resolve all this. At the end of the upgrade program, a ground staff attendant for Airline B could simply type in a code in his telephone at a departure gate and instantaneously the phone became his own, with his personal extension and all systems became aware that this gate now belonged to Airline B. Passengers were directed to the new gate. What a VoIP success story.

Consider what the costs of this project included. Well, it was VoIP, so surely a nice new set of Cat 5 network cables was run throughout the building. The airport authority surely got an opportunity to invest in all the latest switches and routers, certainly with a good supply of uninterruptible power, probably some PoE. They bought all new IP telephones to replace those analog telephones that had been connected to their old PABX system. Of course, the PABX itself was replaced. Just the investment in the VoIP portion of this project must have cost the airport a tidy sum.

But, in fact, the project had little if anything to do with VoIP. It had to do with getting telephony information in and out of computers, and using the information to control other computers. And all those computers were very likely PC servers. The project could have been done retaining those original phones, using the original telephone wiring network and the original PABX (also a PC). All that was needed was access to the dialing codes from the PABX, a fairly trivial exercise.

Fortunately for all, the airport authority is able to charge one of the world's highest range of landing fees, so they were able to afford all this. The supplier and their consultants and technicians all made lots of money, and everyone lived happily ever after.

But what about you in your small business?

Do you have to shell out big bucks for wiring, switches, new phones, new PABX, and costly VoIP consultants just to be able to implement some simple voice/data integration in your own business? To have your CRM system automatically pop up a customer's profile when she calls in? To have your PBX automatically adjust your voicemail depending on where you are, what you are doing and who the caller is? To have voice recordings saved in your customer database?

The answer is no. All you need to have is simple, inexpensive soft telephony technology.

VoIP is nice, it is cool, it is the flavor of the month, but it is totally irrelevant to voice data integration. We at Sangoma have a unique overview of the soft telephony industry, as we supply the premium voice handling hardware used in virtually all the open source and several commercial soft telephony projects. While most of the larger projects have a VoIP component, many of the more innovative, ingenious systems that are breaking ground in the ideas related to voice and data integration have pure PSTN interfaces.

We have never seen a voice/data convergence project or Interactive Voice Response (IVR) system that was dependent in any way on VoIP for its success. It is totally irrelevant in

what form the soft telephony system gets its voice and control information, as long as the audio and the normal caller and calling ID can be captured. You can use your old PBX and its phone and wiring, front it with a PC-based soft telephony system and quite easily implement any innovative telephony based data and control system that you can dream up. Of course, that is not to say that VoIP is not also appropriate in many cases, but the additional expense is never trivial and needs to be justified.

The soft telephony technologies, particularly the open source ones, provide a platform that has almost infinite flexibility for integration of voice with other technologies. The Asterisk project, in particular, has the most built-in applications to deal with the more standard telephony applications like conferencing, follow-me, automatic e-mail notification of telephone messages and customized voice mail responses. But these applications, though considered leading edge by the traditional telephony industry, are actually quite mundane. It is the combination of easy access to voice streams and DTMF or other signaling codes, plus caller and called ID that allows an almost infinitely programmable platform for voice and data manipulation.

Armed with a simple PC, some inexpensive interface hardware with the highest port density possible for future expansion, and, if necessary, a Linux administrator to handle the integration, you too can implement a complex IVR or voice integration project that is limited only by your imagination. And, all this without a forklift replacement of your existing phones, PBX, and wiring.

The beauty of this approach is that it also provides a gradualist replacement and upgrade path for your old system. All the features of the new soft telephony system, such as Internet-based long distance, are available to all your existing phones. As new phones are required, you could install the latest VoIP phones (simply to take advantage of the great phone features that are available at very reasonable cost), or something as simple as $10 analog phones. You are no longer trapped in the world of proprietary systems. IT

David Mandelstam is President of Sangoma Technologies. (news - alert)

Sangoma's Latest & Greatest

By Greg Galitzine

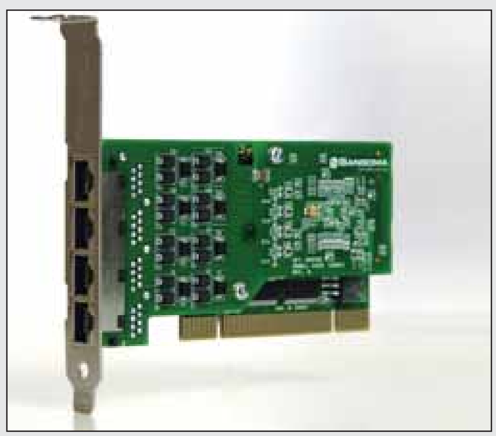
Sangoma Technologies has been designing WAN and telecom/telephony hardware for over 20 years. While Sangoma is well-known as a world leader in support of ATM, Frame Relay, SS7, X.25, PPP, BiSync, HDLC, and SDLC — all popular WAN-related protocols — it has been quietly making inroads in providing analog and T1/E1 hardware to open source PBX solutions such as Asterisk, Yate, and FreeSwitch.

In a recent review of Sangoma's hardware, TMC's Tom Keating explained that they manufacture a range of PCI-based cards with T3/E3, T1/E1 TDM, analog voice and data, ADSL, and serial interfaces. Their cards can turn a server into a voice or data gateway, and their voice and data solutions and communications toolkits are available for all popular operating systems including Linux, Windows, Netware, FreeBSD, OpenBSD , NetBSD, SCO Unix, and Sun Solaris.

The most recent addition to Sangoma's stable of product offerings is the A108 Series of octal T1/E1 cards, designed to support high-density data and voice applications. The A108d version includes a miniature echo cancellation and voice enhancement sandwich board.

The Features of the A108 cards include:

- The highest density T1/E1 PCI supporting the business, making it ideal for high-density routing and TDM voice applications, saving half the PCI slots required.
- Much lower power consumption per system than comparable systems made up of less dense cards.
- Optional hardware DSP on the A108d, offering G.168-2002 echo cancellation with 1024 tap/128ms tail per channel on all channel densities, DMF encoding/decoding and tone recognition and voice enhancement.
- The same PCI interface, architecture and digital path as all Sangoma's AFT voice and data cards, meaning no motherboard or compatibility issues and proper handling of shared interrupts.
- Round robin data buffer handling to minimize interrupts when supporting high volume/small packet data environments.



All Sangoma's voice drivers take advantage of their AFT (Advanced Flexible Telecommunication) technology to substantially reduce the processing required to handle voice calls by the host CPU. This reduces the CPU's workload; resulting in fewer dropped calls, less jitter, and better voice quality.

In his review, Keating pointed out that Sangoma also stressed a key competitive advantage in that their AFT technology enables them to not only field upgrade the device driver code, but also the card's FPGA firmware. In addition, Sangoma's cards are self-sensing for 3.3v and 5v PCI slots and software configurable for T1/E1 or J1. According to Sangoma, they share interrupts properly between themselves and other PCI compatible devices, supporting unlimited numbers of cards per PC chassis. IT

Greg Galitzine is the editorial director of Internet Telephony.

Building the Right Communications System

Advanced Building Products (ABP) is a professional wholesale provider of exterior building products. It has been serving the State of Louisiana for decades from its three locations in New Orleans, Baton Rouge and Lafayette and remains true to its long-time dedication to craftsmanship, quality, and service.

The Challenge

In 2003 ABP began planning for a move to a new location in Harahan, Louisiana, just outside New Orleans. Faced with the decision of whether to move an old, antiquated phone system to the new site or upgrade to a modern one, the company easily reached its conclusion — a new communications system was in order. So, the search was on for a phone system that could be administered in-house without having to frequently call in outside help for minor changes. At the same time, ABP wanted to plan for a way to substantially cut long distance charges, especially for calls made between their own locations.

The Solution

Late in 2003, Advanced Building selected and installed FacetPhone in its New Orleans location. In fact, it already was a user of FacetCorp's FacetWin — an all in one Windows to UNIX/Linux connectivity product — and FacetTerm products — a session manager product. Based its satisfaction with the other products, ABP had confidence in FacetCorp's product quality and customer service. Other factors taken into consideration were the ease of administration, call detail recording, multi-location integration, integration with computer applications, and the ability to "see" all the employees' activity and status through the graphical user interface and presence management.

Then, in the summer of 2005, ABP deployed FacetPhone at its Lafayette and Baton Rouge locations and subsequently connected those to the New Orleans office over the Internet with FacetPhone's VoIP solution. With FP's four-digit dialing, users can now just dial an extension to reach someone in a different location. Not only is the inter-office communication simpler and easier, but the company is saving considerable dollars in long distance charges from internal communications as well as from outside calls with FacetPhone's least cost routing.

> "We really appreciate the ease of use of FacetPhone and the ability to completely administer it ourselves, rather than needing to call in 'the phone guy' every time we need a minor configuration change."

"We have recently installed FacetPhone at our other two locations and, by using FacetPhone for our inter-office calls, we expect a savings of about $1000 per month in long distance charges," noted James Johannesen, ABP's General Manager. "In addition to these hard savings, it is now much easier for our employees to contact co-workers at other locations by simply dialing their extension." IT

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VoIP in the Family... Business

The Challenge

A family owned business launched in 1938, Pacific Lumber is one of the four companies owned by the Morse family. The family has eight locations, including lumber yards, truss plants, door and millwork manufacturing, and sales offices. Their 300 employees cater to large and small home builders in Oregon and Washington.

About three years ago, after opening a new lumber yard in Bend, Oregon, the costs to operate and maintain their legacy phone system soon escalated beyond what they could tolerate. Though the telephone equipment was paid for and worked as advertised, even small changes were complex, expensive, and time consuming.

Alan Churchill, Director of MIS at Pacific Lumber, began looking at possible system replacements. He hoped that moving to an Internet Protocol (IP) telephony solution could provide potential savings in administrative and maintenance costs. Key to the project was the need to connect all locations on a single IP network. “We wanted one person answering the phone for all yards at Pacific,” said Alan. “Plus we needed a system that was cost-effective and easy to manage. We were looking for a phone system that we could simply plug into our existing wide area network (WAN),” he continued.

The Solution

Alan contacted several major VoIP solution vendors. He also contacted Greg Still at Xiologix LLC, a technology solution provider in Tualatin, Oregon, who presented solutions from Zultys Technologies. (news - alert) Alan’s primary objective was to find a system that was low cost, simple to administer, and easy to use. After receiving the proposal from Xiologix, he nearly dismissed it right off the bat.

“We almost didn’t pursue Zultys, mainly because it was half the cost of the others. It just seemed to good to be true,” admitted Alan. “But when we contacted other users and tested it in our own lab, we discovered that it really did outshine the others.”

Pacific Lumber installed an MX250 IP PBX system from Zultys as well as ZIP 4x4 IP business phones. The implementation took a total of two days, which impressed Pacific, especially the fact that administration could be carried out on the entire system from a remote Web site, if necessary. Churchill noted that Pacific’s previous system required a truck roll for each little issue.

Zultys’ end user client interface, MXIE, provides employees with significant productivity gains. With a simple point, click, and dial functionality, users can initiate one-button conferencing. The call handling rules of the Zultys system automatically passes calls to available operators, depending on the caller and his needs. Essentially, the systems ensures that customers are able to contact Pacific at all times, including while off-site, a feature made possible by the ability to route calls to cell phones. Internal communication is also considerably easier with simple four-digit dialing. And for faxes, each employee has his own DID fax number, which makes the process considerably more efficient.

Alan is happy to have found a system that was so easy to integrate into Pacific’s existing IT infrastructure. “Every other vendor wanted us to change our switches and routers,” explained Alan. “We had no issues with QoS and the pricing was simple.”

So, has Pacific Lumber been able to calculate the ROI on the Zultys system? Alan describes it thus:

“We chose to purchase the system outright, but if we had decided to lease, the lease payments would be less than we were paying for support on the old system. We figure we are ahead starting with month number one. It proved to be exactly what we were promised: a system that was simple to implement, easy to use, worked with our existing switches and routers, and cost less than we were paying. The real proof that the system is a good one is that we have installed a second MX250 with Zultys 4x4 and 4x5 phones at Canby Builders Supply (another family-owned company).” IT

UMA and IMS: In Network Evolution

By Steve Shaw

Unlicensed Mobile Access (UMA) has become a hot topic for global operators and service providers looking to address the opportunity for Fixed/Mobile Convergence (FMC) and, more specifically, mobile/WiFi convergence (MWC). This relatively new technology has suddenly found itself compared with IMS, creating confusion in the marketplace where there is little understanding of how the technologies compete, co-exist, or potentially enhance each other's role in the network. This article will clarify the different roles that UMA and IMS technologies play with respect to convergence for mobile operators, as well as for integrated operators that own both fixed and mobile access networks.

Today, mobile operators are looking to capitalize on the revenue potential of IMS-based data applications like streaming video, music downloads, interactive gaming, and videoconferencing. At the same time, they are challenged to optimize such next-generation applications in light of limited bandwidth, high latency, and the relatively high cost of cellular radio access networks (RANs). The economics are clear: the faster the network, the better the user experience and the better the user experience, the higher the adoption rate and average revenue per user (ARPU).

UMA is the 3rd-Generation Partnership Project (3GPP) standard for enabling subscriber access to mobile services — including IMS applications — over WiFi and broadband IP networks. UMA turns existing Wireless LANs (WLANs) into seamless extensions of mobile wireless networks, enabling subscribers to automatically roam between the cellular network and a home,

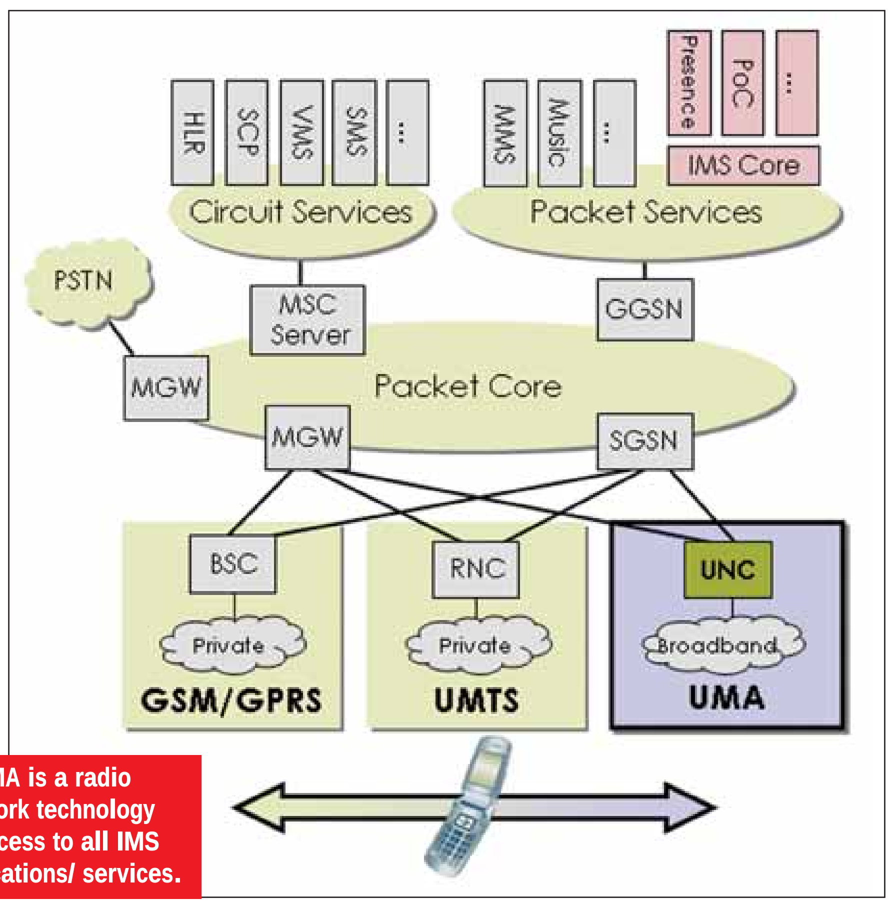
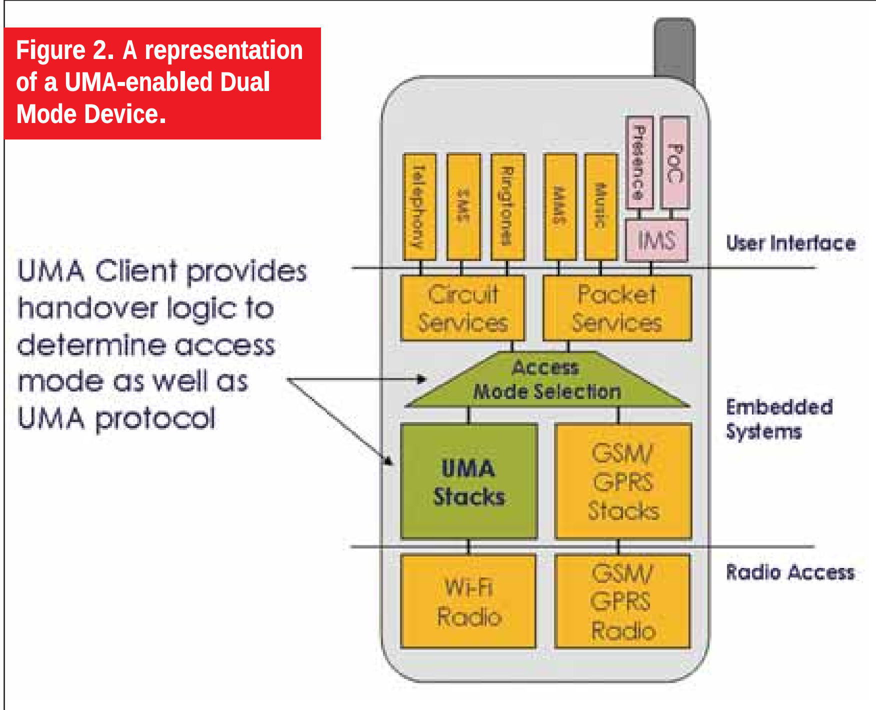


Figure 1. UMA is a radio access network technology providing access to all IMS based applications/ services.

