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SIPQuest CEO Alain Mouttham on the Future of VoIP
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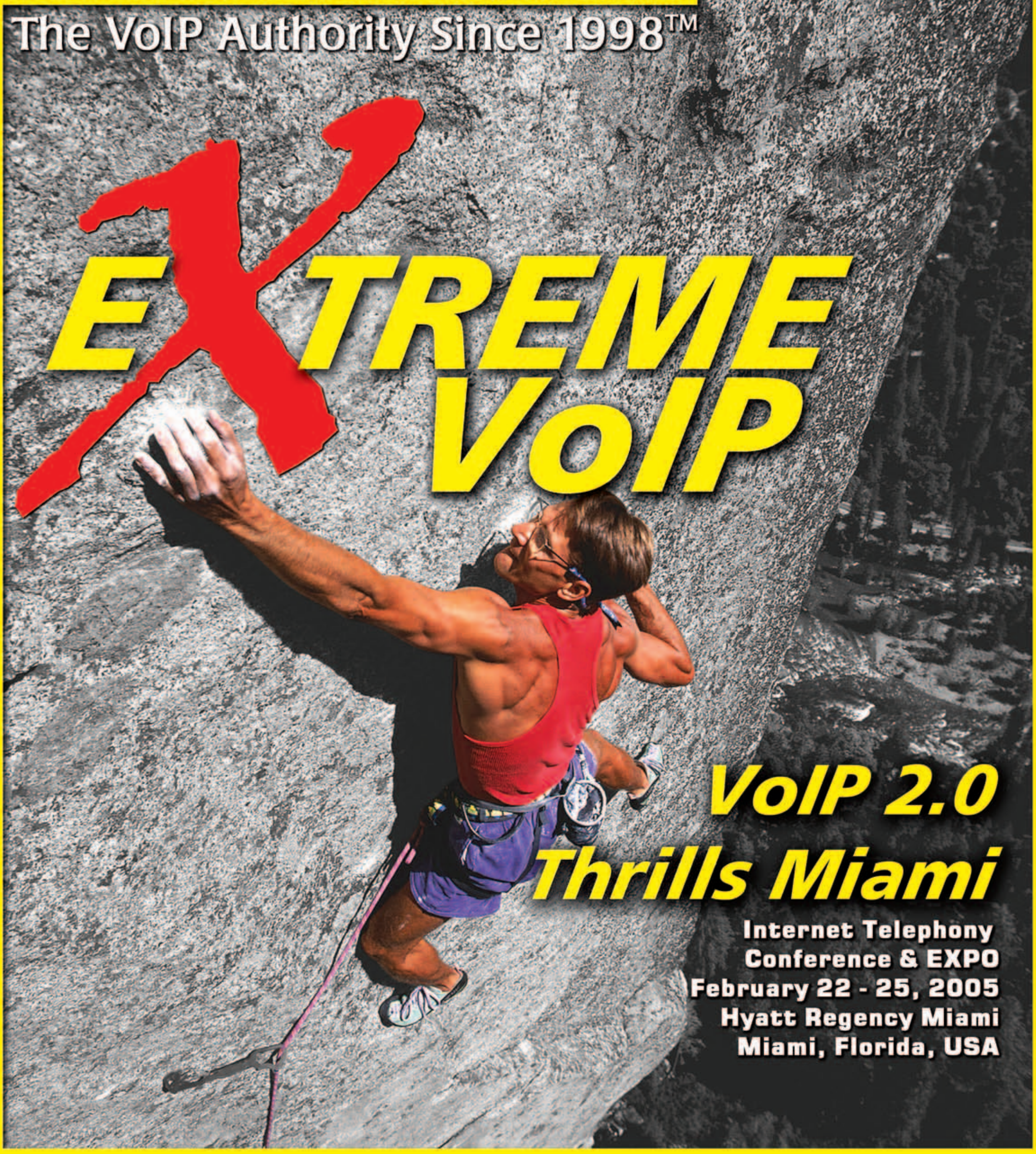

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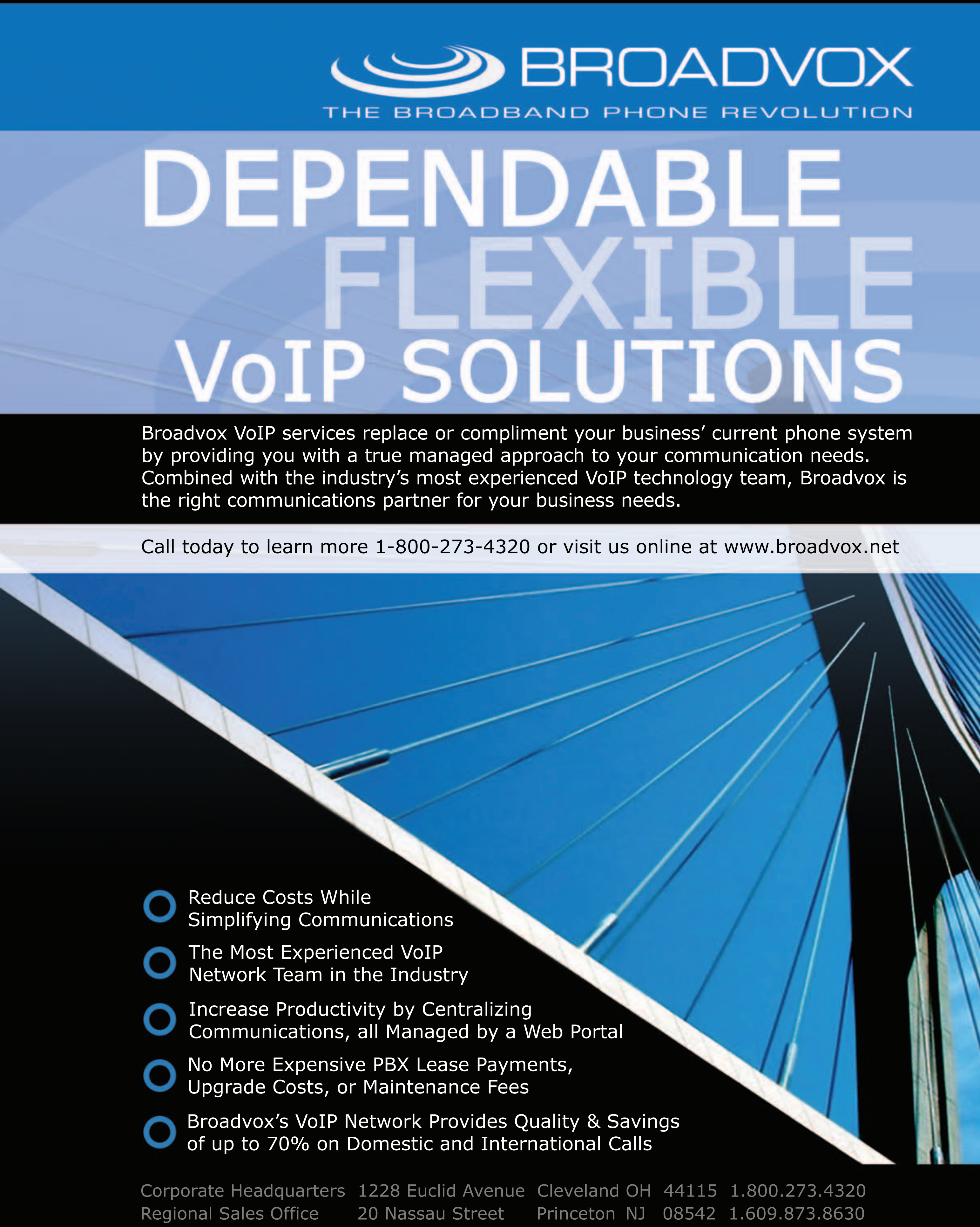
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The VoIP Authority

By Greg Galitzine

Stopping By VoIP On A Snowy Evening

IP is lovely, dark and deep.
But I have promises to keep,
And miles to go…

Who am I kidding?

I just returned from a jaunt through beautiful New England, making a number of stops along the road to speak with industry players in Connecticut and Massachusetts. Leave it to me to go visit friends on the Cape in December. Maybe it's just my conscience keeping me away from the temptations of Cape Cod in July or August. In any event nearly five inches of snow fell on Eastern Cape Cod during my visit. I was spared, but who needs that kind of weather lurking just around the bend?

While on the Cape, I had a chance to visit with Excel Switching. Excel was acquired by Lucent Technologies a few years back, and was spun back out of Lucent in June of 2003. Since then, Excel has reevaluated their position, formulated a plan, and is now excited about building momentum as they head into 2005. The company is doing some exciting things in the service provider space, and is busily promoting and expanding on their "Any-Gen" line of carrier-class service development platforms with advanced graphical service creation and management tools. Watch for the company to make overtures to the enterprise market by mid year.

A few miles onto the mainland, I stopped by to speak with Richard Koch, CEO of RNKVoIP, a division of parent RNK Telecom and a thriving CLEC serving multiple areas in New England, New York, and several other locations on the East coast. RNKVoIP recently unveiled a revolutionary new plan offering consumers a lifetime of unlimited phone service for a one-time cost of $999. RNK's Phone for Life plans will initially be available in Massachusetts, New Hampshire, New York, and Rhode Island, and come with a 60-day, money back pledge.

I had my reservations when I first heard of the Phone for Life plan, but after speaking with Koch, and getting a glimpse of RNK's vision, I'm fast becoming a fan of the idea. Apparently, according to Koch, so are a lot of people. He told me that subscriptions to the service are coming in fast and furious.

Subscribers to the RNK VoIP service enjoy unlimited calls to the U.S. and Canada, as well as 20 other countries and 21 additional international cities. The service will pay for itself in just under three years when compared to the $29.99 average cost of many VoIP service provider's plans. Since RNK is also a successful CLEC, they maintain the ability to port customers' existing phone numbers as well as the ability to offer traditional 911/E911 and 411 services.

RNK is also offering a business version of Phone for Life, for only $99 and a low, per-minute fee for calls.

Further on down the road, I dropped in to see our old friends at Pactolus. Ken Osowski, the company's VP of product marketing and product management shared his thoughts on the state of the industry, and the shape of his firm. I'm glad to report that both are looking up. He also shared some insight into Pactolus' new SIPware Broadband Telephony service. The service is a software solution designed for carriers deploying voice services (primary and secondary line) over broadband. Subscribers can make IP calls to other IP phones or to the PSTN, and receive calls through a PSTN-provisioned phone number. SIPware delivers line-side calling features such as Caller ID, call waiting, and the like in order to provide the subscriber with a similar experience to legacy PSTN-based phone services.

One of the most heartening thoughts that I came away with from this journey through New England is that in speaking with the people who are in the trenches every day — building and promoting and selling VoIP to the world — I get the real sense that good things are happening in the industry. One gets a very different view of the health of the VoIP market when visiting a vendor's office than, say, at a trade show. In the quiet thoughtful conversations that are undisturbed by the din of thousands of people milling about an exhibit hall floor, one sees the real drive and the desire of people to succeed at this game we call VoIP. Every so often, one has to get out of the office and hit the blacktop to really get a sense of where we as an industry are headed. Rest assured I feel good about the year and the road ahead.

Greg - *ggalitzine@tmcnet.com*

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

The Truth About Optimum Voice VoIP Service Pricing
Rich Tehrani believes that Vonage and CallVantage are crazy to engage in any cut-rate pricing because they have established themselves as premium brands. After all, BMW and Mercedes don't get into price wars.
http://www.tmcnet.com/68.1

Top Ten 2005 VoIP Predictions
TMC CTO Tom Keating offers his Top 10 VoIP predictions and ponderings for 2005.
http://www.tmcnet.com/69.1

Will We Need The Cone Of Silence On Airplanes?
Should authorities continue the ban on cell phones on planes? If you're looking for peace and quiet, it might not help much longer.
http://www.tmcnet.com/70.1

VoIP Providers: Think iPod, Not Wal-Mart
Smaller VoIP players will have to differentiate themselves by adding services more quickly than others to generate buzz and compete against colossal marketing budgets from the likes of AT&T and Vonage.
http://www.tmcnet.com/71.1

The Ultimate $25K VoIP Giveaway
Quintum Technologies, Inc., and TMC are joining together to sponsor this exciting contest to make someone's VoIP application come to life.
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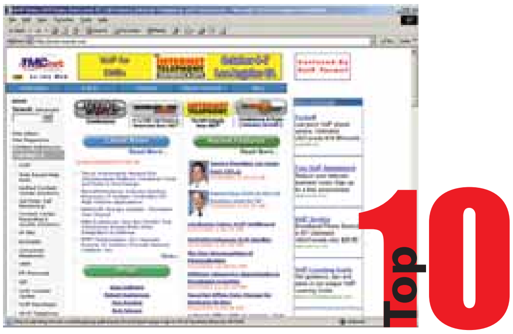

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3. Asia
4. Australia
5. Middle East
6. Northern Europe
7. Eastern Europe
8. South America
9. Southern Africa
10. Pacific Islands

QUOTE OF THE MONTH:

"The fundamental change in VoIP usage and public perception came in mid-2003 with the emergence of Vonage. This is the first time residential VoIP went mainstream in the Western world. (Softbank's Yahoo BB! in Japan was the first commercial, large scale, voice over broadband provider and now has over 4 million users). The Vonage model shows how small companies with limited facilities can offer residential services with PSTN quality at highly competitive costs."