Figure 2. A representation of a UMA-enabled Dual Mode Device.



office, or public WLAN, with full access to all applications from every location. With UMA, IMS-based applications are available to subscribers wherever they roam.

UMA Design Goals

UMA was designed to leverage existing, standard interfaces to ensure the technology would have minimal impact to operators' core networks already in place. Furthermore, it was designed to provide access layer mobility to circuit services (CS) and packet services (PS) to make WiFi and IP a seamless extension of the operator's network. The UMA network controller is integrated as an element of the operator's existing radio access network (RAN), which provides several important benefits.

First, as an element of the RAN, the UMA system is linked at the physical layer with other RAN elements. Handing a voice or data session between disparate network elements is a difficult proposition; however, by integrating directly with the mobile access network, UMA is ideally situated to manage the millisecond timing to ensure "seamless mobility."

Second, integration into the RAN ensures that the switching layer of the mobile network is not impacted by the addition of UMA network elements. The operator's existing MSC and GSN infrastructure can immediately take advantage of IP and WiFi with no modification or disruption, thus ensuring immediate support for all MSC/GSN layer services, such as lawful intercept, emergency services, directory assistance, billing, and customer care (Figure 1).

Finally, as an element of the access network, UMA ensures support for all value-added network-level services. To maintain a consistent experience, operators must provide support for all new and future value-added applications, like SMS, MMS, ring tones, PoC, and other advanced applications. As an access layer technology, UMA ensures that today's applications, as well as tomorrow's SIP-based services, move seamlessly between networks.

Consistent Mobile Experience

From the beginning, the designers of the UMA specification set out to define a solution to break the cost and performance barriers that have to date limited fixed-to-mobile substitution in the home or office.

To increase mobile service usage indoors, UMA is designed to ensure users have a consistent mobile experience, that operators have minimal network disruption, and that the technology fits within an operator's long range network plans. Subscribers expect mobile services to work the same in GSM as in WiFi or any other access technology. The user interface on the handset must be common and consistent (Figure 2). If history has taught one thing, it's that the consumers in the mass market do not want to do anything "special" to receive a service.

To meet these goals, UMA is implemented as an embedded driver within the mobile platform. It is not a client application that is loaded onto the phone. Rather, as the subscriber moves from the GSM network to the UMA environment, the association and handover occurs in the background and the subscriber does not experience any difference in quality, capabilities, and most importantly, user interface.

IMS and UMA

IMS, (define - news - alert) originally conceived by mobile operators as a common application platform to speed time to market for new data applications, has suddenly become the unifying force in the telecom industry. As new approaches to mobile/WiFi convergence are developed, service providers need to evaluate technologies against user experience, ease of deployment, and the ability to align with future business and technology goals. The ability for UMA to provide seamless mobility for IMS applications ensures that the UMA and IMS technologies will be deployed side by side for years to come.

Fundamentally, UMA and IMS operate at different levels within the network (Figure 3). UMA is an access level technology, completely agnostic to the packet services or circuit services delivered over it. UMA provides for the convergence of mobile services over WiFi access networks. IMS, on the other hand, is access layer agnostic, meaning that with a secure and reliable IP connection, IMS applications can be delivered over GSM, UMTS, WiFi, UMA, or even DSL/cable.

UMA provides for circuit services (CS) and/or packet services (PS) to be seamlessly passed between the GSM and WiFi networks, which is accomplished through the use of a single UMA network controller. For many operators, the UMA approach to MWC meets the requirements for a dual-mode service.

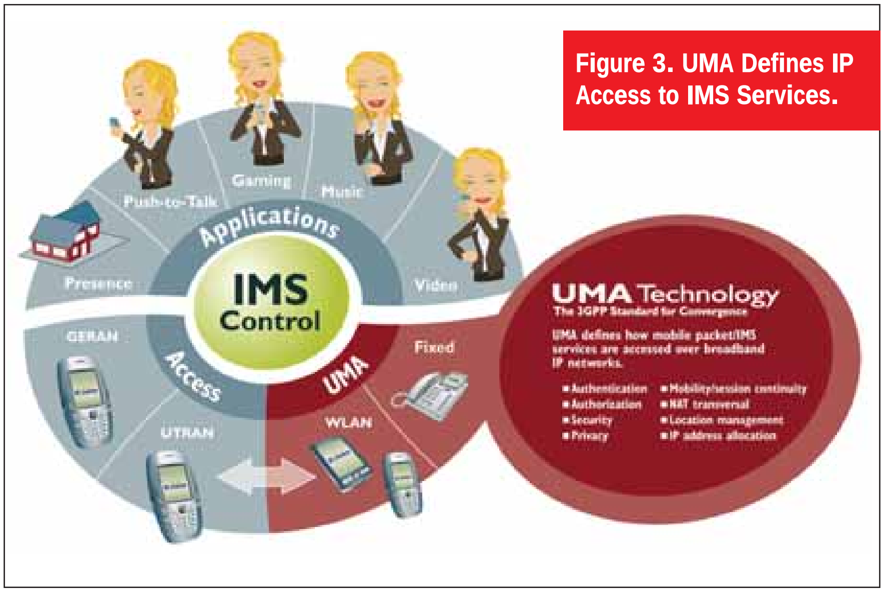


Figure 3. UMA Defines IP Access to IMS Services.

What About SIP?

One of the most common comments about UMA is that it “doesn’t support SIP.” SIP, (define - news - alert) of course, is the preferred protocol for initiating application and service sessions within the IMS domain. As mentioned, UMA is agnostic to the application/service layer. Conversely, applications and application protocols (like SIP) are essentially agnostic to the underlying access network.

As an access technology, UMA is more akin to the GSM or UMTS radio technologies than to SIP or H.323 signaling/application protocols. The industry understands that SIP runs on top of the UMTS network, yet SIP running over UMA can sometimes seem incomprehensible.

In fact, many operators today are deploying UMA-enabled devices with SIP applications. UMA provides the mobility for the SIP services between the WiFi and GSM networks, and SIP delivers new applications for the consumers.

UMA and VCC

While SIP does not specifically address mobility between radio networks, there is a work item started in the 3GPP to address how mobile/WiFi convergence may be addressed without UMA. The IMS approach to providing MWC is actually a new work item in the 3GPP known as Voice Call Continuity (VCC).

VCC is working to define how a SIP/VoIP packet service can be transitioned to the existing GSM circuit services network. Beyond the obvious technical hurdles of transitioning a call context from a packet network to a circuit network, VCC relies on the operator to invest in additional network elements to accomplish the vision.

The first investment is in the VCC network element itself. Touted by nearly every IMS vendor, VCC is a transitional gateway between the existing CS network and the PS VoIP core. Because it is still quite early in the definition phase, it is unclear if — or how — the VCC network element will support non-voice packet services that rely on session continuity will be supported.

Secondly, a new network access element is required to provide an access interface between the public internet and the operator’s IP core network. Based on the Inter-worked Wireless LAN (I-WLAN) specification, this network element was originally defined for laptops to gain access to mobile data services.

Lastly, VCC is reliant on the operator having invested in a new IMS voice switching infrastructure, which mobile operators today are not doing. Currently, the investment is in new Release 4 “soft” MSC switches from which the operators are looking to achieve significant operational savings from over the coming years.

Interestingly, the only operators investing in packet voice switching infrastructures today are the fixed line operators. Most fixed operators have parallel VoIP core investments underway to provide fixed line VoIP services in response to the competitive threat from Skype and others.

The Battle for the Building

Previous to fixed/mobile convergence, subscriber ownership was clear. Mobile minutes of use were billed to the mobile operator, regardless of the location of the call, and calls on the fixed network were billed to the fixed operator. Yet in a converged world, services and service providers are brought together. The fundamental question now is if a mobile phone is receiving service over the fixed line broadband network, which operator owns the subscriber and bills for the minutes — the fixed operator or the mobile operator?

If the fixed line operator wishes to own the minutes of use when the sub-

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Web Site	Alexa Site Rank
TMCnet.com	**1,348**
eWeek.com	1,948
Computerworld	2,657
InfoWorld	3,807
Network World	5,709
Light Reading	13,113
Pulver.com	39,793
Wireless Week	40,024
Telephony Online	43,927
Destination CRM	52,688
VoIP News	60,688
Telecomweb	165,585
America's Network	190,704
Telephony World	202,053
Call Center Magazine	255,735
CommWeb	260,840
Communications News	636,037
Wireless Review	844,641

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Web Site	Alexa Site Rank
TMCnet.com	**1,348**
Smart Money	2,950
Fast Company	3,687
Red Herring	3,991
Inc. Magazine	5,848
Barron's Online	6,545
Technology Review	7,298
Weekly Standard	8,259
CIO Magazine	12,014
BtoB Online	20,539
Fortune Magazine	76,528
Worth Magazine Online	242,558

TMCnet.com Traffic vs. Fortune 500 Company Web Sites

Web Site	Alexa Site Rank
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GE	3,525
General Motors	3,924
Coca-Cola	5,008
State Farm Insurance	5,890
DuPont	16,464
Kroger	22,247
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Exxon Mobil	30,888
Chevron Texaco	34,608
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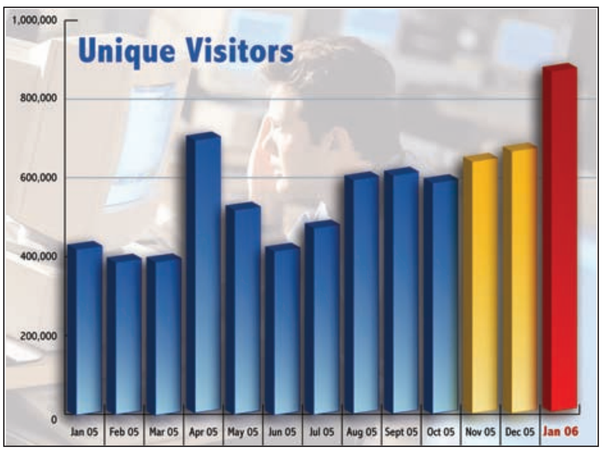

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eWeek.com	1,942	Smart Money	2,992	GE	3,209
Computerworld	2,940	Fast Company	3,748	Ford	3,329
InfoWorld	3,980	Red Herring	4,370	General Motors	3,797
Network World	6,195	Inc. Magazine	6,008	Coca-Cola	5,718
Light Reading	12,471	Barron's Online	6,551	State Farm Insurance	6,513
Wireless Week	42,137	Technology Review	7,382	DuPont	17,070
Pulver.com	43,326	Weekly Standard	8,342	Kroger	22,656
Telephony Online	52,860	CIO Magazine	12,663	AIG	29,404
Destination CRM	56,033	Fortune Magazine	13,063	Chevron Texaco	34,412
VoIP News	68,122	BtoB Online	25,004	Exxon Mobil	34,607
Telecomweb	191,198	Worth Magazine Online	217,538	Fannie Mae	39,772
Telephony World	229,201				
Call Center Magazine	234,504				
CommWeb	243,661				
America's Network	251,349				
Wireless Review	604,039				
Communications News	673,705				

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scriber is indoors, VCC is the technology approach of choice. VCC enables the fixed operator to leverage the VoIP switching infrastructure to deliver services when the subscriber is indoors and to transition the packet service to the GSM MSC when the subscriber moves outside range of the WiFi network. Conversely, UMA maintains subscriber ownership with the mobile operator when the mobile device is indoors.

As voice services and revenues continue their relentless migration to mobile devices, this becomes a question of investment in continuing to maintain voice services on the fixed network, or accelerating investment in the migration of voice services to the mobile network.

UMA for IMS Packet Voice Services

As mobile operators invest in IMS for the delivery of value-added packet services, UMA natively provides mobility for those services. Streaming audio, video conferencing, Push-To–Talk, and all other packet services are supported with seamless mobility today in UMA.

The next step comes when mobile operators begin the investment in IMS/SIP for packet voice services. The timing of this investment is predicated on the ability to provide a common user experience between networks. The consumer does not want to receive an "enhanced" SIP-based user interface when on the WiFi network and transition to the traditional circuit services interface when on GSM.

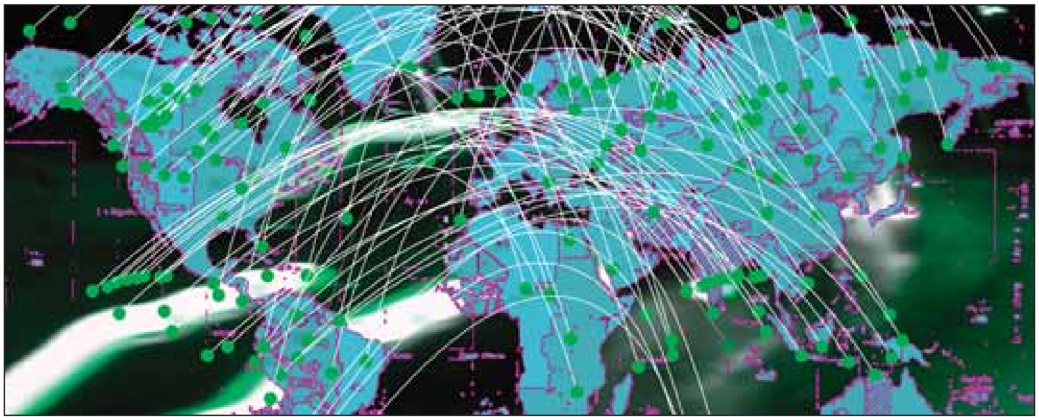
When the operator can provide a common user interface and experience, regardless of network, then investments will be made in new technologies. For the mobile operator, this means the ability to deliver a SIP-based enhanced user interface on the RAN, as well as on WiFi. As the operator invests in HSDPA/HSUPA and other evolutionary RAN technologies that enable the delivery of packet voice services, then a common user interface and user experience will be possible between the RAN and WiFi/IP networks.

Ultimately, it is UMA that will provide seamless mobility between the RAN and WiFi for these packet voice services. IT

Steve Shaw is director of marketing for Kineto Wireless. For more information, please visit the company online at http://www.kineto.com. (news - alert)

CLECs Embracing VoIP

By Joel Fisher



Remember how exciting the communications world was back in the 1990s? The Internet was going to revolutionize the way people communicate, 3G wireless networks were going to turn mobile telephones into personal video players, and CLECs were going to overtake the big incumbents who would stand by paralyzed, unable to take action. Then the bubble burst. Fast forward to 2006.