– Ari Rabban

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Letters

I just received my first issue of Internet Telephony Magazine (November 2004) and the first thing I read was The VoIP Authority "Letter(s) To Myself." I thoroughly enjoyed the Greg-to-Greg exchange and before I read any further, I knew I would enjoy this publication.

Thanks for the laugh.....and for your dedication to providing honest value to your readers.

– Brenda S. Thompson
Corporate Manager
SBC Corporation
Richardson, Texas

I just finished reading the November cover story and just had to laugh. Let me begin by stating that I have not used the Peerio product and it may be the best thing to ever hit VoIP. I simply don't know. What struck funny, though, was Tom's commentary leading up to the interview.

In particular, the commentary states that the Peerio's design is "more than just an evolution — it's a revolution." So what is this revolutionary design? It's simply a key system! In fact, if you look at Figure 1 in the article and remove the data switch and router, you have a traditional key system. Key systems are neither new nor revolutionary. In fact, I suspect most large companies have grown from Centrex to a key system, and then to a PBX; and are now looking for something IP based. They may opt for an IP PBX or an IP Key System like the one described, but revolutionary? Hmm…

Again, the product may well be absolutely wonderful, but the concept is not new and is certainly not revolutionary. Thanks for listening.

– Rod Krause

Tom Keating responds:

While your analogy of Peerio simply being a key system has some merits, I think there's a need to look at the whole picture. First off, Peerio is IP-based where as key systems are not. Thus, managing Peerio phones is easier, and since it is IP-based it has some nifty CTI applications and has the potential to integrate with CRM applications more easily than a key system. Secondly, key systems require a centralized server, whereas Peerio is peer-to-peer. Thus, if you reboot the key system or it fails, the entire phone network goes down, whereas with Peerio, if you reboot your phone, only that phone goes down. Lastly, a company purchasing a key system has to purchase the whole "shebang" with "room for growth." So let's say it's a 10-person company; they will need to buy a key system that will grow to at least 30 people, whereas Peerio you can just buy 10 phones and then grow simply by adding more relatively inexpensive phones. The price tag of a 30-person key system will be much more than the 10 Peerio phones. In addition, a key system will eventually "max out" and you will have to throw the whole thing out and buy a new system. So I hope now you can see that a P2P phone system is indeed revolutionary over traditional centralized PBX/key systems.

I read the recent interview with Peerio's founder that you wrote and I thought it was amazing. If indeed this technology is as you describe it, then what a coup!

What do you think is the likelihood of mating Peerio and WiMAX in the near future for last mile deployment and for developing countries?

I enjoy reading your magazine and keep up the great work.

Thanks,
Mohamed

Thanks for the feedback. There is no technical reason why Peerio cannot be mated to WiMAX or any high-speed wireless connection. Not only will P2P VoIP technology be hot in 2005, but watch for Triple Play voice/video/data to take off in 2005 as well.

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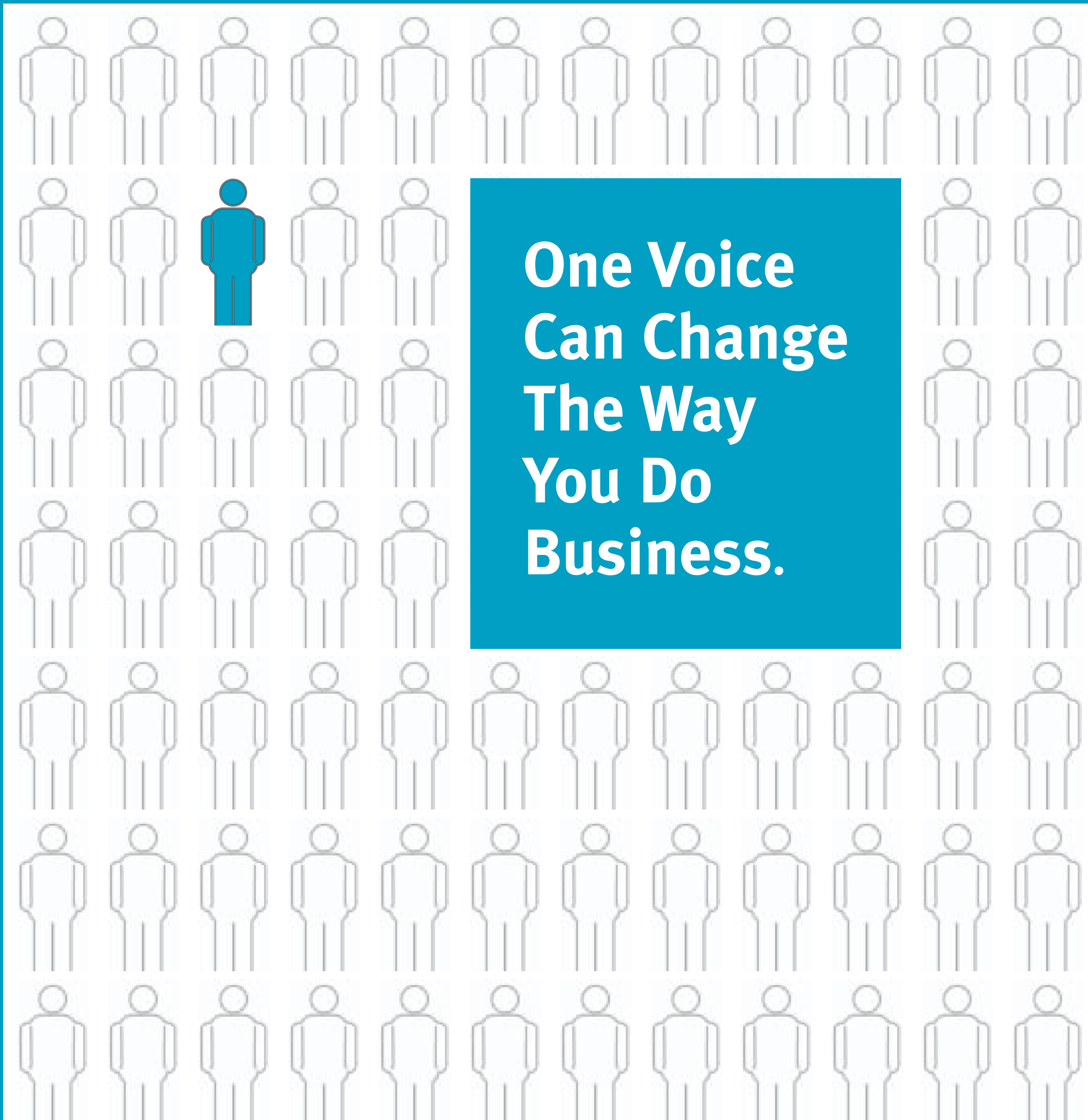

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The New Voice of Business

By Rich Tehrani

VoIP Predictions Revisited

About two years ago at Internet Telephony Conference & Expo in Miami, February 2003, I made some predictions about where the industry was going. I finally got to writing them in the May, 2003 issue of this magazine (http://www.tmcnet.com/73.1). At the time the market was still slow and in fact, by today's standards it was downright depressing. It wasn't until the last quarter of 2003 that things started to slowly heat up. Lots of people are making predictions about the future these days, so I decided to go back see how my predictions fared.

10) Hosted Communications (define - news - alert)

I was right on as far as I can tell. The market is doing great. Lots of companies like M5 Networks and others are doing well, in fact they are thriving in this space. This trend will continue.

9) IP Centrex (define - news - alert)

Why I decided this was its own number I am not sure except for the fact that I lumped the conferencing vendors such as WebEx above. Really these two categories are here for the long haul and will continue to grow. The fact that AT&T CallVantage (quote - news - alert) now charges for the conferencing features that were once free tells me there is lots of demand for conferencing services.

8) Linux Telephony

This is still hot. Asterisk and Pingtel are among the leaders and they will continue to gain traction and new customers. I probably should have used open-source telephony as the heading here.

7) Video Conferencing

This one is on hold. What I said last time is still true:

Let's face it, video conferencing has been "right around the corner" for 15 years. But we may finally really be there. IP telephony reduces the cost of video conferencing and better yet, cameras are being embedded in tablets and PDAs. Finally, most of what we want to show others isn't at our desks anyway so it makes sense that mobile computing will be the driver for video conferencing. I firmly believe that we will see a merging of consumer electronic video equipment with WiFi networks resulting in video conferences on-the-fly from theme parks and other WiFi-enabled locations.

6) Telecommuting

This is happening in a big way. Companies like West Corporation - one of the world's largest call center outsourcers - are using home agents to compete with the offshoring threat from India. Many think this is the way to battle low cost offshoring as once you factor in the risk of moving overseas as well as the other problems, you may as well try something like this. There is no reason that this concept can't be applied globally. The only requirement is to make sure your employees have broadband.

5) Cable/DSL telephony

Bang. Nail hit on the head. We had a keynote speaker from Vonage at this same event and as I recall they were a sponsor. This is back when no one had ever heard of them.

4) Enhanced IM

What I said still rings true today. IM clients are morphing into personal communications managers. First they began to link you to e-mail accounts and then telephony. Expect this trend to continue and we may see these clients becoming our primary communications portal.

3) SIP (define - news - alert)

SIP gets hotter every day. It is an essential component of VoIP 2.0 (www.voip20.com).

2) WiFi Telephony (define - news - alert)

Still hot, still happening. In fact, recently we ran news regarding WiFi telephony on commercial flights via an announcement between Cirilium Holdings and Honeywell (tmcnet.com/74.1). Wireless VoIP will be pervasive soon.

1) IP PBX (define - news - alert)

I said that 2003 was the year of the IP PBX and I think in hindsight was right.

Rather than make any new predictions or update the list I thought it worthwhile to comment on Tom Keating's Blog (Tom is TMC's CTO and executive technology editor). If you search VoIP blog on any search engine (I tried the top three) Tom ranks number one on all of them and has been at the top consistently for months. He seems to have the best-read blog out there as far as I can tell, so I decided to use his predictions and comment on them. (My comments appear in blue.)

1) VoIP providers will continue to run to the FCC (a VoIP proponent) for protection from the "big bad bully" RBOCs, ILECs, CLECs, etc., as they try and lobby Congress to regulate VoIP. It will be a fun battle to watch.

Agreed but I am not so sure that the FCC is a proponent of VoIP. Although the FCC says it is pro VoIP, the regulatory environment is in flux and makes for a turbulent time if you are looking to fund a business in this space. We need more clarity from the government on VoIP. We need to put this issue to bed.

2) VoIP providers will continue to harp that the government shouldn't impose any regulations on VoIP and that the industry should be open and free, while simultaneously VoIP providers will continue to alienate their customers by password-protecting and locking the customer's ATA (analog telephony adaptor), thus preventing customers from easily switching to another VoIP provider and using the same ATA. This is hypocrisy at its worst! Customers will continue

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You just bought your entire company
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That's because its high-fidelity voice quality is not only better than competitors, it's better than traditional phones. In fact, independent tests by CT Labs and the Tolly Group give the phones top scores for "excellent speech quality" and optimal flexibility with "greater management functionality than rival products tested."
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to be left with useless ATA "bricks" which eventually will make it the local landfill when they switch to a better VoIP provider.

This isn't unlike what we see in the cell phone world. If a provider subsidizes equipment they expect to make sure you use that equipment for their service only.

3) With millions of customers using VoIP and with the ability to now easily switch to another voice provider and keep your existing phone number, more customers will switch to the best value, which means more ATAs will make it to the landfill drawing attention from the EPA. Rather than let the EPA regulate recycling of ATAs, in 2005 VoIP providers will offer a rebate or discount to "turn in" your old ATA so they can recycle it. It can even be a selling point to get you to switch from a competitor: "Be green! Send us your old VoIP ATA and we will give you the first month for free!"

Agreed. This is the same reason I have two old cell phones on my desk that one day will find themselves listed on eBay.

4) 2005 - The Year Triple Play takes off.
Other than WiMAX (define - news - alert), the Triple Play has got to be one of the most hyped technologies of 2004. Well, watch out in 2005. The Triple Play will take off in 2005, you can bank on it. I examined one Triple Play technology provider (Pannaway) in the labs recently and the technology is ready. The technology used by them is ADSL2+, targeting the DSL providers (typically phone companies). This company already has some actual deployments - not trials - across the country.

Pannaway has really incredible technology. I think it may take one more year to get the marketing right and for the general media to understand what is happening. This doesn't take into account the fact that lots of fiber is being laid down and that takes time. This prediction will take place in mid 2006-2007.