Now that the communications sector has seen double digit growth for multiple years, things are heating up once again. VoIP is on a path (although slower than we all originally projected) to overtake the PSTN, the investment in wireless networks is leading to the adoption of mobile broadband applications, and the CLECs are not only back from the brink of extinction, but are rapidly becoming a force that the large incumbents must compete against, thanks to a combination of attitude and technology.

"It is no longer enough to think outside the box, you need to tear it apart!" A paraphrase from the COO of Empire One Telecom, a NY-based CLEC, helps illustrate the kind of attitude CLECs must adopt. This positive attitude is reminiscent of the entrepreneurial efforts in the 1990s.

Although, in the early days of VoIP, there were many good ideas about the next 'killer app' that could be offered thanks to IP technology, the willingness for business users to take on the role of early adopter was lacking. It became clear that VoIP needed to go mainstream and prove itself with standard voice centric PSTN replacement applications before exciting new killer apps could become a driving force. Fortunately, with the aggressive attitude of the CLECs coinciding with the mainstream approval of VoIP and the pent-up frustrations customers have with incumbent carriers, the CLECs can now make a run at expanding their businesses and using new VoIP technology to succeed.

CLECs must offer three major service elements to their customers:

1. Rich feature set.
2. Security.
3. Voice quality.

Rich Feature Set

At the end of the day, communication is all about connecting people together. Whether it's a point to point voice call or a multiparty, multimedia collaboration session, service providers rely heavily on customer management and feature delivery. One VoIP switching vendor who has been successful helping CLECs make the jump to VoIP is Coppercom. According to Chuck Harris, Vice President of Marketing for Coppercom, "CLECs have aggressively been making a move from the UNE-P model to facilities based and using VoIP is a natural choice for them. It gives them flexibility in where they locate their equipment (since VoIP can be deployed in a hosted model) and allows them to deploy a differentiated set of services due to SIP's ability to easily integrate with third-party application platforms."

Security

One of the most attractive opportunities CLECs find from VoIP technology is the ability to serve customers regardless of location and regardless of who provides the broadband access. However, with this opportunity also come deployment challenges. What kind of firewall does the customer have? Is the customer's equipment behind a NAT? How do you ensure privacy of both signaling and media? How do you protect your core servers from denial of service attacks?

CLECs have found the answer to deployment challenges by deploying Session Border Controllers. SBCs are able to bridge signaling and media across VoIP elements to ensure successful call flows, protect core servers, and enable the secure delivery of new applications. The focus of the CLEC must be on how to offer new services and how to go to market faster than their competitors. Allowing the VoIP border services experts to solve the security and access challenges facing VoIP providers enables the CLEC to focus on new revenue generating applications.

Voice Quality

In the early days of VoIP, it was generally accepted that voice quality was far behind the toll quality voice we all have come to expect. There was really no effort to challenge the voice quality perception, but rather an effort to focus on the benefits of VoIP that would more than compensate for sub par voice quality.

However, we now have an entirely new 'tear apart the box' understanding of what service providers can do regarding voice quality. For example, you can use "Hi-band" codecs and 'treat' the media to offer voice quality that is superior to the standard toll quality voice we use today. Whoever thought the killer app of VoIP would be superior voice quality!

In summary, VoIP allows CLECs to compete. They can move faster than the large incumbents and they are well suited to offer services at newer and higher levels than ever before. From exceptional customer service to innovative multimedia applications to better voice quality, CLECs are deploying VoIP and are trying to live up to the original promise of the Internet — to change the way people communicate. IT

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Speaking With Empire One Telecom's Paul Butler

Greg Galitzine, editorial director of *Internet Telephony*, interviewed Paul Butler, COO of Empire One Telecom. (news - alert) The results of their dialogue appear below.

GG: *Who is Empire One Telecom?*
PB: Empire One Telecom (EOT) is a small, ethnic-niched facility-based CLEC/IXC/Broadband Provider based in Brooklyn, NY, and covers about 22 states. The company was established in 1996 and survived the telecom bubble and is successfully launching a comeback from the dark days of telecom on its own. This little but profitable carrier is using all its resources to create competitive products and grow its existing business as well as generating new business. EOT offers local, long-distance, broadband, dial-up, voice-over-broadband, prepaid, and wholesale.

While most UNE-P CLECs were taking the "wait-and-see" approach to the ruling on unbundled switching, EOT was already deploying its facility model and mapping out its game plan. Empire never saw the value in UNE-P since most of its customers were residential and used dial-up service had no interested in unlimited packages, and actually found UNE-P not a cost effective solution for an overwhelming majority of its customers. EOT saw no light at the end of the tunnel for UNE-P and realized the facility advantages to owning a network and went to work on a better model. EOT has grown 30 percent in the last six months, and by Wall Street standards, is pretty profitable. EOT is reinvesting its earnings back into the network, marketing, and new products that will make it a more formidable and profitable carrier in the near future as the company expands its facility foot print and product set.

GG: *You have been quoted as saying, "It is no longer enough to think outside the box, you need to tear it apart!" What do you mean by that?*
PB: Well, to think out of the box is saying, 'let's try something different.' To me, tearing up the box is tearing up the model, stepping back, and looking at the shredded box and building it all over again and better.

You have to take what you've learned and what you have, then, think of what it is you want to do and put that box back together again. It better not take long or cost a lot and you need to fill it up quickly, "Build a better model based on knowledge and experience." I don't believe that every telecom executive out there is as creative as me because they lack the experience in all aspects of the business and fall short of the real understanding of the business.

Since EOT was originally founded in 1996, I have worked every aspect of the business and understand it all from customer service, regulatory, network engineering, marketing, and more. If you can't see what all the pieces of the box mean after you tear it up how do you put back the pieces and build a better box? When you build that new box it has to make you money right away or you end up back in the same hole CLECs were in with the "build it and they will come" ideology. What CLECs lack in financial muscle can be made up with marketing savvy, and that's a big piece of the box.

GG: *Several years ago, it was thought that CLECs were heading the way of the dinosaurs. Describe the turnaround of the CLEC market. To what do you attribute your ability to pull through the "lean years" and survive the telecom meltdown?*
PB: To be honest, we had a great model but bankers never understood the business and looked for the quick buck with big fees. So money went to the big names, not those who exhibited fundamentals, and so we saw the Dinosaurs. The Big Dinosaurs needed a lot to survive and there just wasn't enough for all. But, just like in the Dinosaur days the smaller survived, as did we. What has enabled EOT to survive is the ability to move quickly, lower costs, retool on the cheap, and spend marketing dollars where we got the best return, acquisition costs under $20.

EOT has taken advantage of everything we could find. Better digs, tax breaks, lower equipment costs, and most of all product development and marketing, and execution. We didn't blow $5 Billion like a few companies out there, we made sure if we spent $50,000 it came right back three-fold! We didn't have the private jets or leased BMWs or blown-out networks to weigh us down. We ran lean and looked ahead, just like we are right now. We know what's coming and we are building, getting ready to fight and win more business than ever before.

GG: *Has VoIP played any role in your success?*
PB: Yes, we quickly deployed VoIP with Coppercom and it was a significant part of our growth in the last six months and continues to grow every month, and will be a significant piece of our product set in the future. You won't survive without it.

GG: *What role do session border controllers play in the Empire One network?*
PB: They hide your network and are the first line of defense; they help find paths to customers behind firewalls and manage NAT-ing and provide some level of port management. We work with Ditech and it's nice to talk with them about ideas and hear, "yeah that would be great;" suddenly, you're on the phone with product development and the next thing you know you have what you asked for. A vendor who can see value in what you ask for and how and what it means to other customers of future customers is invaluable.

The SBC plays a small role today but a smart one will add applications and value for its customers, and they are from what I am hearing, and that's good. I see SBC becoming more valuable to IP networks and service providers.

GG: *Do you have any further thoughts you'd like to add for any CLECs reading this article?*
PB: You either understand this business or you better have big bucks behind you. Lee Iacocca put it very well when he said, "Lead, follow or get out of the way." If you think spending 187 million to get 170 million is revenue (Vonage) is great, think again. You could have saved 17 million and given it away.

Think of the CLECs who built networks, because it was how Wall Street valued them one day, and the next day said networks have no value, where is the revenue? Don't get caught up in the "build and burn," you never know where your next meal is coming from. Build and Market smart, learn about what you have and how to apply what you have to make money, and you won't have to find a banker — they'll find you! Stick to business fundamentals of making money. You don't have to be everything tomorrow, you have time if you do it right. IT

Meeting Business Communications Challenges Head-On

It has been 40 years since Intel (quote - news - alert) co-founder Gordon Moore made his now famous observation, which appeared in the April 19, 1965 issue of Electronics magazine and states that innovations in technology would allow a doubling of the number of transistors in a given space every year and that the speed of those transistors would increase. About 10 years later, Moore adjusted the rate to every two years to account for the growing complexity of chips.

Moore's prediction, now popularly known as Moore's Law, had some startling implications at the time, predicting that computing technology would increase in value at the same time it would actually decrease in cost.

There is no better example of Moore's Law in practice than the collaborative, real-time communications software market. The development of Web and video conferencing technologies has paralleled Moore's prediction and, today, communicating instantly using video and audio, and other applications is as easy as just a click of the mouse. What, approximately 10 years ago, could be accomplished in a hardware-based video conferencing system, costing between $20,000 and $50,000, has now made it to the desktop. With equal, if not better video and sound quality, boardroom-quality video and Web conferences can now be conducted utilizing a standard desktop PC and any Web camera.

Video conferencing has an illustrious history, making its debut just before Moore's prediction was made public, at the 1964 World's Fair in New York. At the time, it was no more reasonable to launch a space shuttle into orbit than to use video for any of the applications its being used for today, including telemedicine, live distance learning, business meetings, sales presentations, job interviews, and the like. The first attempts at using traditional telephony networks to transmit slow-scan video failed mainly because of the lack of efficient video compression capabilities that produced a very poor picture quality. By the 1980s, ISDN digital transmission networks became available to the public. These networks assured a minimum bandwidth (generally 128 kilobits/sec) for compressed video transmission. In some instances, multiple ISDN lines were used, often in groups of four and eight, to achieve acceptable video performance. Throughout the 1990s, video teleconference systems continued to rapidly evolve due to standards-based technology advancements. The decade also saw the emergence of Internet Protocol (IP)-based video conferencing and more efficient video compression technologies were developed that permitted PC-based, desktop video conferencing. Free services, such as Microsoft NetMeeting, MSN Messenger, and Yahoo! messenger, brought the general public and consumer market inexpensive, yet low-quality video conferencing.

The shift to "on-demand" desktop Web and video conferencing didn't happen overnight, but accelerated advancement of IP networks and other technological innovations have all helped to drive collaborative applications to desktops around the world.

Web Conferencing Software Outshines Overpriced Hardware-based Video Conferencing

Perhaps most important to the evolution and growth of the popularity of Web conferencing is the rapid advancement and development of the Pentium processor. New generations of processors are able to process and transmit data, voice, audio, and video at amazing speeds. High-speed Internet, standards-based technologies, and processor speeds have combined to bring real-time communication and

By Lori Scribner

collaborative computing to desktops in schools, business, government, and the military.

While there may always be a market for high-end hardware-based video conferencing systems, the cost is out of the question for most mid-market organizations — which limits its accessibility and is definitely a barrier to ubiquity. Traditional boardroom conferencing systems generally require physical leased lines and two hard wired fixed route endpoints cost upwards of $50,000. For installations with more than two endpoints, the routes and endpoints are pre-determined and conferencing connections are established using video-specific routers called Multiple Channel Units (MCUs) or Video Multiplexers (MUXs), which often double or triple the cost of a video installation.

Still, there are many large corporations, educational, and government and military facilities where the scale, staff, and costs can sustain an expensive hardware-based, fixed route solution. However, video capabilities are limited to pre-determined endpoints and there is no collaboration component without additional cost.

The ability for video and Web conferencing to be accomplished via software applications, on standard PCs and with browsers on standard Internet connections has made the widespread adoption and proliferation of Web and video conferencing possible. It has reached desktops in the smallest of organizations and the cost savings benefits are starting to make sense to even the largest Global 500 companies. Standard applications for Web and video conferencing are familiar to most: distance learning, sales and ad-hoc meetings and presentations, human resource interviews, company orientations and employee training, to name some.

Web conferencing has enabled companies to save money on business travel for personnel training, meetings, and has maintained the integrity of the meeting by adding a new dimension of interaction and collaborative abilities that are second only to being there in person. Browser-based video conferencing technology, and the end user experience, have caught up and, in many cases, have surpassed that of hardware-based video conferencing systems. A Pentium processor running at 1.0 GHz or better is far more powerful than legacy video hardware codecs. Although both types of systems use standard algorithms for encoding/decoding, there are major variations in implementation. Software-based IP video can be browser-based, dramatically increasing the reach and applications set for a video conferencing investment. Software can also connect via port 80, and maintain compatibility with industry standard Web services, so as to maximize connections among employees and customers behind firewalls. Legacy hardware solutions cannot do this without MCUs and MUXs. Most importantly, high-quality software can take advantage of symmetric multi-processors, multithreading, Intel MMX optimizations, and especially good techniques in synchronizing VoIP, video, desktop sharing and collaboration. Which brings full-screen, high-quality video plus desktop sharing into a single, fluid and productive conferencing experience — right to and from the user's desktop.

In short, Moore's Law has caught up with video conferencing hardware endpoints. Well designed software on a Pentium desktop now matches previous generation full-screen, full-motion video conferencing. Moreover, it brings expanded reach to anyone with a PC and an Internet connection, connectivity, and a fully collaborative desktop sharing experience far beyond what legacy video systems can provide.

Deploying Web Conferencing in Complex Networks and Secure Environments

Managed service providers provide organizations of all sizes from nearly all industries Web and video conferencing services. However, services can be expensive, because of per minute prices and inevitable overcharges. This type of pricing model makes it difficult for any size organization to manage Web conferencing costs, not to mention the reliability of the application due to bandwidth constraints, and the security of the conference session and its contents. Now that Web and video conferencing can run on any standard PC server, many organizations are finding out firsthand the benefits of owning the Web and video conferencing application. For example, Hawaii Pacific University (HPU), which has 10 campuses on the island of Oahu, including eight military satellite campuses, decided to purchase Web and video conferencing software for use throughout the campus. HPU found that the application was easy to install and use and it didn't have to assign any special IT resources to manage the application on the back end.

"While the simplicity of the Web conferencing application was critical, so was its affordability," said Justin Itoh, CIO at Hawaii Pacific University. "We are saving about one half of what we paid previously for just remote control from a leading service provider. Not only are we saving money, but because we own the software, we can offer it to other departments as needed."

HPU has found that Web and video conferencing comes in handy for job interviews in the Human Resources Department, which may be interviewing potential employees on another island, or three time zones away. It is also being used for live distance learning, administration staff meetings, IT helpdesk, and enabling teacher and administrators from all campuses to gather virtually.

Without a tremendous strain on IT assets — already stretched to the maximum in many organizations — Web and video conferencing software can provide a fast return on investment by keeping key personnel in the office to focus on their daily responsibilities.

And because it is so versatile, Web and video conferencing software can find a home in nearly every department in an organization and actually become a critical component in the continuity of business processes — especially to organizations in regions that experience harsh seasonal weather conditions that make it difficult, or even impossible, to make it to the office.

Purchasing on-premise Web conferencing software has become affordable for every organization that can afford a standard PC server.

Organizations that place a high priority on security, such as financial, legal, government agencies, and the military can also benefit from the collaborative, real-time communications capabilities that Web and video conferencing offers. Because the installation of the software is behind the firewall vital information being presented during the conference session is secured. However, not all Web conferencing applications are created equal. Some are designed to work seamlessly, reliably, and to scale easily in a secure network environment. Other approaches to real-time communications utilize T.120, peer-to-peer connections, and UDP broadcasting, all of which can be difficult to deploy in a distributed enterprise. In some traditional communication software solutions, especially those that provide voice and video, many attempt to connect endpoints directly between peer computers using a combination of protocols. It is common for peer-to-peer protocols to utilize TCP and UDP transports to communicate as quickly as possible.