5) Return of the Jedi (Return of telemarketing calls to switch providers).
Remember the days when MCI (quote - news - alert), AT&T (quote - news - alert), etc. would call you at home and ask you to switch phone carriers and they'd often bribe you with $50 or even $100? Have you noticed that the volume of these calls has dramatically gone down? Want to know why? It just costs too much money for the carriers to pay a call center agent to call you and get you to switch. The conversion rate isn't that great to begin with and with the ROI going way down with the price of voice minutes tanking, it just doesn't make sense. Of course the Federal Do Not Call list could have something to do with the call volume drop as well. But does this mean the end of telemarketers trying to get you to switch?

Unfortunately, I don't think so. There is a loophole in the DNC that lets companies call you if they have done business with you in the past six months, which surely will be exploited. As I mentioned in my prediction #4, the phone companies will soon offer Triple Play voice/video/data. If the phone companies don't already have you as a DSL customer, they could in the near future have you as an ADSL2+ TV customer. If they have you as a customer in ANY of the Triple Play offerings, they can call you and upsell you on the other services. So if you are one of the millions of DSL users, watch out in 2005! Your DSL provider WILL BE calling you to offer you TV access bundled with voice and/or data.

This is a huge competitive advantage for the "big boys" to go after

tel(x) Atlanta Opens at 56 Marietta Street

December 3, 2004 was a momentous occasion for me as it was the first ribbon cutting ceremony I have ever attended. tel(x) opened their new facility, tel(x) Atlanta, at 56 Marietta St. in Atlanta, GA, an address that was formerly the site of a carrier hotel. Atlanta is a major interconnection point from Florida to NY to South America and 56 Marietta sits on top of most of this fiber. This facility nicely complements tel(x) New York. The company's CEO Rory Cutaia cut the ribbon with the assistance of his team. This took place just after his charismatic speech, which was followed by speakers from the local area.

After the ribbon cutting ceremony, the crowd went to the Omni hotel where we had a day of networking and educational sessions. Many universities and some corporate delegates were in the audience.

I learned that this Atlanta property is important as it relates to educational networks, Internet 2 and much more. VoIP is an important part of the company's future. In fact, Hunter Newby, the Chief Strategy Officer of tel(x), mentioned as part of his presentation that a new voice Internet is being built. As with other technologies such as Internet adoption, we are seeing this trend starting in universities and subsequently growing outward to enterprises. This voice Internet is in fact a private Layer 2 VoIP interconnection network. VoIP interconnection networks are being connected much the same way the Internet came into being.

People at the conference asked me what tel(x) will do for their local market and I explained that peering has the potential to fundamentally change the way VoIP works. Peering of networks coupled with free ENUM services from companies like Stealth Communications allow tremendous savings on calls from parties that are peered.

Universities, enterprises, and small and eventually larger service providers will all need to peer to save money and increase call quality. Ironically, today a Vonage to CallVantage call still travels over the PSTN. If these carriers peered, there would be no PSTN connection, resulting in fewer delays, better quality, lower cost, and an escape from myriad government regulations, taxes and fees (at least for now).

The momentum is building in the peering space and I get frequent calls from all types of service providers asking me about the world's first VoIP Peering Summit taking place in Miami, Florida at TMC's Internet Telephony Conference & EXPO, February 22-25, 2005. Peering will dramatically affect service providers, large enterprises, call centers, the government and universities. If you fit in any of these categories you need to be at this event.

One sure sign of success of this summit is the strong sponsorship support we have received from three of the leaders in the peering space, tel(x), Terremark, and Sansay. The latter, Sansay is a relatively new session border controller company founded by the highly successful management team from Nuera, one of the original VoIP gateway providers going back to the late 1990s. IT

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Softphone too soft?

Clear) Simple) Durable

www.gnnetcom.com

GN 8120 USB

It's nice to have something to hold onto. The GN 8120 USB delivers telephone-like simplicity to the VoIP softphone experience.

The GN 8120 USB combines a digital USB-to-headset adapter for superior voice quality with inline softphone controls, enabling easy access to the most common calling functions without a keyboard or mouse.

- Enhanced digital audio quality for VoIP applications
- Buttons at your fingertips for easy access to the most common call functions
- Visual LEDs quickly alerts you to incoming calls or when new voice mail is waiting
- Compatible with any GN Netcom commercial-grade headset

To learn more about GN Netcom headsets and VoIP products, visit us at **www.gnnetcom.com** or call us at **800.826.4656**.

Avaya COMPLIANT TESTED DeveloperConnection Program

GN Netcom

Vonage, which has cut into the carrier's market share. I suppose the Triple Play offering is one way of striking revenge against Vonage and the other Internet phone providers.

You are right Tom. Good point.

6) The Empire Strikes Back.
Phone companies (The Empire) will go after the cable companies' TV business just as the cable companies have successfully gone after the phone companies' voice and data business.

Agreed. This is a foregone conclusion. In fact the first moves in this battle are starting to take place already.

7) Colleges ramp up on VoIP
When I was in college they just added Ethernet to the dorm rooms. Too bad VoIP didn't exist back then as I often had $100 phone bills. Most colleges built their own phone system acting as their own little phone companies so they could charge students a "premium" and rake in the profits. Well, the colleges are really starting to hurt with cell phone market penetration as well as students using Skype and other VoIP solutions. Many colleges spent millions on their legacy phone systems and haven't recouped that investment. Well, if you can't beat em' join em'! Many colleges have already started deploying VoIP, often giving the students a Cisco IP phone or other IP phone to use. You can expect more of this in 2005. Fortunately, the easier administration (as compared to traditional PBX/phone systems) as well as the ability to partner with less expensive VoIP termination providers such as Level3 could make the colleges more competitive and with good margins.

Every university should have WiFi (define - news - alert) or WiMAX coverage. If they don't there is something wrong with them. Wireless VoIP is what students will use. All cell phones will have WiMAX and WiFi radios built in.

8) Cities become their own phone companies
You will start to see more cities not only offering high-speed wireless broadband using WiMAX and other high-speed wireless technologies, but you will start to see cities offering their own phone services as well. Just think of the loyalty they can build! If I have my choice between paying a private VoIP company based in New Jersey versus paying my local town, heck I'll give the money to my local town. My local town can simply send out an ad in one of those "coupon mailers" that most of us in the country receive and say, "Use us as your phone company and your property taxes will go down." SOLD! End of story. I'd drop my current VoIP provider in a heartbeat! I'd tell my neighbors to join so we could reduce our property taxes and then in turn would tell other neighbors in the town. The old "peer-to-peer" system if you will!

I predict that if cities wise up and become their own phone companies, this could be among the most revolutionary changes in the telecom industry ever. Instead of a few dozen phone companies you could have thousands of phone companies - with each town being its own phone company.