Many, if not most, video and audio real-time products utilize UDP. This is especially true of educational applications originally designed to operate between desktops on the same LAN. These products require opening entire ranges of ports, which is often a violation of security policies, or the application simply cannot route between multiple offices or over the public Internet. However, Web conferencing solutions that utilize only TCP/IP connections over Web services standard ports (80, 443) within their routing schema solve these problems. While TCP adds additional latency

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versus a UDP classroom broadcasting approach, for practical purposes the former successfully connects users over the Internet and between offices while the latter does not.

However, as network address translation (NAT) devices, proxies, and firewalls have become more pervasive, solutions that use these scenarios tend to be unreliable in many corporate networks. Organizations that employ stringent security policies can still take advantage of the latest Web and video conferencing software. As with any application, some software developers take security more seriously than others. There are Web conferencing solutions that secure conference sessions from end-to-end. Those that route audio and video over separate ports and separate broadcast or TCP/IP connections make security and encryption problematic or nearly impossible to implement. However, those that transport data, video, and audio over the same TCP/IP connection can provide end-to-end security using industry standards.

In Web conferencing, it is standard practice to utilize SSL and TLS for securing conferencing sessions. IT administrators can employ the routing and server access restrictions, plus account names and passwords for conferencing rights. Meeting hosts can also add conference room passwords. The IT administrator can configure the SSL and TLS features to use RC4, DES, AES, and even RSA depending upon their security requirements. This provides the highest degree of commercial secure encryption. In addition, well-designed software will also allow customers to use their own security certificate, and for sensitive installations, even allow the customer to use their own Public Key Infrastructure (PKI). By using SSL and TLS, customers can leverage existing investments in their current Certificate Authorities (CA) and PKI. In addition, organizations can confidently operate their Web and video conferencing applications without making any exceptions to their internal security policies.

Conclusion

No longer relegated to the largest organizations and corporations that have immense IT spending budgets, the Web and video conferencing software industry has flourished because of PC processor advancement, technology standards, and widespread access to high-speed Internet connections. Purchasing on-premise Web conferencing software has become affordable for every organization that can afford a standard PC server. And it can truly benefit any entity by saving money on business travel, decreasing time spent out of the office, and increasing the interaction between conference attendees who can simply click to join a meeting from anywhere in the world.

Even the most secure, locked down enterprise that employs firewalls, NATs, and proxies can easily deploy and utilize Web and video conferencing software. The IT administrator needs only to find a suitable solution that utilizes a pure TCP/IP approach that only utilizes one secured connection to transport data, voice, video, and audio.

Web conferencing software is on par with expensive video systems in terms of the visual experience, especially with equivalent bandwidth availability, but it definitely surpasses legacy hardware systems in terms of affordability, dramatically expanded reach, ease of deployment, security and enhanced multiparty video and collaboration capabilities. IT

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Contextual Collaboration:

The Next Step in Effective Communications

Conferencing and collaboration have been ‘buzz’ words in the business world for quite some time. Although the definition of these terms varies depending on who you ask, the core concept is that conferencing and collaboration solutions are tools to enable geographically dispersed individuals to communicate and work effectively together. However, the majority of solutions in use today force participants to step outside of their normal work environments, and adopt a new environment, in order to conduct a conference or collaborate on a document. In other words, the current video conferencing, Web conferencing, data collaboration, and application sharing solutions require users to open a new application/tool or utilize a piece of equipment that is not part of their normal work processes. Contextual collaboration is the next generation of these types of solutions.

Contextual collaboration enables users to conference and collaborate in real time within the context of their existing work environments. Contextual collaboration solutions are embedded within the applications that people use everyday in business — leveraging the power of conferencing and collaboration from inside the user’s most critical business tools. In addition, contextual collaboration takes advantage of presence technology to simulate the in-person encounter — when individuals can ‘see’ when a person is online and available for an ad-hoc conversation or work session with the power of all of their business tools facilitating the process.

The Present State of Conferencing and Collaboration

The use of real-time Web conferencing and collaboration within the enterprise is on the rise. As a result of globalization and an increase in mobile workforces, enterprises must maintain productivity while managing communications between dispersed, national and international teams. The cycle of communication has to be shortened and decisions must be made quickly in the face of rising competition.

It is apparent that real-time Web conferencing and collaboration that incorporates audio, video, and application sharing in a single collaborative platform addresses the communication needs of today’s enterprise. It replicates the personal nature of face-to-face meetings and engenders the kind of creative brainstorming that drives success. Helping to speed the decision making process and enhance productivity, today’s conferencing and collaboration solutions are empowering employees, customers, and vendors to effectively and equally contribute to the dialogue and streamline business processes. However, these traditional solutions fall short because they do not work seamlessly with other business applications. This incongruence is disruptive and can make conferencing and collaboration technology awkward and often painful to implement and utilize in the day-to-day office setting.

As enterprises realize that real-time conferencing and collaboration is no longer considered a luxury, but a necessity, developers are embarking on the next generation of tools that are exponentially more effective than their standalone application predecessors.

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By Philippe Szwarc

Enter Contextual Collaboration

Contextual Collaboration is a new approach to conferencing that combines presence technology, real-time communication, and resource sharing to make online meetings as simple, natural and productive as face-to-face encounters without abandoning the business tools and applications that are a critical part of daily operations. By either combining all relevant applications (i.e., Word, IM, calendars, conferencing software) into one easy-to-use interface, or embedding the collaboration and conferencing tools into those applications, contextual collaboration allows corporations and individual users to launch a meeting, seminar or training session at any time from any document and collaborate in real time.

The goal of contextual collaboration is to empower users to instantly share any resources at their disposal without forcing them to leave their core application or tool to launch a new one for the purpose of sharing it with others. If the collaborative component is a built-in feature of the initial application, it is much more intuitive and simple to use. It becomes an embedded feature from a drop-down menu or a benefit that is only one click away. Combined with presence technology, employees can literally 'tap' a colleague on the shoulder for a quick look at a document, whether they are in the next cubicle or on the next continent.

With contextual collaboration, users do not need to leave an existing productivity tool for the purpose of sharing, sending, or collaborating with remote associates. These are now capabilities that are embedded in the core productivity tool or application. The impact of contextual collaboration solutions is not just about saving time or brainpower for the individual user to learn a new application, but the repercussions for the enterprise on the whole. Supporting, implementing and training individuals on monolithic collaboration tools adds to the complexity and costs of enterprise technology. Although individuals may

Traditional solutions fall short because they do not work seamlessly with other business applications.

have some productivity gains with standalone collaboration solutions, they may be offset by the toll they can take on the average IT department. Senior management needs to be concerned with enterprise-wide efficiencies and business process management beyond the individual.

The Elements of Contextual Collaboration

There are a number of basic components that are typically present in contextual collaboration.

These include:

- Presence Technology
- Real-time Communication
- Resource or Data Sharing

Presence technology is a key component of business communications today and needs to be incorporated in contextual collaboration solutions. This technology empowers users to be 'aware' when a colleague is available. Just as employees in a close office environment are equipped with the knowledge and ability to 'check the pulse' of a colleague on a moment's notice, with presence technology, so are employees that work in offices an ocean apart. When presence technology is embedded in the application infrastructure, it enables individuals to see who else is online, who else is using a particular application, and who else is viewing a particular document — engendering ad-hoc communication and power information sharing.

Real-time communication is an extension of the awareness established with presence technology. This can take the form of textual communication, voice communication, video communication, or a combination of all three. The ability to 'talk' with colleagues is infinitely more effective than sending materials or documents back and forth. It speeds up work flow and decision making and enhances personal relationships. When video is added to the mix, the ability to see one another adds another layer of information and intimacy. The expressions, body language, and subtle nuances that individuals project in personal encounters cannot be translated into other modes of communication.

Resource or data sharing is a critical element as well. By providing the capability to show, share, and change documents, files and applications, productive workgroups can be established beyond traditional geographic boundaries. The goal of contextual collaboration is to further make online conferencing and collaboration as simple and intuitive as it is to work with people in the same room, while enabling that capacity between people anywhere in the world. The ability to share knowledge and resources remotely is at the core of these initiatives.

Think Outside the Office

Corporations and individual users using contextual collaboration can launch a meeting, seminar, or training session at any time from any document, and collaborate in real time. Users can collaborate from a personal computer or laptop, PDA, mobile phone, or with a conventional or IP telephone. Participants gain the freedom to join a meeting or session from wherever they are available: a hotel, coffee shop, airport lounge, or even the office. As business is conducted in all types of environments, contextual collaboration is not limited to the desktop.

Enabling remote or mobile employees to participate in and drive the discourse is an absolute necessity in today's business environment. Communication with colleagues and customers should be accessible 24/7, from anywhere in the world, with any resources necessary at the fingertips of any executive. Contextual collaboration should encompass not only the applications in one's business environment, but its devices as well. There are no technical obstacles today that would prevent enabling a traveling employee to launch a document on his PDA, see his supervisor online, send him an IM asking his opinion, initiate a conference call with a customer, and share the document that they both annotated to secure an immediate approval or electronic signature on a contract while waiting in line for coffee from a street vendor.

Benefiting the Individual and the Enterprise

Contextual collaboration provides employees with the most up-to-date knowledge and tools they need, when they need it and where they need it. It dissolves traditional communication barriers, speeds response time, and significantly enhances productivity for the employee as well as the enterprise.

The transition to contextual collaboration will be transparent and seamless to users because they have nothing new to "learn." Ease of use is the essence of this technology because users are already familiar with the application or tools they are using. The collaboration is achieved by choosing an option, tab, or icon located within a known application interface.

Enterprise-level benefits are also significant. By removing the obstacles to conferencing and collaboration deployment, IT departments are liberated from the financial and human resources normally dedicated to deploying, training, and supporting users on these standalone applications or systems. By embedding these capabilities into existing business tools, they become part of the natural fabric of the work environment — encouraging usage and eliminating obstacles. IT

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Take 5: The Case for Conferencing

The case for investing in enterprise-wide conferencing and collaboration has never been stronger. The track record of voice, Web, and video conferencing in reducing costs and increasing productivity has earned them recognition as “birthright” workplace tools — tools that are universally useful and should be deployed to almost everyone in the company, similar to e-mail and the telephone.

Most businesses are already using some combination of teleconferencing, Web conferencing, and video conferencing. As currently delivered, however, these applications impose economic and technical limitations that prevent companies from deploying them more widely.

High-performance companies are asking how to put conferencing and collaboration tools in the hands of everyone without breaking the bank or the network. They can achieve this goal by considering five factors: licensing model, breadth of functionality, scalability, integration capability, and deployment options.

Licensing Model

There are two basic license models:

- Usage-based licensing — in which you pay only for what you use, but the more you use, the more you pay.
- Fixed price/unlimited usage licensing — which treats conferencing as a fixed price utility similar to e-mail.

Virtually all conferencing offerings today use some form of usage-based pricing for at least one component of their offering. The most common forms include (1) simple per-minute pricing for the combined voice, Web, and video event and (2) per-minute audio pricing combined with either named user or concurrent user pricing for the Web and video components. In addition, concurrent user pricing itself usually carries a stiff overage penalty for usage that exceeds the licensed number simultaneous participants.

Fixed price/unlimited usage license models have, until recently, been limited to on-site, hardware-based systems. They are now being offered by select voice and Web conferencing vendors under plans similar to local and domestic long-distance telephone services or enterprise software site licenses.

Licensing models have a strong impact on the use of conferencing. Simply put, the adoption of conferencing within the enterprise is hindered by usage-based models and is helped by fixed price/unlimited usage plans.

Questions to consider:

- Which licensing model makes the most sense for me?
- If I unexpectedly exceed my license, what penalties or charges I will incur?
- Will the vendor allow me to convert from a usage-based plan to a fixed price/unlimited use plan?

Breadth Of Functionality

Businesses will find three major approaches to conferencing functionality, each with its advantages and disadvantages:

- Individual point products for voice, Web, and video conferencing — This model permits organizations with the time, money, and expertise to assemble best-of-breed products in each category; however, this approach requires a good deal of IT support, user training, and may cause confusion if the various products include similar functions.
- Loosely bundled “integrated solutions” — An example of this approach would be a Web conferencing vendor who bundles third-party voice and video conferencing services and “integrates” them just enough to allow users to

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By Neil Lieberman

schedule voice, Web, and video events at the same time and receive a single bill for the multiple elements. While the components may share a similar interface, this approach is more cosmetic than substantive, lacking the critical architectural and data level integration.

• True integrated conferencing solutions — This model offers seamless operation between the voice, Web, and video components. The integrated nature of these offerings provides a broader range of functionality as well. For example, these products typically offer advanced eLearning, Web seminar, and on-demand recording capabilities, in addition to basic voice, Web, and video conferencing.

Companies seeking to encourage conferencing will benefit from the broader functionality of true integrated products. These products can meet the needs of all of a company's business groups, allowing the company to standardize on a single common product. This simplifies life for end users, who have just one product to learn, and of the IT teams, who have just one product to manage and support.

Questions to consider:

- What business functionality do I need?
- Can I find this functionality in a single product or must I build my own solution from multiple products?

Scalability

The more a company encourages conferencing, the more important scalability becomes.

There are two aspects to scalability: the number of people that can participate in an event, and the number of events that can occur simultaneously. The type of audio conferencing also has an impact: most conferencing products that scale well using traditional TDM audio have problems supporting the same load if participants use VoIP.

Questions to consider:

- What conferencing volumes do I expect, both average and peak usages?

Most conferencing products that scale well using traditional TDM audio have problems supporting the same load if participants use VoIP.

- Can the product demonstrate the ability to meet my volume needs with both TDM and VoIP participants?
- How does the product handle usage spikes, both planned (e.g., an all-hands meeting each quarter) and unplanned?

Integration Capability

The ability to integrate with business applications, IT infrastructure, security frameworks, telecom networks, and other systems and processes offers usability, security, administration, and cost benefits. For example:

- Usability — Being able to schedule and enter a conference via the Outlook or Notes interface increases adoption and usage.
- Security — Integrating with a reverse proxy server increases the security of the conferencing application and eliminates the need for separate conferencing domains for internal and external participants.
- Administration — Managing user access via an existing LDAP directory greatly eases the workload on an IT staff.
- Cost Savings — Integrating with a company's PBX (define - news - alert) or IP Gateway can dramatically reduce conference calling costs.

A product's integration capability is a function of both the application itself and its deployment model. On the application side, some products have rich APIs designed to simplify linking with products from multiple vendors; others don't. True integrated products (described above) are simpler to integrate into infrastructure elements than multiple products from different vendors. On the deployment side, on-site products tend to offer simpler, richer integration capabilities than hosted solutions; and sound security policies argue against making sensitive data, such as directory servers, accessible to hosted services over the public Internet.

Questions to consider:

- How rich are the product's integration capabilities?
- Can the vendor put you in touch with companies who have done the type of integrations that we want to do?
- Which of my specific application and infrastructure products does the conferencing product integrate with?

Deployment Options

There are four major deployment options available today:

- *Hosted Service* — Generally chosen when the volume of conferencing is not well known, when most meetings are external, and if there is a lack of in-house IT skills to manage the solution.
- *On-site Implementation* — Generally chosen when the level of conferencing use is high, most meetings are internal, security is a key concern, integration with existing IT infrastructure tools and processes is important, and there are sufficient in-house IT resources to manage the solution.
- *Managed Service* — An on-site deployment that is managed on an ongoing basis by the conferencing vendor or third-party service provider.
- *Blended Deployment* — This hybrid, available from vendors who offer both hosted and on-site implementation, can be managed as a single integrated system. It provides the cost savings, security, and control of an on-site implementation with the rapid rollout, broad reach, and overflow/failover protection of a hosted service. This model is especially popular with mid-sized to large organizations due to the flexibility it provides.

Forcing businesses to choose between either a pure hosted or on-site deployment is unnecessarily limiting in today's business environment. Most companies need to work with both internal and external participants and few would accept a lower level of security if given a choice. To provide broad access to conferencing and collaboration capabilities, and to avoid buying multiple conferencing products to meet different requirements, companies are often best served by the blended deployment model.

Questions to consider:

- What are my needs from a volume, security, integration, and geographic perspective?
- What options does the vendor offer?
- How can the vendor help me if my needs change?

Summary

High-performing companies understand that broad use of conferencing and collaboration can increase innovation, shorten response time, strengthen customer and partner relationships, and improve process effectiveness. They have demonstrated that conferencing saves money and increases individual productivity. They have also shown that giving conferencing to everyone, like e-mail, is not just desirable; it is technically and economically feasible.

Mid-sized to large companies looking to use conferencing throughout their organizations can use the criteria above to guide their search. For many, the ideal solution will combine a fixed price/unlimited usage license model, broad functionality to serve as an enterprise-wide standard, scalability to support significant demand without requiring active resource management, integration with key business applications, IT and telecom infrastructures, and a blended deployment model. IT

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Communications in Context —

Moving Beyond the Lowest Common Denominator

Let me express my respect for the International Telecommunications Union (ITU) and what it has given the world in terms of quality voice communications — before I tell you that it has led us to a world where communication has become more and more unnatural.

Technology always comes with limits. Unfortunately, we have adopted the limitations of the lowest common denominator technologies and made them the basis for how we communicate. The good news is that with IP communications, we have the opportunity to create communication services that are tailored for their context, useful and natural for a given situation or activity.

Today, we are network-centric — I use my cell phone to call your landline; my PC to call your soft phone. For each, I need to know an arbitrary identifier (ok, a phone number) and do not know if you are "there." IP communications gives us the ability to combine the power of the network and endpoints to make communication natural in the context in which it occurs. Companies that implement these types of communications as part of their product/solution, with an understanding of these issues, can increase revenue, improve user experience, and help build solutions for the next generation of communications.

Context is not something we have traditionally thought about in the communications field, but in life it shapes all communication. When you talk to someone in a business meeting it is a different style of communicating compared to talking to your daughter's boyfriend (in one instance politeness is required; in the other it is appropriate to stare them down — you decide which is which). We spend many years educating children on the importance of communicating in context — don't talk in school unless you raise your hand; talk to your parents, unless your mom is on the phone. These lessons are learned and pushed down into our psyche to a level where they control where, what, and how we communicate. This natural expression in many ways conflicts with the world of technology.

The Internet gives us a "reach anyone at anytime" way of thinking. Unfortunately, we do not want the world to reach us most of the time (contrary to what spammers think). The early development of online communications has been focused on making it work. Now we need to combine the power of technology with the lessons our parents taught us. This changes both the experience for the user and the financial model for the industry.

In developing solutions for context-specific communications, you need to consider the context, the mode, and the control of the communication.

Context comprises three elements:

- Presence — focuses on the user and what they are currently doing: what device are they using, are they active, where are they, etc.
- Rules — created by the community owner as to how they want communication to occur in this instance.
- Preferences — the user should always have the ability to specify their preference within the rules set by the service provider.

Together, these elements can provide a logical construct to build many different services from a common service delivery platform.

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By Monty Sharma

When thinking of communications platforms, we often think of them as distinct elements — voice network, IM network, e-mail server, and others. In the next generation of services, these lines will blur as we use multi-modal communications, mixing voice and text. The best examples of this I have seen has been in game play, where people mix voice and text on the fly. In some cases, it is the longevity value of text; it remains on the screen for people to refer to later. Locations, values, and instructions are well suited to this. On the flip side, when an anonymous call is scheduled for five o'clock and one party is in a situation where they can not answer the call, their ability to send an IM, e-mail, or SMS to the service to change the time of the call has great value. The world has long imagined voice commands as an interface and day by day we see more of this in the real world. What this leads us to is an architecture where all modes of communication need to be integrated at the control level.

For communications companies, context impacts revenue models substantially: some calls are worth more than others. Mobile operators have long offered service for less than 10 cents per minute; fixed operators offer unlimited usage for $15. But in the right context a minute can be worth as much as a dollar.

The online dating industry, which has 40 million users in the U.S., delivers the opportunity to find a mate in a simple way through one of their databases. But, after finding a possible match by looking at their profile, there comes the time when you need to take the relationship to the next level and talk to the prospective love of your life. However, by this point in time, you may not know this person well enough to want to give up your phone number. In the age of Google and stalker horror stories, this is a valuable piece of personal data. To minimize the risk, an anonymous call costing from $.60–$1.00 per minute that hides your personal data is well worth the cost. Value comes from several things: maintaining your anonymity, reaching your possible mate without having to giving up your personal information, and the spontaneity and immediacy of connecting at the click of a button.

The opposite, in terms of dollars per minute, is seen in the online gaming industry, which sees players spending 20 hours a week in online worlds. To this audience, the value of the communication comes from reinforcing the immersive nature of the virtual world. Augmenting the game experience with voice may be worth from one to ten dollars per month with unlimited usage. Everything from "click to call" to customer service will establish price points that break our old network-centric business models.

Context also impacts users' experiences and their sense of what they are buying. In the "old days," the limit of context was Find Me/Follow Me functions: a complex set of IVR commands that would supposedly make our lives easier. Users become frustrated by having to constantly tell the system where they are and what they want to have happen. Now, with technologies like "presence" we know where and when a user is online and, by connecting to their calendar, we can generate a sense of what they are doing. Companies can then use this information to determine if the call from someone you are going to meet with next should ring while you are in a meeting.

In another context, new companies are emerging who take conference servers and wrap them with rules that allow online communities to create "Bar Stools." These bridges can be set with rules that allow two males and two females in, with the ability to spawn a new call bridge when needed. The experience for the user is the ability to have a small group conversation. The value for the site operator is the charge of $10 per hour.

The future of communications is centered on a merger of physical item and expansion of virtual intelligence. I would much rather use my cell phone than a land line; the cell phone offers a phone book that syncs with my PC, a call log, the ability to erase a digit when I miss a key. But is it a phone? I get e-mail on it, send IMs, and use it to browse the Web and the time spent on those often

Context is not something we have traditionally thought about in the communications field, but in life it shapes all communication.

exceeds the minutes I spend on a voice circuit.

The simple lesson is: devices will merge and add functionality as we move further and further away from the black phone with the rotary dial. The merger of physical devices provides us with more information to improve the user experience. Since we have the phone book and the calendar, decisions can be made; calls that are important go through, others are sent to voice mail. Services that reside in the network will negotiate with the end point on how communication will be handled. Combining presence and location information generates even more information to deliver the best user experience. These options provide higher value services to users and break the industry out of the per-minute thinking.

The power of communications in context will change the way the world interacts. The physical and logical separation of networks will end and we will mix control and data across them all. As the user steps away from the network they will be able to use communication options that are integrated into the context and deliver a more natural experience. The value of the experience will determine what the communication costs. At the end, you get what you want and you pay for what you need. IT

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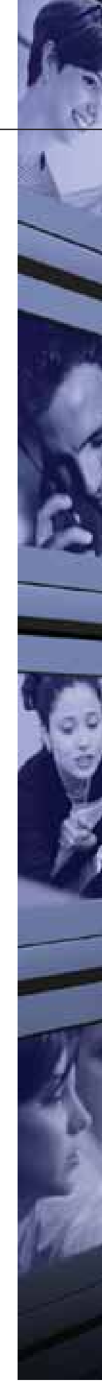

Successful IP Communications Starts With The Basics

A growing number of organizations are deploying Internet Protocol (IP)-based communications solutions to reduce their long-term operational expenses, improve efficiency, increase employee productivity, and differentiate themselves in the marketplace. Their objective is to take advantage of new integrated communications capabilities that combine speech, video, and e-mail (or text messaging) over an IP infrastructure to support any mode of communication anytime, anywhere.

While IP communications offers tremendous business benefits, converging voice, video, and data on a single network requires a structured approach for implementation, just like any other effort to integrate applications and systems. Moreover, organizations must pay careful attention to the implications that an IP deployment could have on operational processes, organizational structures, telecommunications carriers, and outsource (or out-task) partners.

If you're planning an IP deployment, steps taken early on will put your team on a path to success. Careful consideration during the planning cycle simplifies deployment, increases customer satisfaction during and after the migration, and makes for a smoother transition between implementation and post-implementation operations.

Approach IP communications systematically

A lifecycle-services approach offers a systematic way to deploy IP communications and mitigate risk. Not only does such an approach allow you to build a platform for achieving your short-term objectives, but you can also use it as the communications application evolves to meet your business requirements and enhance enterprise productivity. Through adherence to best practices and methodologies, the lifecycle-services model provides a clear roadmap for each critical step in the technology lifecycle: prepare, plan, design, implement, operate, and optimize.

A lifecycle approach puts enterprises on track to achieving the greatest success in the shortest time. It also helps lower deployment costs and enables a quicker return on investment.

Within the lifecycle-services model, infrastructure and operational readiness require special emphasis, but they are often overlooked as enterprises begin their migration to IP communications. Determining readiness requires an assessment of and, in some cases, modifications to the existing IP infrastructure and operational processes, skills, and tools. Elements of the infrastructure and operational assessments occur during the preparing, planning, and designing phases.

Evaluate the readiness of your existing infrastructure

Early in the process, you'll need to evaluate your existing network infrastructure to determine if it is ready to support IP communications. It's critical to understand the impact of voice call flows on your network and determine if the IP infrastructure requires greater levels of survivability, security, availability, or capacity to support that traffic. Moreover, voice and video traffic require the network to support quality of service (QoS) parameters to avoid jitter and excessive delay, which may occur when voice and video traffic converge with data traffic.

A critical first step is to evaluate availability across your enterprise's network architecture. You'll need to determine if there are any points of failure in your network and assess redundancy and availability requirements of the new voice traffic on the IP network. Eliminating points of failure requires close scrutiny of the entire network infrastructure, a task that paid off for a major financial services company. A

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By Kathryn Robinson

thorough examination of its infrastructure exposed serious weaknesses in availability, putting elements of critical data applications at risk. This evaluation not only aided in the success of the IP communications project, but also exposed existing risks that, if not discovered, could have been very costly.

Do you have backup power in the event of an outage or emergency? Backup power is much more important for networks carrying voice than those supporting only data.

Are there any incompatibilities among servers or devices? If so, you'll need to configure your switches and routers to match the requirements of the IP communications solution.

What impact will the IP communications solution have on existing network applications and services? How will the new solution function on the network?

Does your infrastructure have sufficient bandwidth to support voice traffic during peak periods? If not, you'll need to add the necessary resources. Whereas in traditional voice networks, users get a busy signal when circuits aren't available, an IP communications network continues to accept all calls, but voice quality can be degraded. Taking advantage of infrastructure features such as call admission control techniques can help avoid these issues.

Does your infrastructure implement class of service (CoS) or QoS technologies on its switches and routers? If not, you'll need to correct this in order to prevent large file transfers from degrading the quality of voice transmission. It's equally important to monitor and review QoS regularly. An East Coast utility company experienced problems when it failed to routinely monitor QoS queues for congestion. Within a period of six months, increased traffic across core wide area network (WAN) links changed queue size requirements, causing intermittent outages of voice control traffic due to the loss of keepalive packets.

As you evaluate your network infrastructure, take a holistic approach: Base your assessment on the readiness of the entire infrastructure, not just parts of it.

Don't underestimate the effort required to get your staff ready to effectively manage an IP communications deployment.

Assess operational readiness

Even when enterprises focus on planning and implementation, they often neglect to adequately assess their operational readiness. As a result, issues of operational effectiveness can arise after implementation, which will have a dramatic effect on the users' experience with the new solution. Therefore, early in the planning phase, it's important to evaluate your operational environment to determine if there are procedures that must be modified (or added) to meet telephony and business requirements.

Most operational environments are not inherently ready for an IP communications solution. A network must meet many requirements to be considered operationally ready to support voice services.

When converging voice onto an IP network, you'll need to determine if your existing operational environment is able to:

- Rapidly detect, isolate, troubleshoot, and resolve telephony problems.
- Pinpoint performance problems such as excessive latency, abnormal jitter, packet loss, and lack of bandwidth.
- Gather and store configurations from network devices.
- Monitor network trends such as capacity and reliability metrics.
- Gather usage statistics and bill for network services.
- Protect the network against unauthorized users and physical and electronic sabotage.

Establish organizational alignment with key stakeholders

Most companies delivering traditional circuit-based voice on private branch exchange (PBX) devices have two networks: one for voice, the other for data. As these networks converge, realigned organizations must work together to deliver high-quality IP-based voice services. Meeting this objective is easier if you establish a clear vision, enabling everyone to work toward a shared goal.

Establishing organizational alignment is absolutely crucial; it can make or break a deployment project. Therefore, you will need to form cross-functional teams consisting of key stakeholders that include executives and end users as well as specialists from support, engineering, and finance. Early buy-in from the cross-functional team strengthens organization-wide support for the goal and makes for more effective individual leadership.

A multinational financial services firm discovered the advantages of establishing organizational alignment and assessing its operational capabilities for supporting voice services. First, the firm identified key areas in which its voice support organization was particularly adept at interacting with business end users. After evaluating the technical skills, processes, and tools of its voice, data network, and information technology (IT) operations teams, the firm identified key areas that would require the teams to work closely on operational issues. Based on the assessments, the firm consolidated its voice, data, and IT operations staffs into two groups: one for network infrastructure services, and the other for IP communications applications. Working together, these teams deliver and support voice services more efficiently and effectively than before.

Integrating voice and data on an IP infrastructure will create new tasks and responsibilities. To avoid confusion and finger pointing, teams must clearly understand their roles. In the world of IP, that's not always obvious. For example, a PBX device can be considered a computer or an IP voice appliance. As a result, it may not be clear who is responsible for managing it. In an actual case, a small company that failed to clarify responsibilities for managing voice servers couldn't restore its database after an outage because it hadn't assigned a team to back it up.

Develop staff expertise

Don't underestimate the effort required to get your staff ready to effectively manage an IP communications deployment. Although your staff may insist that it has the expertise, it probably doesn't.

Therefore, it's prudent to build a solid technical foundation for the entire team, establishing repeatable processes, a knowledge repository, and best practices to avoid reliance on one or two individuals. This approach will help neutralize the effects of employee turnover, especially of highly experienced IP specialists, who are in great demand.

As you build your team, make sure it includes project management expertise. Since IP communications solutions are often deployed in phases, these specialists play a vital role in ensuring that each phase, as well as the entire project, meets both business and technical requirements.

Migrating to IP communications will be particularly challenging for personnel from the traditional voice network environment. They'll need to understand the basics of networking, learn to troubleshoot problems in the IP network, and manage IP communication components. In addition to training, pilot testing offers an excellent opportunity to gain IP technology experience and develop comfort with the new solution.

Acquire the right tools

To maintain the highest level of performance and availability of your IP communications solution, it's crucial to invest in the right set of tools. Base your choice on tools with proven stability, robustness, and scalability.

Fault and performance management must be the primary drivers for tool selection. The tool set must provide a service level view of the entire IP communications infrastructure, including gateways, IP PBX server clusters, unified messaging services, and contact center solutions. It also should be able to correlate multiple and seemingly disparate events to a root cause and point out the overall impact to the service. For performance management, your tool set must be able to perform trend analysis on key IP communications performance indicators such as latency, jitter, and packet loss. In addition, it must be equipped with reporting capabilities to report on IT service-level agreements.

Finally, make sure that the tool set can be integrated with current centralized fault console and ticketing systems. This will not only facilitate standardization, but also help your voice operations group embrace operational procedures.

IP popularity driven by tactical and strategic advantages

IP communications is growing in popularity because of the tactical and strategic advantages it brings to the enterprise. This enabling technology fosters better customer relations and more productive employees through applications such as unified messaging, advanced call center solutions, and voice and video conferencing.

If you're planning to deploy an IP communications solution, remember that success starts with the basics. Adopt a systematic approach, involve key stakeholders early on, build an in-house team of IP experts not dependent on one or two individuals, and determine if your operational environment and existing network infrastructure are ready for IP communications. The time you invest upfront will pay dividends down the road. IT

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Three Steps For Success

Prepare

- Establish a technology vision and business case
- Create a high-level conceptual architecture

Plan

- Create a project plan
- Verify customer requirements
- Develop solution requirements
- Assess network readiness
- Assess operational readiness
- Assess site readiness

Design

- Develop a detailed design
- Create a staff training plan
- Develop a network implementation plan
- Develop a certification test plan
- Develop a site-specific network implementation plan

Understanding and Mitigating Risk in the VoIP World

As more and more businesses and consumers are finding out, there a lot of inherent benefits in using Voice over IP (VoIP) technology.