Pennsylvania recently announced it was rolling out WiFi service in parts of the state and Verizon via its lobbying efforts has put itself in a position to actually veto the roll-out of such technology throughout the state. You can't blame Verizon as it makes its living by providing paid broadband service and our political system allows well-funded individuals and corporations to change government policy in a way that is not in the public's interest. Companies like Verizon will fight state competition tooth and

TMCnet: A VoIP Community Expands

Thanks to all of you that have been going to our Web site, TMCnet.com. Our traffic has swelled and we now receive more Web traffic on average than any other VoIP or communications site. I recently found an independent Web site called Alexa.com that tracks this sort of thing. (Alexa is a division of Amazon.com.) According to Alexa, TMCnet.com is in the top 12,500 sites in the world. Yahoo! is #1. Avaya is just under 16,000. Skype is around 1,000, catapulting from nothing to 6,000 in about a week according to their graph. Vonage is around 1,400, Sonus is around 200,000 and Polycom is around 47,000.

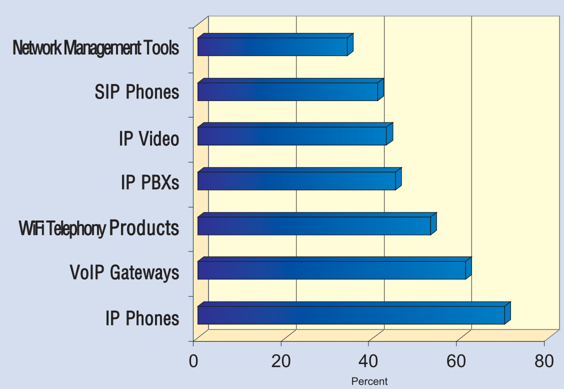
Alexa.com actually allows you to compare sites, such as TMCnet.com, to any other site you choose. Have fun. If you find a competitor that has a better ranking than we do, I'll be very interested to know about them so please drop me line.

Survey
Recently we decided to survey some of our Web visitors, including the Internet Telephony magazine online readers, and it is amazing to see how much purchasing is going on among this community. Here are the stats.

87% of the visitors are personally involved in the purchase of VoIP products.

83% plan on purchasing products and services in the next 12 months.

These are the actual products and services that will be purchased:



I didn't expect the WiFi telephony or video numbers to be quite so high but this is definitely great news for the industry. This survey should definitely foreshadow another year of rapid growth and excitement in VoIP. IT nail. So far Verizon is winning this battle.

9) VoIP Spam + 1st VoIP spam lawsuit

2005 will mark the first really bad VoIP spam incident. Often referred to as SPIT (Spam Internet Telephony), I predict someone looking for a quick buck will send automated recorded messages (.WAV file) to thousands of SIP addresses. If the VoIP call is IP-to-IP and never touches the PSTN, the stringent laws governing the PSTN won't apply. The first lawsuit will ensure, and the spammer will win since VoIP is still classified as an "information service" not bound by the Federal Do-Not-Call rules. The DNC law will be amended as a result. Perish the thought, but the FCC may be forced to reclassify all VoIP calls (IP-to-IP, IP-to-PSTN, PSTN-to-IP) not as an "information service" but as a telecom service bound by all telecom regulation. It's a scary thought and not necessarily a prediction.

This is a rather thought provoking prediction. I'm curious to see how this will play out.

10) Microsoft tries its hand again at VoIP

Let's recap: Microsoft develops NetMeeting, which has VoIP and video capabilities, but doesn't really capture the imagination of the market. Microsoft launches MSN Messenger with VoIP features and video, and although many people use MSN Messenger, it's primarily used for instant messages. Yet another Microsoft VoIP failure. Next, in late 2004, Microsoft pushed Live Communications Server (LCS), which boasts VoIP with SIP capabilities and collaboration features. It's a good product, but very complex to install and requires integration with Active Directory.

I foresee Microsoft taking another shot at the VoIP market. I predict Longhorn, which has had its deadline pushed back several times will have some nifty VoIP features.

I should point out that Microsoft does have one VoIP success - the XBOX Live service, which uses Level3's backbone and which lets gamers talk trash over the Internet as they frag one of their buddies.

Another point to ponder… SBC and Microsoft recently announced a $400 million, 10-year agreement that calls for Microsoft to provide the software SBC will use to provide television services to U.S. consumers. The technology will let the company deliver 1,000 or more new TV channels, far more than cable-TV providers currently offer. But could this be Microsoft's Trojan horse? What's to prevent Microsoft from using the experience gained and then launching their own Triple Play offering? Of course, Microsoft doesn't own any last-mile copper connections to the home, but with Microsoft's cash they could buy the last mile or simply deploy high-speed wireless.

This seems right on. All people in the industry I speak with are quietly scared about what Microsoft has up its diamond-cufflink-adorned sleeves. Microsoft could kill competition in this market overnight. This is not unprecedented. They destroyed the VoIP client market back in the late 1990s. VocalTec had a successful client that they charged money for and Microsoft launched NetMeeting, a direct competitor with more features and gave it away for free. I think this single act could have retarded VoIP development and profits by three years or more. By embedding a strong VoIP client in IM, they could do tremendous damage to Skype's growth curve as well.

What is amazing to me is that while ILECs battle CLECs by putting up regulatory roadblocks, they don't realize that Skype and Microsoft could easily take away the market they are so desperately trying to protect. Sure they own the pipes but if power line access, WiMAX, and other new technologies start to proliferate, they will have real trouble on their hands. IT

VoIP: My How You've Grown

Internet Telephony Conference and EXPO is a perfect barometer for the VoIP market. This is the third show in a row where we will sell out the exhibit hall. In fact, we recently had to go back to the hotel where the show is being held in Miami and plead for more conference and exhibit space. The hall is now well over 100 booths and as usual, many of the exhibitors aren't at any other show.

Many people have been asking me about VoIP 2.0 and how this event will differ from others. The reality is that VoIP is more mature now than ever. We don't ask "why" anymore. We all know we need VoIP. What we ask is "What's next?" There are so many technologies such as peering, p2p, open source, triple play and others that are so crucial and have the potential to fundamentally change the landscape of communications. **Internet Telephony** remains dedicated to covering these exciting new developments in our market.

People tell me Skype is disruptive. I say that is hogwash. VoIP is disruptive. Skype is just a great service leveraging VoIP. Others ask me when the killer app will emerge. To them I say please read Marc Robin's excellent Mindshare 2.0 column in the December issue of this magazine (http://www.tmcnet.com/75.1) where he tells the industry to wake up and see the killer app is here, under your nose.