Businesses appreciate VoIP (define - news - alert) for a multitude of reasons, such as the fact that they can collapse redundant voice and data networks into one infrastructure, and that they can transparently link several remote locations and mobile workers through a single system, not to mention the reduction in network administration costs they enjoy in most instances.

Judging by the adoption rates the cable providers claim they're getting for their VoIP offers, consumers, too, seem enamored with the technology, particularly with the "unlimited" calling plans that have permeated the market.

With the air of panacea that VoIP is currently enjoying, one would conclude that telcos, cable companies, and other service providers that offer VoIP solutions would be delighted with this uptick in the public's embrace of technology. After all, sales are up, demand is up, and the reliability and stability of VoIP technology is constantly improving. So what's there to worry about?

According to the VoIP providers, there's plenty to worry about. Proffering VoIP services is one thing, but managing usage, maintaining revenue, and complying with various security, financial, and regulatory mandates is what's keeping the providers up at night.

Voice over IP offers a unique set of challenges that all providers must wrestle with in order to maintain the financial and operational integrity of their respective organizations.

The Myth of "Unlimited" Services

VoIP providers face several inherent risks in offering VoIP services — especially to consumers who are using "unlimited" services. First, let's be clear: there is no such thing as "unlimited" services. Every provider we've come across — including Vonage and the cable providers — place limitations on their VoIP offerings. Most contracts list a number of conditions that give providers the right to shut down services. These may include unauthorized business use, auto dialing, faxing, or other behaviors not typically related to consumer use of phone service. One nationally known provider even goes so far in its contracts to state that it reserves the right to shut down service immediately if in its sole discretion it determines that calling patterns appear to be irregular.

These alleged "unlimited" services are starting to cost VoIP providers a bundle, particularly in international calling plans. Each time those "free" calls come off the IP infrastructure onto the public telephone network, the VoIP provider must pay a termination fee to the destination telco. That would be reasonable in normal phone usage, but now consider the very nature of these unlimited plans. In many ethnic neighborhoods around the globe, the VoIP phone becomes the neighborhood phone. These termination charges rack up pretty fast when 20 or 30 international calls are made from a single location, eating up whatever profit was built into the unlimited offer that was sold to the consumer.

Beware: Sarbanes-Oxley (And Other Mandates)

There are other areas of concern VoIP operators should be aware of in addition to the inherent financial risks of offering unlimited packages.

VoIP providers that trade publicly, including international-based operators that are listed on U.S. exchanges, must

By William “Duffy” Mich

comply with the various regulations of the Sarbanes-Oxley Act, which mandates certain operational and financial reporting criteria. The problem for VoIP providers in meeting these mandates is twofold. First, the very nature of VoIP, as a portable communications technology that has the ability of masking caller origination and identity, presents a challenge for service providers who must be able to draw a clear line between services offered, revenue, costs and profit in order to meet Sarbanes-Oxley requirements. In the world of VoIP, it is exceedingly difficult for providers to capture accurate call data, and assign the appropriate fixed costs and toll charges to these calls.

The other reality is that many VoIP operators simply lack the sophisticated reporting and accounting systems that can accurately track and monitor usage among customers. For most of these providers, the lack of systems is not intentional. VoIP is still a largely unregulated business, but the emergence of Sarbanes-Oxley, along with other regulations coming out of the European Union and other countries, will force providers to pay meticulous attention to VoIP reporting and accounting activities.

Meeting DHS Requirements

Security and other related initiatives will also have a profound affect on VoIP providers. As it stands today, the United States Department of Homeland Security has the legal authority to demand call detail records from any service provider that does business in the United States, including telecommunications operators, cable companies, Internet service providers, and the emerging VoIP providers, like Skype. Again, these federal requests may prove to be problematic in the unregulated world of VoIP. While call records and Web hits are fairly easy to collect and produce in the traditional telecom and ISP models, they are not as readily available for VoIP companies. Some providers do not have systems in place to track customer activity in sufficient detail and, with the portable nature of VoIP, many have chosen to overlook these fundamentals. Unfortunately, the U.S. Government is not nearly as forgiving, and failure to produce accurate customer data could cost VoIP operators dearly.

Simple Strategies to Mitigate Risk

All is not gloom and doom for VoIP providers who wish to minimize potential risk. Every operator already owns the critical data to protect against these threats. All they need to understand is how to go about this task. There are a number of strategies that VoIP operators can easily and cost-efficiently employ to protect their organizations against unauthorized usage and the subsequent negative impact. For example:

• Near real-time switch data and traffic pattern analysis. These technologies enable providers to obtain a clear snapshot of what typical VoIP usage looks like. As the baseline profile is drawn, ongoing analytics and evaluation can help identify customers that regularly exceed acceptable thresholds of usage. Usually, three continuous months of data retrieval and analysis is sufficient to generate accurate profiles. Implementing these analytics are relatively easy for the provider, most can be readily integrated into an operator's customer relationship management (CRM) platform.

• Once customers who have abused these VoIP services have been identified, operators should be very cautious in how they approach these end users. A tiered approach is often the best solution.

For example, if a very good cable customer (one that uses voice, data, and video services) is regularly at the threshold of VoIP usage, the provider may wish to send this customer a subtle reminder of the terms of VoIP services, as to not alienate or offend a customer that spends a good deal of money on multiple services.

On the other hand, a customer that uses only VoIP services, resists cross

In many ethnic neighborhoods around the globe, the VoIP phone becomes the neighborhood phone.

selling overtures and regularly racks termination charges as a "neighborhood" phone, is the perfect candidate to have his or her service terminated immediately, no questions asked.

Enforcing these risk management policies demands insight and nuance on the part of the provider — but all decisions can be reinforced by properly evaluating call data and other analytics.

Conclusion

As VoIP services continue to gain increased market and mind share among businesses and consumers, providers need to be cognizant of the potential risks they may face if these services are continually abused. Expensive termination charges can quickly add up by users who make an abnormally high volume of calls, particularly in "unlimited" plans that may include international dialing.

In addition, operators are required to maintain meticulous call details, that could end up justifying billing and accounting processes to meet financial reporting mandates, or may even be subpoenaed as part of Homeland Security compliance.

The easiest way for VoIP providers to manage any risk continues to be a thorough analytics capability using call data records that can build up a reliable profile of users who have the propensity to use VoIP in unauthorized ways — and then take the necessary steps to either persuade the user to curb this behavior, or shut them down completely. The one thing a provider cannot afford to do is ignore this issue entirely. IT

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The Enterprise PBX is Dead!

Long live the enterprise PBX.

Organizations first began installing private branch exchanges (PBXs) to avoid paying their telephone service provider for a call to a colleague down the hall. That has evolved over the years. Today, IP telephony is increasingly sold as unlimited calls for a single monthly rate and the cost avoidance justification no longer applies.

Over the years, a variety of additional services have been added to the definition of an office PBX, (define - news - alert) and those services have come to define customer expectations. Services will continue to be enhanced, but the ways in which they are delivered will change radically from the closet full of equipment with which we are familiar.

For one thing, much of the equipment will sit in racks in service provider's colos (co-location facilities). Unless an organization can take advantage of the productivity payback from integrating telephony into its line of business applications, it will most likely obtain its PBX functionality for free from a telephone service provider, bundled into a flat-rate monthly charge that includes unlimited calling. Small and medium-sized enterprises are the most likely to go the service provider route.

When organizations have a business reason for maintaining their own equipment, it'll most likely be because they get major productivity benefits from integrating voice into their business processes. Their PBX functionality will be running as software on just another rack of standard servers in the computer room with the manageability, programmatic interfaces, and support that enterprises demand.

Barring FCC action that changes the playing field, it's not a very long journey from where we are today to that vision of the future. And this is seismic change that will impact most of the players in the telephony business. According to the latest *Enterprise Telephony* report by Infonetics Research (http://www.infonetics.com), TDM and IP PBX system revenue totaled $8.1 billion in 2005, a 12 percent increase over 2004 and a trend that is accelerating. As a significant fraction of that spending shifts into new patterns, fortunes will be lost as well as made.

The first part of the change in corporate purchasing behavior will be the collapse of the low-end PBX market as service providers respond to competition by bundling complete PBX replacements into the price of their IP telephone service offerings in order to slow price erosion. Richly functional hosted PBX products based on industry standard hardware and open source software are offered currently for as little as $10 an endpoint. Service providers already spend a lot more than that on free customer premise equipment to acquire a new customer and the low cost of a hosted PBX offering will inevitably make it a standard part of the service portfolio within a few years.

The functions of a hosted PBX are usually a superset of the five to eight year-old hardware PBX systems installed at customer premises. Hosted PBX customers don't have to buy, install, or maintain any PBX equipment. Instead, the PBX equipment is kept by the service provider, who maintains the system and shares access among many users.

This Is Not Our Parent's Centrex

Developed in the 1950's, Centrex service is a business telephone service offered by what used to be the local telephone company from a local central

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By William Boehlke

switching office. Centrex service offers a very restricted feature set that includes:

- Call Forwarding (eight varieties)
- Key System Emulation
- Call Hold
- Call Pick-Up
- Call Park
- Call Waiting
- Voice Mail
- Automatic Callback
- Automatic Recall
- Call Transfer
- Conference Calling
- Music on Hold
- Speed Calling
- Intercom
- Message Light

Hosted PBX functions, on the other hand, start with all Centrex features and add:

- Auto attendants
- Voice mail for groups as well as people
- Custom pre-recorded greetings
- Multi-level voice menus for directing calls, connecting to them specific extensions, with a user interface for designing custom voice response systems
- Enterprise dial-by-name directories
- Placing callers on hold
- Playing custom music or messages whenever callers are waiting on hold
- Transfer of calls between extensions
- Conferencing multiple incoming calls with employee extensions
- Detailed call records for billing
- Individual user options such as e-mail integration with voicemail, call screening and follow me
- Multi-language voice prompts
- Self-service graphical user interfaces for system management and individual handsets

Expect to see hosted PBX offerings for small business that include all this functionality as well as unlimited calling in the United States for as little as $30 a month per extension later in 2006.

Where the hosted PBX probably does not make sense is in larger enterprises that have economies of scale comparable to small service providers. There, the integration of voice into line of business applications will keep a form of IP PBX in-house, one that's likely to look a lot like a software service running in the data center.

The software development industry is currently in the midst of solving telephony's enterprise integration issues. The most important pre-condition for dramatically reducing the time required to integrate is that services developed for one application should easily be combined and re-purposed for other applications. SIP is the likely vehicle for solving this problem.

SIP, or Session Initiation Protocol, is the application layer signaling protocol defined by the Internet Engineering Task Force (IETF) for initiating, modifying, and terminating communication sessions between endpoints in an IP network. SIP is rapidly becoming the de facto underlying session control protocol among service providers and its current momentum will almost certainly carry it as a standard into the enterprise.

SIP (news - alert) signaling capabilities were once the purview of specialized telephony protocols, such as Signaling System 7 (SS7), running on dedicated hardware. Today, SIP application servers connect to a SIP network via a SIP Proxy Server, enabling calls to SIP functionality from widely used programming languages. For example, SIP has recently been incorporated into the Java language under JSR-116. SIP also has hooks into SNMP-enabled enterprise consoles, like HP's OpenView, potentially addressing enterprise concerns about manageability.

Some examples of the integrated forms of communication that SIP enables include:

- Click-to-call
- Voice instant messaging
- Voice-video telephony
- Video conferencing
- Real-time video sharing
- Rich-media enterprise collaboration
- Auto-initiated conference calls

The Benefits of VoIP without the Hassle

By Michael Shelton

Today's call center managers face a continual struggle to improve productivity and customer service levels while keeping costs down. This is especially true for the 90 percent of U.S. contact centers that have less than 100 agents. These centers find it particularly difficult to cost-justify the purchase of traditional premise-based, advanced contact handling features that bolt on to an existing PBX or phone system. As a result, many companies are finding hosted IP services more attractive because of the ability to gain access to these productivity-enhancing features without the heavy, upfront capital expenditure.

While many companies are exploring VoIP, many are under the mistaken impression that VoIP equipment is required at their premises in order to take the next step in their call centers. In reality, companies are able to deploy a hosted solution on a VoIP network using their existing phone systems. Now, a company can access advanced technologies without the hassle and expense of purchasing new phone equipment.

Managed and hosted contact center services are typically paid for on a monthly or usage basis, with little upfront cost. This means that rather than committing significant upfront capital expenditure, contact center technology is purchased as an ongoing operational expenditure. According to Datamonitor, small organizations with tight controls on cash flow and capital expenditure benefit the most from this approach, but the cost measures and limits on capital expenditure are currently being experienced by organizations of all sizes.

Hosted services offer several advantages over premise-based solutions. Most customers of less than 100 seats are not able to cost-effectively deploy systems that have the capabilities found in standard offerings from the best hosted service providers. In addition to the costs associated with acquiring, installing, and integrating these advanced systems, scalability of each of these system elements also has a big impact on the cost structure of the operation.

For example, with an on premise solution, the operation must build out each system element to support peak periods. That means the capacity to support all call center functions during seasonal spikes must be accounted for in available phone lines and circuit cards. It's a fixed cost that can not be "turned off" during slow times.

With hosted services, companies are able to manage capacity on a week-to-week basis, scaling up one week and scaling back down, if need be, the next. Companies utilizing a hosted solution only pay for capacity as they need it.

Another advantage of hosted services is that feature integration and upgrades are handled by the system provider and are no longer the regular responsibility of the end user. This is especially beneficial to smaller businesses that do not have the time or technical team to deal with system integration and maintenance issues.

Datamonitor has also said, "purchasing a premise-based all-in-one solution rather than integrating a number of best-of-breed components will reduce integration and installation costs, but a managed or hosted service offers a number of other benefits on top of that. These benefits derive from the expertise and economics of scale that a service provider can achieve, and that very few enterprise or public sector bodies could."

While many companies are exploring VoIP, many are under the mistaken impression that VoIP equipment is required at their premises

A common myth among call center managers is that you lose control with hosted services. While this is true with some hosted providers, companies utilizing the better service providers have found the opposite to be true. The best hosted services conform to the specific requirements of the business, rather than forcing the business to conform to capabilities of a canned solution. Hosted services today offer a level of configurability and flexibility that was only dreamed of by customers utilizing the traditional equipment of the past. Customers are finding the new breed of hosted systems to provide an even greater level of functional control than they currently experience with their on-site equipment while simultaneously ridding themselves of the burdens of day-to-day management.

Hosted services can provide the benefits of VoIP without requiring the customer to upgrade on-premise telecommunications equipment. One of the biggest motivations behind moving to VoIP for many companies is the fact that they can seamlessly connect multiple sites and at-home workers with one unified system. However, some companies incorrectly see new IP-based phone equipment as the only solution.

Some hosted vendors actually enable a company to keep its existing phone system, yet obtain the advantages of an IP-based telephone system. These solutions allow customers to create any combination of TDM and VoIP connections. Agents at home might opt for a VoIP connection while agents at the corporate headquarters utilize their existing corporate PBX-based telephones. Since the system capabilities exist in the VoIP network, each connection, regardless of type, is viewed as a virtual extension. Each user, regardless of location, has the benefits of the network-based VoIP system.

The flexibility provided by blending TDM and IP connections to VoIP services enables a customer site to receive the benefits of a VoIP solution without the hassle or expense of switching over to VoIP-based equipment. As a result, the cost and risk of obtaining the benefits of VoIP is drastically reduced.

IDC predicted that, by 2008, IP will account for 72 percent of new telephony connectivity. Companies are moving to VoIP, but they can now immediately realize the benefits of VoIP but transition their equipment on their own schedule. With the maturing of VoIP technology and the availability of advanced contact handling technologies in a hosted environment, companies have a very low-risk option. They are able to make the transition from a premise-based environment to a hosted solution with minimal disruption and expense. IT

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• Converged call center communication

These new functionalities will be embedded into line of business applications in task-specific ways. Just as contact center agents use very specific calling functions instead of general purpose PBX capabilities, many other business functions will see improved productivity from embedded communications. For example, salespeople will have tight integration with their customer relationship management application so they can organize customer contact based on deals in progress and initiate a call with a simple mouse click. Accounts receivable managers may need to periodically contact all past due customers, where the application presents them with a series of calls and the appropriate information for each call before it happens. Early versions of this type of integration are already deployed in leading edge companies.

Where does the growth of hosting and telephony integration leave the general purpose PBX? There's an old marketing saying that success is found at the ends of the spectrum. Leaders typically emerge in the low-priced, high-volume segment and the high-priced, high-service segment, but rarely in the middle of a market. The general purpose PBX will increasingly be squeezed between nearly free hosted IP PBX service and highly productive integrated telephony capabilities in line of business applications within the enterprise.

So, can we say that the enterprise PBX is dead? Not really, but it's definitely entering old age. The capabilities of the enterprise PBX will live on in other forms. IT

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VoIP Peering for MSOs

Now That We Know It Works, What's Next for VoIP?

When cable operators first approached Voice over Broadband (VoBB) services, there was much about the technology that was yet unknown. Even once the case was made for the technology in general, cable operators still faced many challenges in actual deployment of VoIP networks. Although many of these challenges remain, the indisputable fact is that VoIP is here and it is here to stay. With the abundance of VoIP networks, whether or not VoIP will take the Darwinian step of replacing the traditional Public Switched Telephone Network (PSTN) depends on the ability of innovators to bridge the islands of VoIP, and provide interoperability and interconnectivity solutions, ultimately turning all VoIP networks into one global network.

Voice over Broadband services provided by cable operators are a compelling offering for a number of reasons. On the business side, a triple-play package of Internet access, content (TV channels), and voice is extremely attractive to consumers. On the technical side, since the cable network spans that last mile directly into the home, Quality of Service (QoS) can be controlled to assure positive user experience. By contrast, non-cable Internet Telephony Service Providers (ITSPs) like Vonage or Skype will have an uphill battle on both these points. It is little wonder that the cable VoIP (define - news - alert) offerings are the fastest growing segment in the consumer space and have by far overtaken other (paying) VoIP services in numbers of users, minutes of use, and revenue generated.

Cable companies are now looking beyond the birth pains of their initial deployments to the next generation of opportunities and challenges — especially to VoIP Peering between Multiple System Operators (MSOs). Peering allows the cable networks to keep a VoIP call IP to IP, end-to-end. Currently, a VoBB user calling another user of the same Cable voice service will never go "off-net." But a call placed to a user of a different VoBB provider, even if both end users have identical IP phones, will leave the first cable operator's IP network, drop off to the legacy PSTN, and continue on to the terminating MSO's IP network and on to the end user's IP phone. This detour in the communications route presents MSOs with a number of challenges and difficulties.

Firstly, multiple conversions between VoIP and PSTN protocols degrade voice quality. Secondly, an unnecessary cost component of PSTN per minute charges is added. This can be quite substantial in international dialing, and ultimately causes service providers to either transfer costs to users, or reduce revenue. The third, perhaps most compelling argument for peering is the retention of advanced features allowing IP communications to realize its true potential. When a call leaves the IP network, it drops to the lowest common denominator with the PSTN, namely toll-quality voice. On the other end of the spectrum, IP end-to-end communications allow for video, CD quality voice, voice/data integration, and features and applications yet to be developed.

While VoIP peering makes sense for other ITSPs, organizations, universities, and large enterprises as well, it is natural for the Cable industry to be the first mover in this space. MSOs are typically regional and don't compete for each other's user base, so cooperation between them is a non-threatening, win-win situation.

The challenges involved in VoIP peering fall into three categories: Interoperability, Routing, and Security. VoIP network deployments using different softswitches or SIP proxies (not to mention different protocols) rarely work seamlessly with each other. It is usually necessary to tweak the call flow in one way or another. Session Border

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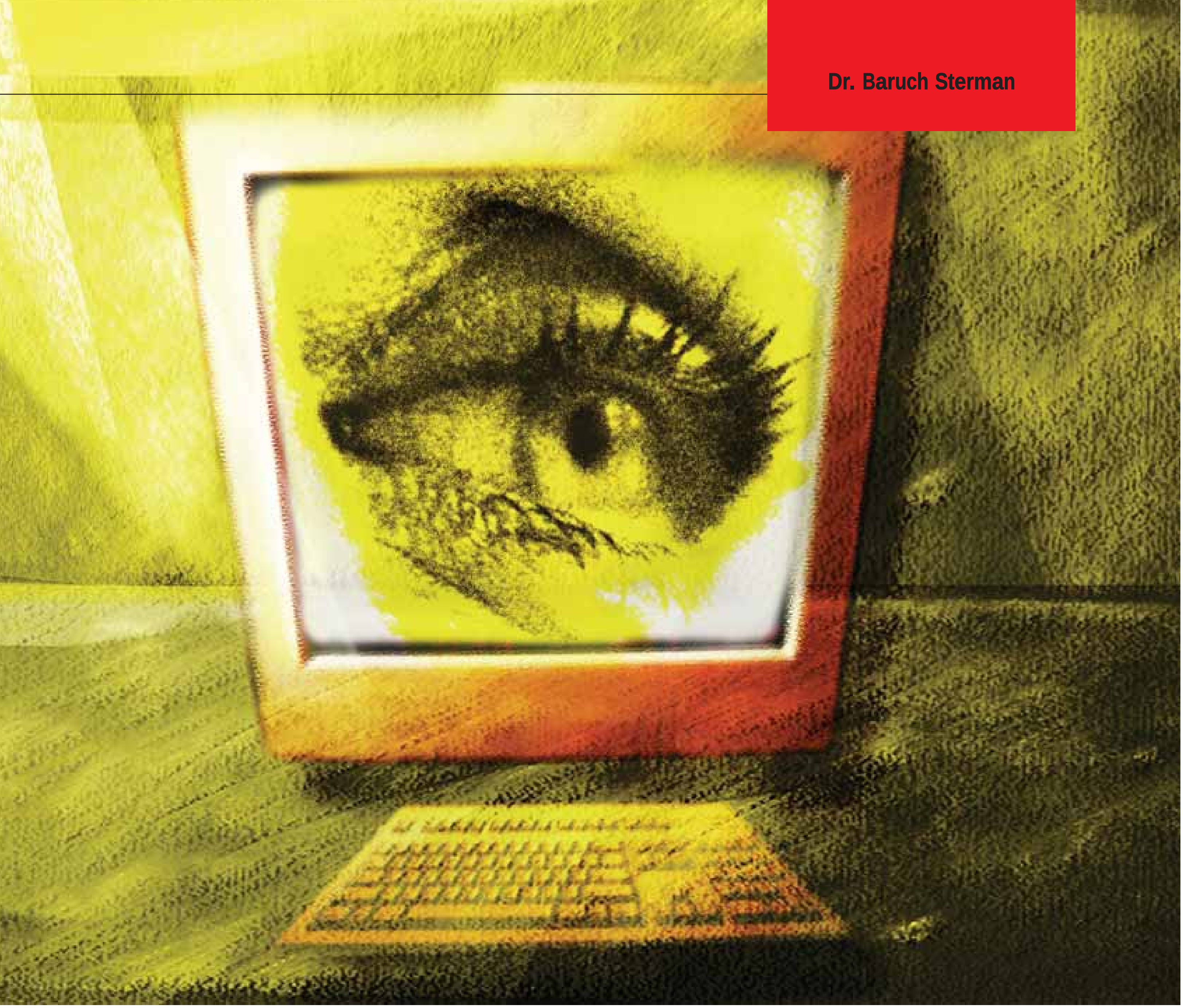

Dr. Baruch Sterman

Controllers (SBCs) can often solve these problems, but the dynamic nature of the problem set with new versions of software or additional features and applications causing trouble makes maintaining, patching, and upgrading an SBC an operational headache.

Routing calls between MSOs requires each operator to determine if the call is destined for a VoBB user of another MSO or for the regular PSTN. Since many countries support Local Number Portability (LNP) there is no way a-priori to know if a particular phone number will end up on an IP network without a directory service comprising the phone numbers of all members. That database needs to be queried on a call by call basis. The protocol of choice for this lookup is Electronic Number Mapping (ENUM).

Security challenges are the most formidable and complex, and certainly not yet completely understood. By opening up their networks to each other, MSOs are breaching their "walled garden" and are thus susceptible to new vulnerabilities. These include things like Denial of Service (DoS) attacks, worms and viruses embedded in the signaling or media, and malformed protocol packets (accidental or malicious) capable of crashing VoIP systems. There are also higher level problems such as Identity theft (which is especially tricky in VoIP as the CallerID is usually open and unprotected) and Spam for Internet Telephony (SPIT).

SPIT differs from regular telemarketing on the one hand and from e-mail SPAM on the other. For spammers, since there is essentially zero cost per call, there is no economic constraint on the volume of calls. MSOs know that as soon as the VoIP space becomes a safe haven for spammers, users will relinquish these services, and they must act fast to prevent SPIT. However, methods of detecting SPIT are problematic since there is no content to examine and filter, the caller identity is generally untrusted, and the ringing phone alone is already intrusive and annoying.

Many of the above issues are best addressed by having a federated peering

MSOs are typically regional and don't compete for each other's user base, so cooperation between them is a non-threatening, win-win situation.

authority instead of the ITSPs attempting to peer directly with each other. The bookkeeping involved in managing the directory services, for example, is much simpler in the federated architecture. There is also greater flexibility for solving security issues with a central authority involved. Policy can be enforced and abuses monitored more efficiently. SPIT is another area that can be combated more effectively within a federation. It is easier to detect patterns that could identify a caller as abusive if calls flow through a central point than if traffic is solely peer to peer. For example, if a user sends out a large number of calls to users of multiple MSOs, each network will receive only a small fraction of those calls, and may therefore not notice any irregularity. Mining the call patterns at the crossroads between all federation members, however, allows for greater statistical accuracy in identifying abuse. The federation also has information about the type of service its different members are providing, like call centers, and that profile may be important to other members deciding whether to accept or filter calls. The information regarding the profile, identity, and security level of the call needs to be embedded within the call flow so that it is available to a network or end user on a call by call basis. For example, a user may decide that specific types of calls at certain hours should divert directly to voice mail instead of disturbing him/her with a phone ring.

The first generation of VoIP was about proving the technological concepts, and the second generation focused on aggregating groups of users and deploying VoIP networks. The future of the industry lies in bridging these islands and connecting up the networks so that VoIP can reach its full potential and eventually replace the legacy phone system with a feature rich and robust communication platform. IT

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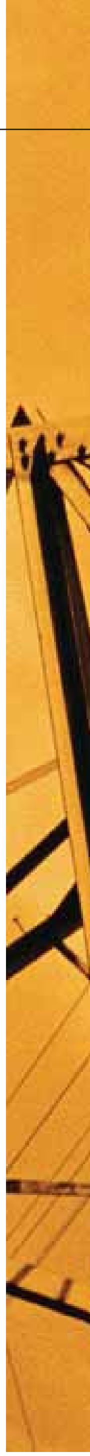

Powerline Communications Extend Converged Home Networks

Within the past few years, American homes have become saturated with new forms of communications. Digital phones, data, and entertainment services such as video over IP were novelties in the home a decade ago, and are now commonplace, with more on the way. As the need to share different types of information among these systems becomes increasingly evident, service providers are looking for networking technologies that are capable of converging data, voice, and audio/video for delivery throughout the home. The solutions they discover should employ existing physical media whenever possible, since the converged home networks will have to be easy for consumers to understand, use, and afford.

One type of network that comes close to fitting the bill is the household electrical grid. Residential powerlines are not usually thought of when considering communications networks, but in many cases, they can provide an excellent medium for home networks transmitting converged digital services. With an average of over 40 power outlets in a typical U.S. home, each one becomes an entry point into a potential home network. In comparison, there are fewer numbers of coaxial cable or telephone jacks. As different technologies become spotlighted by various suppliers and the pros and cons become more evident, consumers and service providers may soon begin to take advantage of this valuable but overlooked resource.

Advantages of Powerline Communications

Consider the physical media available in today's typical home.

• Telephone Lines — Over 95 percent of U.S. homes have phone lines based on copper-based twisted-pair, but the number of telephone jacks in a home typically don't number more than six or seven. Bandwidth on the phone network may vary, depending on the wiring, but is usually reasonably high.

• Coaxial Cables — Over 90 percent of homes have coaxial TV cable, and coax is high-capacity by design. But, like telephone jacks, the number of coaxial cable jacks seldom number more than five or six.

• Category 5 Wiring — Like coax, Cat 5 is satisfactory for the high throughput requirements of digital video, but these networks normally go to only certain rooms in the house, and they provide a single outlet in each room. In addition, Cat 5 is available in less than 10 percent of U.S. homes and, thus, requires a laborious installation for a majority of Americans.

Household powerlines, on the other hand, go to every room and provide multiple sockets in each. If the electrical network were used for communications, every power outlet in the house would be a potential connection.

Wireless local area networks based on the 802.11 standard, are intended to reach everywhere in the home — or at least they should. To their credit though, WiFi systems are inexpensive and easy to install and, as a result, they are already widely deployed as laptops with WiFi have also proliferated both consumer and business PC arenas. Unfortunately, wireless LANs do not have enough capacity for multiple streams of high-quality video and, in

By Michael Stich &
Corey Chao

many installations, they vary in their transmission data rate and range, due to local interference such as cordless phones and other devices that use the same frequency spectrum. For data networks, wireless drop-offs normally do not present a serious problem because PCs (notably laptops) can be moved around for better reception, and the traffic tends to be low bandwidth, low priority data. But, for high bandwidth content that has to be received in real time, such as HD or SD video, erratic transmissions are unacceptable. As for the future of wireless LANs, the Ethernet 802.11n standard is intended to address these issues in the near future.

Today, powerline-based networks are intended to avoid the rate/reach issues that are inherent in wireless networks, delivering a more consistent performance throughout the household grid. Powerline networks provide a much larger, more reliable channel for communicating converged data, voice, and audio/video content than WiFi networks. If coaxial and CAT5 cables provide the most consistent medium and wireless LANs the least consistent, then powerlines sit somewhere in between. Powerlines can, thus, provide a useful extension to the home network, supplementing and overlapping the other media, though not completely substituting for any of them.

Technology Industry Initiatives

Currently, there are no ITU or IEEE standards established for home powerline LANs, though the technology issues are being widely discussed. Of the many alliances among companies that address home networks in general, the HomePlug Alliance, with more than 60 members at all supplier levels, is the largest organization devoted to defining

If the electrical network were used for communications, every power outlet in the house would be a potential connection.

powerline LANs. In 2001, the HomePlug 1.0 standard established powerline networking and today is supported by products from four integrated circuit (IC) manufacturers. The more recent HomePlug AV standard has the announced support to date of 11 IC suppliers and a number of service providers.

HomePlug 1.0 tops out at 14 megabits per second (Mbps), limiting its usefulness to data exchange. On the other hand, HomePlug AV extends the bandwidth minimum to 70 Mbps over powerlines, and it also delivers transmission over cable with a minimum of 100 Mbps. Coaxial transmission could support multiple channels of high-definition (HD) video, depending on the compression used, with additional bandwidth for data and voice. Powerline transmission would be limited to two channels of standard definition (SD) video or a single HD channel, plus data and voice. That HomePlug AV can span different physical media helps enable powerlines to serve as a mid-capacity extension to higher-capacity carriers in the household LAN.

HomePlug has been extremely successful in North America, but in Europe it faces a challenge from the Universal Powerline Alliance (UPA), which delivers maximum transmissions of 90 Mbps over powerlines and even greater performance over coax. Like HomePlug, UPA proposes open standards, though with 11 alliance members, a single IC supplier and no committed service providers, it is not yet widely supported. A different alliance, CEPCA, is focused on the Japanese market, but its standard is unlikely to find use elsewhere.