What amazes me most about this year's event is how early the hotel is filling up. It will sell out sooner than ever so please book your space early so you don't have to commute. Being a Miami-based event we naturally attract a good portion of the U.S. and Latin America. What amazes me most about this year's show is how early the international crowd is showing up. We have exhibitors from countries that never participated in this event before and conferees are trekking from countries like Hungary, Hong Kong, Ukraine, Singapore, Jamaica, Canada, India, Austria, Egypt, Mauritius, UK, Cameroon, Bangladesh, UAE, Nigeria, and Saudi Arabia.

This is now our 11th Internet Telephony Conference and we consistently hear from past conferees that we have the best conference program out there. We don't recycle the same self-promotional speakers that are friends or business partners of the conference chairman who subsequently give you a "perspective" read (sales-pitch). We have great speakers that are there to educate you and we don't invite the "sales people" back. This is why our VoIP show has a satisfaction guarantee. You don't have to come to our conference but if you go to a competing one, please ask for a guarantee in writing before you spend a dime.

I want to wish you all a wonderful new year and tremendous success in all of your endeavors. From our vantage point everything in VoIP is growing as quickly as ever and if you are in this space, you are in the right place at the right time. Hope to see you in the sun this February in Miami. IT

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Polycom Delivers SoundStation2

Polycom, Inc., provider of unified collaborative communications solutions, delivered the SoundStation2, the next generation of its line of triangular-shaped SoundStation conference phones. The SoundStation2 has twice the loudness and 50 percent better microphone pick-up than the original SoundStation enabling high-quality conferencing for everyone on the conference call.

Since launching the original SoundStation in 1992, Polycom has shipped more than 1.7 million units, received numerous industry awards for the original's breakthrough quality and patented design. The newest edition, the SoundStation2, delivers improved voice quality, provides the ability to talk up to ten feet away in a normal voice and still be heard clearly, and adds new features including a backlit LCD display, international Caller-ID, an address book for speed dialing and a cell phone connector.

"Polycom defined voice conferencing quality with its SoundStation line and is setting the bar higher with the new SoundStation2," said Roopam Jain, multimedia/conferencing and collaboration group senior industry analyst with Frost & Sullivan. "The advancements in Polycom's microphone technology ensure a dramatic improvement in the quality and clarity of a call and make voice conference calls much more natural, like being in the same room with the people on the other end."

"The SoundStation2 is unlike any 'speaker phone' I have ever used," said Jim Wolf, I.S. project manager at Whirlpool Corp. "After using the phone for a few days, I found that I couldn't live without it and carried it between home and the office so I could use it for calls after normal work hours. The voice quality is superb and we no longer have to repeat ourselves during a call, which makes meetings run much smoother. I would highly recommend the Polycom SoundStation2 conference phone for every conference room and office."

The SoundStation2 is the latest in a line of next-generation conference phones from Polycom, which also includes the SoundStation2W, the first wireless SoundStation conference phone; the SoundStation IP 3000 standards-based VoIP conference phone; and the SoundStation VTX 1000 wideband conference phone with high fidelity sound and microphone capabilities.

The SoundStation2 is now available worldwide through Polycom channel partners for a manufacturer's suggested retail price starting at $599 (North America).

http://www.polycom.com

Alcatel Launches K-12 Education Initiative

Alcatel is bringing communications ‘Best Practices’ to the K-12 education market with the launch of its K-12 education initiative. Alcatel’s focus is on delivering communication applications that help school officials share knowledge more effectively to create a supportive and safe environment for children.

As a provider of network infrastructure and Internet Protocol (IP) Telephony products for enterprises, Alcatel is providing a corporate-like voice solution to both large and small school districts. The Alcatel solution enhances the student and teacher experience and creates a better educational environment that includes connecting school and community officials, improving security, increasing parental involvement, and reducing operations costs.

Alcatel, working closely with educators and school districts, has been able to create and deliver a solution set of K-12 specific applications designed to increase security, improve student database management, perform environmental monitoring and facilitate improved collaboration between distributed and mobile workforces. These services have been made possible through a combination of the open standards design (XML, SIP, Webservices) of Alcatel’s OmniPCX Enterprise communication server, the expertise of Alcatel’s Professional Service Group, and the capabilities offered by independent software vendors and XML partners. These include companies such as Service Objects, Status Solutions, StrikeIron, Xtend and others. The OmniPCX Enterprise can also link technology and facilities operations to deliver real-time management capabilities through the phone system.

Among Alcatel’s recent successes, the School District of Cheltenham Township in Pennsylvania was recognized for its education expansion and security programs. Using the Alcatel OmniPCX Enterprise, Cheltenham is the first school district to implement Alcatel’s K-12 education district alert application, which works to simultaneously send each classroom district wide instant notifications regarding weather, news, events or specific security concerns.
http://www.alcatel.com

AltiGen Integrates IP Telephony For Citizens Bank

AltiGen Communications, Inc., a manufacturer of VoIP telephone systems and VoIP contact center solutions, announced it integrated IP telephony solutions for Kansas-based Citizens Bank, N.A. to improve customer-focused communications across multiple locations and to maximize employee productivity. Citizens Bank, N.A implemented an AltiGen AltiServ VoIP telephony solution to connect 10 disparate locations and reports a “night and day difference” since shifting from outdated telephone systems to an integrated AltiGen VoIP telephony solution.

Citizens Bank chose AltiGen citing increased efficiency, ease of administration, reliability and cost savings as the foundation decision criteria. For Citizens Bank, calls and call transfers from one location to another use simple four-digit extension dialing. This makes employee communications more efficient and enables customers needing service to easily reach the most appropriate bank employee, on the first call, in any location, no matter what branch was initially dialed.

Citizens Bank has also reduced its dependency on expensive, individual telephone lines while dramatically reducing long distance charges. In addition, the bank has begun enjoying the benefits of built-in system capabilities not originally rated as priority features. For example, Citizens Bank now looks to utilize more advanced call routing and workgroup support capabilities like built-in call recording and monitoring to improve customer service. The bank is also pleased with the advanced capabilities like E-911 support for emergency calls. AltiGen has also enabled businesses like Citizens Bank to manage their entire communication system themselves, from a single location.
http://www.citizensbankna.com
http://www.altigen.com