Other industry initiatives that also seek to leverage the existing infrastructure for home networking include the Media over Cable Alliance (MoCA) and the Home Phone Network Alliance (HomePNA). The 802.11n Ethernet committee and WiFi Alliance will also have significant impact with a standard that upgrades wireless LANs. Given the likelihood that the home network will provide converged information exchange over diverse media, all of the preceding alliances and possibly others may be important in defining residential networking for the future.

Converged Services for Home Networks

For service providers, home networking brings new business opportunities, along with some concerns. Cable TV providers can add VoIP phone services to their products, and telephone providers can offer IP-based television (IPTV). Both types of providers, of course, already supply Internet access. Whatever the source of the converged information services, digital voice calls and radio/TV transmissions take place in real time and require priority over the bursty traffic of the data network. Reserved channels and quality of service (QoS) mechanisms can help resolve these potential conflicts.

Home entertainment LANs will add to the complexity of digital rights management (DRM). The content will not only be more widely disseminated, but it will also be harder to control on the wireless residential connections than on the wired ones. When these factors are added to the basic technical problems inherent in WiFi transmissions, service providers may prefer to keep their support for wired and wireless networks separate for the time being, at least until the industry has resolved the DRM, rate, and reach issues.

System Considerations

While consumer products that can support powerline communications have not yet appeared, the technology is available to integrate directly. RGs, STBs and other central units will be based on IC technology that provides high performance for signal processing, along with flexibility for adapting to emerging standards, different products and various regional requirements.

As residential communications and entertainment requirements continue to evolve, powerlines will become an important element of converged home LANs. No medium goes everywhere and carries all information today, nor is there likely to be an ideal medium in the near future, even after 802.11n Ethernet adds considerable capacity to wireless data LANs. A number of industry initiatives look to bring powerline networking to the mainstream, with the HomePlug Alliance taking the lead in North America. DSPs and advanced analog ICs provide the technology that will enable residential networks to deliver data, voice and audio/video information via a variety of carriers throughout the home.

Programmable digital signal processors (DSPs), which have demonstrated their value in systems with these requirements, will be widely used as an enabling technology. In addition to handling the basic data manipulations required to shift data among different carriers, DSPs provide the performance needed for encoding and decoding voice and audio-video information — functions that are essential for home entertainment networks. High-power analog ICs will also be indispensable for linking digital circuitry with carrier electrical wires. An IC manufacturer with leadership in both DSP and analog processes is well-positioned to provide the enabling technology for this emerging form of communications. IT

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Telecom Developers' Guide to Open Telecom Platforms

CompactPCI, ATCA, AdvancedMC and MicroTCA

Throughout the world, the requirements for new packet-based triple play and other enhanced services are driving the need for improved infrastructure. In order to satisfy that need, suppliers of telecom equipment are also looking to more flexible telecom platforms that enable them to reduce cost while speeding time to market. For many equipment providers, this means a move away from in-house proprietary systems toward open platforms, such as PICMG 2.16, ATCA, AdvancedMC and MicroTCA.

Open architecture systems reduce cost and time to market by making it easier for equipment providers to outsource a large portion of their system design and utilize cost-effective, best-of-breed off-the-shelf hardware and software components. In this way, equipment providers can streamline their engineering staffs and focus on network applications and services that offer greater value added and differentiation.

Open Platforms Evolve

CompactPCI was one of the first open platforms to peak the interest of equipment providers, particularly after adding telecom-friendly features like hot swap and a dedicated H.110 telephony bus. Hot swap enabled service providers to remove and replace individual CompactPCI blades from live shelves in the field without having to disable the shelf and disrupt overall service. The dedicated H.110 telephony bus enhanced data flow efficiency by enabling CPCI systems to acquire TDM (time division multiplexed) data, process that data, and move it between multiple blades in its native format.

What really made CompactPCI attractive to TEMs, however, was the addition of support for packet transport and system management. PICMG 2.16 (CompactPCI Packet Switching Backplane) added support for Ethernet backplane transfers, a key requirement for next-generation IP-based packet infrastructure. PICMG 2.9 added a system management framework based on the familiar enterprise Integrated Peripheral Management Interface (IPMI) model, which made it easy for remote shelf management systems to monitor and control individual blades.

AdvancedTCA Provides Next-Generation Open Framework

PICMG has continued to develop and improve the CompactPCI specification by exploring new telecom-friendly enhancements, consolidating those enhancements under the CompactTCA banner, and phasing out the use of PCI as the primary control and data plane. Collectively, these enhancements enable the blades in a CompactTCA chassis to be treated as "network elements," a significant improvement over the traditional CompactPCI master/slave bus architecture.

Even as CompactTCA enhancements continue apace, Advanced Telecom Computing Architecture (AdvancedTCA or ATCA) is quickly emerging as the leading contender for telecom infrastructure applications. Adopted in January of 2003 with input from key equipment and service providers, ATCA provides a high-performance, high-density platform aimed squarely at next-generation packet networks.

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By Todd Wynia

AdvancedTCA is an open architecture framework for building high-performance, high-density, high-availability (five nines or greater), NEBS-compliant, 19-inch, rack-mountable telecom shelves. The foundation for ATCA is its high-speed switched fabric, which provides a peak throughput of 10 Gbit/sec per link, ten times that of PICMG 2.16 backplanes. The ATCA fabric supports a full mesh interconnect, which enhances availability by enabling each blade to simultaneously communicate with every other blade via dedicated channels. The ATCA fabric is also protocol agnostic, enabling it to support multiple packet-oriented protocols, including Ethernet, Infiniband, PCI Express, and Rapid I/O. PICMG 2.16, by contrast, specifies Gigabit Ethernet as the transport.

In addition to its high-speed fabric, ATCA provides numerous other features that are also critical for equipment providers. Its large form factor (8U) and high-power capability (200W per blade, versus 50W for CompactPCI) give ATCA the capacity to support complex functions and high-density configurations. And its redundant fabric, redundant power, and hot swap capabilities reduce susceptibility to point failures and enable individual blades to be serviced and upgraded without disrupting overall service.

One of ATCA's most attractive features for the service providers and carriers is its support for IPMI system management, which enhances availability by facilitating active monitoring and control of individual ATCA blades. IPMI utilizes an I2C-based physical interface to link chassis management with board-level FRUs (field replaceable units). Through this interface, chassis management can monitor physical system health characteristics such as voltages, fan speeds, temperatures, and power supply status. Chassis management can also utilize IPMI for automatic event notification, remote shutdown/restart, and to dynamically allocate power to individual blades, which helps optimize system-wide power consumption and cooling.

ATCA provides a high-performance, high-density platform aimed squarely at next-generation packet networks.

CompactPCI, through the PICMG 2.9 add on, provides a comparable management framework. The ATCA framework, by contrast, is incorporated as part of the baseline ATCA spec (PICMG 3.0), building on the PICMG 2.9 spec to provide a higher level of detail and additional IPMI commands.

AdvancedMC Modules Enhance ATCA Flexibility And Scalability

ATCA carriers can be equipped with up to eight AdvancedMC modules, which come in four sizes: half-height single-width, half-height double-width, and a full-height version of each. The field replaceable modules have escalating power limits of 20W for the smallest module to 60W for the largest module.

AdvancedMC enhances ATCA flexibility by extending its high-bandwidth, multi-protocol interface to individual hot-swappable modules. This provides TEMs with a versatile platform for building modular telecom systems that can be outsourced, designed/manufactured, stocked and spared at a lower cost. The modular architecture also reduces service provider operating expenditures by reducing the impact of component failures, and enabling service providers to scale, upgrade, provision, and repair live systems with a finer degree of granularity and minimal disruption to overall service.

MicroTCA Addresses Low- To Mid-Range Applications

The ATCA/AdvancedMC platform is an outstanding solution for many mid-range to high-end telecom infrastructure applications because of its high performance, modularity, and five nines reliability. These features, however, come with a price tag that can be too expensive for many central office, outside plant, and customer premises applications. ATCA's generous form factor is also a stumbling block for outside plant applications such as wireless base stations with tight space constraints.

To serve these low- to mid-range telecom applications with space and/or cost constraints, PICMG is in the process of developing a new specification based on the AdvancedMC platform known as MicroTCA. With ratification expected in June 2006, MicroTCA essentially eliminates the ATCA carrier, enabling equipment makers to directly utilize AdvancedMC modules in a variety of enclosures.

MicroTCA enables equipment providers to leverage the installed base of off-the-shelf AdvancedMC modules, while achieving lower cost in a smaller footprint. MicroTCA also enables equipment providers to utilize the same serial fabric interface and integrated IPMI system management used in ATCA/ AdvancedMC systems. This combination makes MicroTCA an outstanding complement to ATCA for small form factor central office and outside-plant applications like wireless base stations, WiFi/ WiMAX radio boxes, next-generation digital loop carriers, optical ADMs, and Fiber to the Curb optical network units.

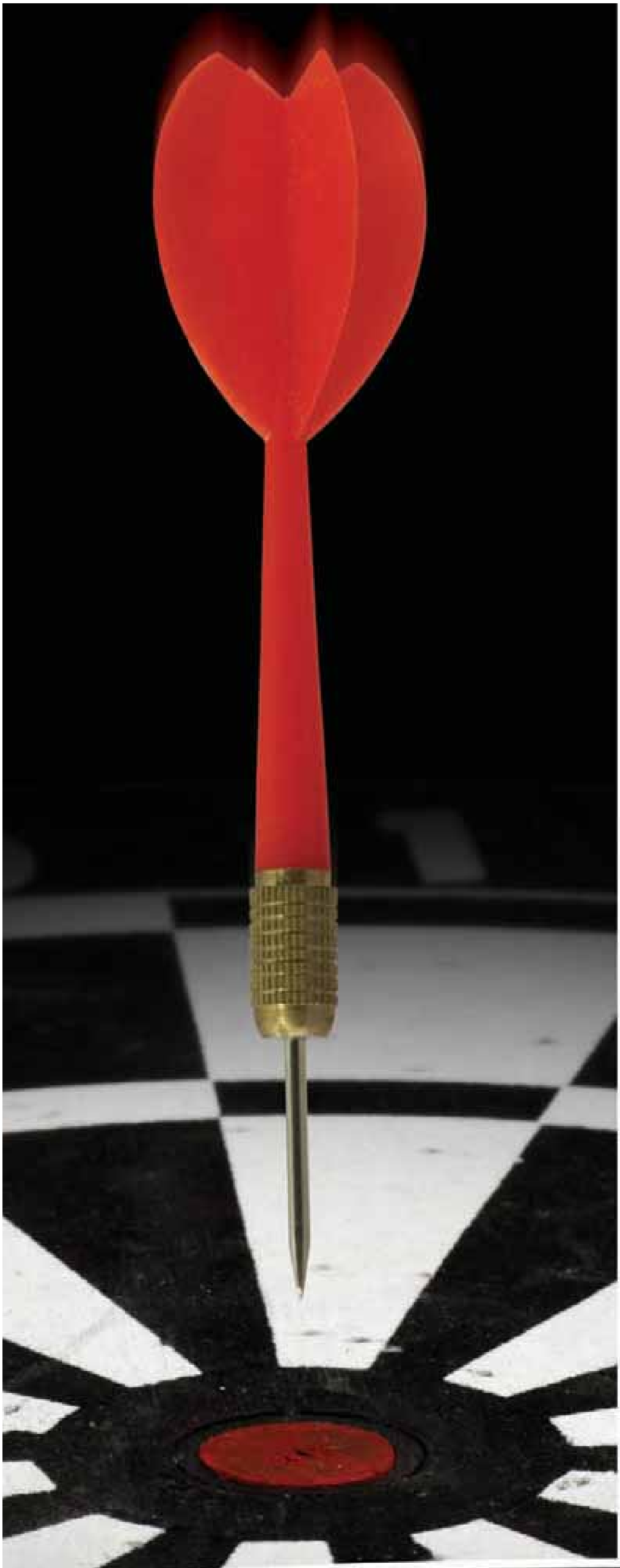
The foundation for the MicroTCA chassis is the MicroTCA Carrier Hub (MCH), which provides the switched fabric and shelf management functions. MicroTCA backplanes will provide scaleable bandwidth up to 40 Gbit/sec. Using the same serial transport mechanism as AdvancedMC, MicroTCA backplanes will provide a raw bandwidth of 12.5 Gbit/sec per channel while supporting star, dual-star, and mesh topologies. Like ATCA and AdvancedMC, MicroTCA is also protocol agnostic, and supports a variety of packet-based protocols, including Ethernet, PCI Express/AS, and Rapid I/O.

To enhance availability, MicroTCA shelves support hot-swappable AdvancedMC modules, enabling service providers to replace individual modules in the field without taking the entire shelf off line. The MicroTCA backplane also provides IPMI, which enables the shelf management to monitor and control each module installed in the backplane.

MicroTCA shelves will be able to accept any standard AdvancedMC module in a variety of form factors, including half-height/single-wide, half-height/double-wide, full-height/single-wide and full-height/double-wide. A typical high-availability shelf could combine redundant MCHs and power modules with up to 12 AdvancedMC modules. MicroTCA shelves will take power from an AC main or traditional -48 Vdc telecom source, and convert it to 12V for delivery to individual AdvancedMC modules.

CompactPCI based systems can still provide a good solution for many telecom projects. However, taken together, ATCA, AdvancedMC, and MicroTCA provide a modular, scaleable end-to-end framework that addresses the full spectrum of next generation, high-availability packet-based telecom applications, from core routers and WDMs, to converged customer premises equipment. This open framework helps drive equipment costs down by enabling equipment providers to quickly develop and configure systems using affordable, off-the-shelf hardware and software components. It also reduces operating costs, providing a modular, field replaceable framework with integrated system management that enables carriers to scale, manage, and service their systems with a higher degree of granularity. IT

Todd Wynia is vice president of marketing for Artesyn Communications Products. (news - alert) For more information, please visit the company online at http://www.artesyncp.com.



VoIP MARKETPLACE

VoIP MARKETPLACE

CEO Spotlight

Ron Bregman
Chief Executive Officer
Tadiran Telecom, Inc.

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 Ron Bregman, Chief Executive Officer of Tadiran Telecom, Inc. (news - alert)

GG: What is Tadiran Telecom's mission?

RB: Tadiran Telecom is an International producer of business communications solutions. Our tag line — "a world of communications for everyday business" — defines a commitment to providing real-world solutions for everyday use. We strive to continue to design and develop solutions that embrace today's diverse work environments.

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

RB: Tadiran designs and markets solutions for enterprises of all sizes. Our product offering is positioned with solutions that address the high-value market as well as the more application- or feature-oriented markets. Tadiran enjoys the position of being just the right size organization to be "big enough to pay attention to the smallest details." Big enough to have the resources needed, small enough to listen and react to customer on a more granular level than the market leaders. The focus of our newest solutions is no longer device-centric, but user-centric. And that is a significant departure from the PBX days.

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?

RB: We expect that the transition to SIP will open many doors for end users. The reseller or Partner community must be able to help customers enjoy the benefits that a standards-based solution offers. But the flexibility of diverse SIP endpoints can create implementation and maintenance challenges for both the reseller and end user. The roles and responsibilities of the resellers are different in the new generation.

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

RB: Tadiran continues to focus on communications solutions that provide a predictable experience for users. We are fortunate to have an R&D group that has both TDM (define - news - alert) and IP experience. Although IP is the obvious choice for our recent and upcoming products, the lessons learned in 40 years of telecom experience is paramount. Too many other IP products fall short of customer-expectations. Tadiran is committed to marketing products that meet and exceed expectations. Our session border controllers (our Sentinel products) have been a strong element in our success. We view the line as a good example of a product that makes the solution complete and the user-experience positive.

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

RB: Times are good right now and we are expecting more of the same. Our customer base is in a steady transition to IP. Tadiran's offerings allow existing PBXs to become telephony gateways controlled by a centralized softswitch. So our customers are moving from networks of switches to distributed solutions. This greatly simplifies the management of the facilities and decreases costs.

We are experiencing steady growth, most of which can be equated to VoIP upgrades and increase market penetration in the small and mid-markets. IT

Tadiran enjoys the position of being "big enough to pay attention to the smallest details."

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