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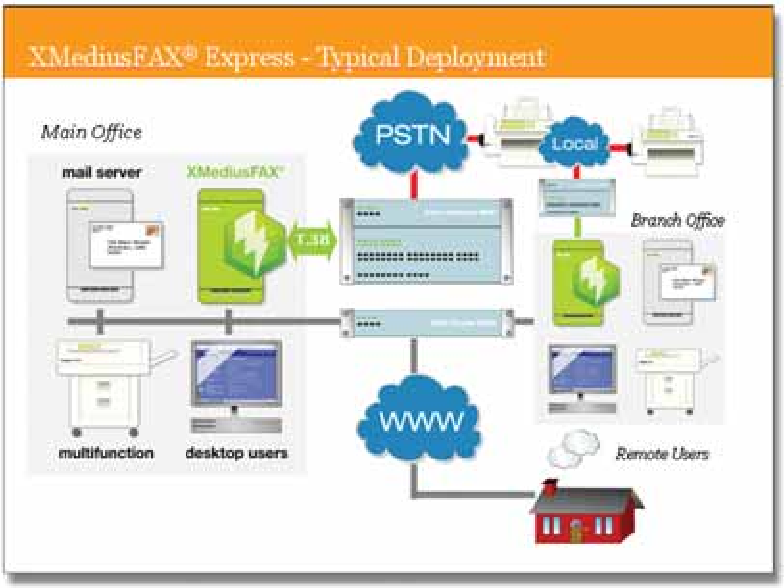
Interstar Launches XMediusFAX Express

Interstar Technologies Inc., a developer of enterprise fax solutions for converged IP networks, announced it launched XMediusFAX Express. The product is a more cost-effective and efficient option for customers requiring a boardless T.38 Fax over IP (FoIP) server solution with only one to four channels for simultaneous fax transmission.

XMediusFAX Express allows lower-traffic fax users or small to medium size businesses (SMBs) to incorporate fax into their existing VoIP infrastructure, seamlessly integrating with IP PBX and voice mail systems. XMediusFAX Express offers controlled, centralized faxing from the desktop or a multifunction device (MFD), as well as automatic inbound routing to boost productivity by 90 percent and reduce costs associated with traditional faxing.



XMediusFAX Express, version 4.1 is boardless, T.38 FoIP software-based product, XMediusFAX lowers communications and network management costs by at least 40 percent, since expensive fax hardware and leased lines are no longer required. It is fast, efficient, secure and simple to use. Faxes are routed to the VoIP branch office closest to the destination area code via Least Cost Routing and Intelligent Number Recognition. Long-distance charges are virtually eliminated. The new Express version provides these same benefits, but in a package that is designed for quick and easy deployment.

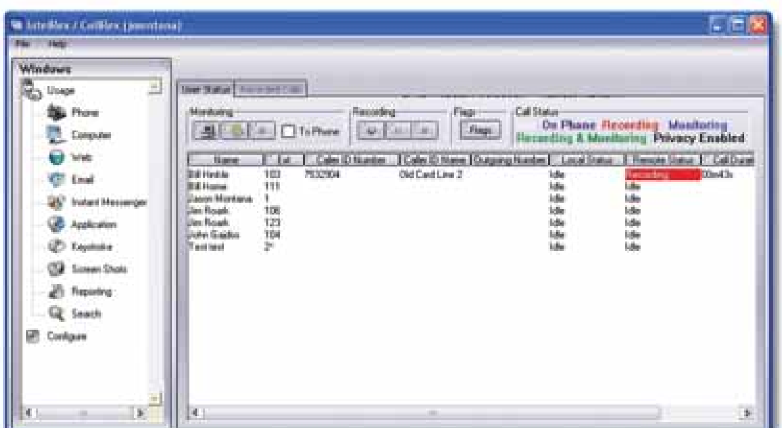
XMediusFAX Express is ideal for SMBs with less than 100 users, and requiring no more than four ports for simultaneous transmission. Assisting organizations with regulatory compliance (i.e. HIPAA), it routes sensitive documents in TIFF or PDF format via the security of VoIP and VPN encryption directly to approved recipients, and designating secure printers for any required hard copies. The product supports 3Com, Cisco and Alcatel gateways, as well as Microsoft Exchange/Windows/Office 2003.

http://www.faxserver.com

Telrex Announces Version 3.0 Of CallRex

Telrex, a developer of VoIP call recording and monitoring software for small and medium-sized businesses with IP or IP enabled telephone systems today announced the release of CallRex Version 3.0.

CallRex Version 3.0 includes many new and unique features including; look-back call recording, support for Citrix terminal services, the ability to export multiple calls, improved trigger filtering, automatic deletion of recorded calls according to pre-defined criteria, remote polling of recorded phone calls, improved recording quality, re-start capability for remote data collectors, improved call compression, improved record-on-demand capabilities, the ability to automatically receive software updates online and many other additional features.



"The newest version of CallRex's reinforces its position as the leading VoIP call recording and monitoring software solution." said Robert Kapela, product manager at Telrex "CallRex is entirely software based, thus it is 50-60 percent less expensive than legacy-based solutions that require expensive proprietary telephony cards and are complex to install" said Kapela.

CallRex works with more IP PBXs than any other vendor and currently supports; 3Com, Mitel, Avaya, Cisco, Shortel, Nortel, Siemens, NEC, Zulty, Artisoft, and many softswitches and gateways. CallRex is sold through a network of resellers.

http://www.telrex.com

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SunRocket Selects Telco Systems' Access211

Telco Systems announced that VoIP service provider SunRocket selected the Access211 VoIP broadband Internet Gateway to be the official SunRocket plug-and-play Gizmo, which enables subscribers to make and receive phone calls and faxes over broadband Internet connections using standard telephones and fax machines.

Some of the Access211 features and capabilities absent on other gateways that prompted SunRocket to select the Access211 include distinctive ringing; Internet faxing; two telephone ports that concurrently support advanced compression technology for two different phone numbers; integrated routing, which eliminates the need for an external router; and advanced Quality of Service features such as traffic shaping, which contributes to the superior voice quality of the Access211.

SunRocket selected the innovative Access211 gateway and now includes the device in the self- installation kit that SunRocket provides to its customers when they sign up. Customers simply connect the Access211 between a DSL or cable modem and a personal computer, plug a standard telephone and fax machine into the Access211, and within ten minutes the customer is ready for Internet phone calling.

SunRocket has integrated technology from VoIP service providers to deliver a simple-to-use home phone service, complete with a full set of built-in features. Priced at $24.95 per month or $199 for a full year, with no additional charges for service activation, equipment, taxes or shipping, SunRocket is bringing the power of Internet phone service to homes across America, with no hidden surprises.

http://www.sunrocket.com
http://www.telco.com

InfiniRoute Networks Selects tel(x) For VoIP Peering

InfiniRoute Networks, Inc., announced that InfiniRoute Networks will work with tel(x), an operator of telecom "meet-me" interconnection facilities, to provide a fully-managed VoIP Peering facility, hosting InfiniRoute's Managed VoIP Peering (MVP) service solution at tel(x)'s New York interconnection site.

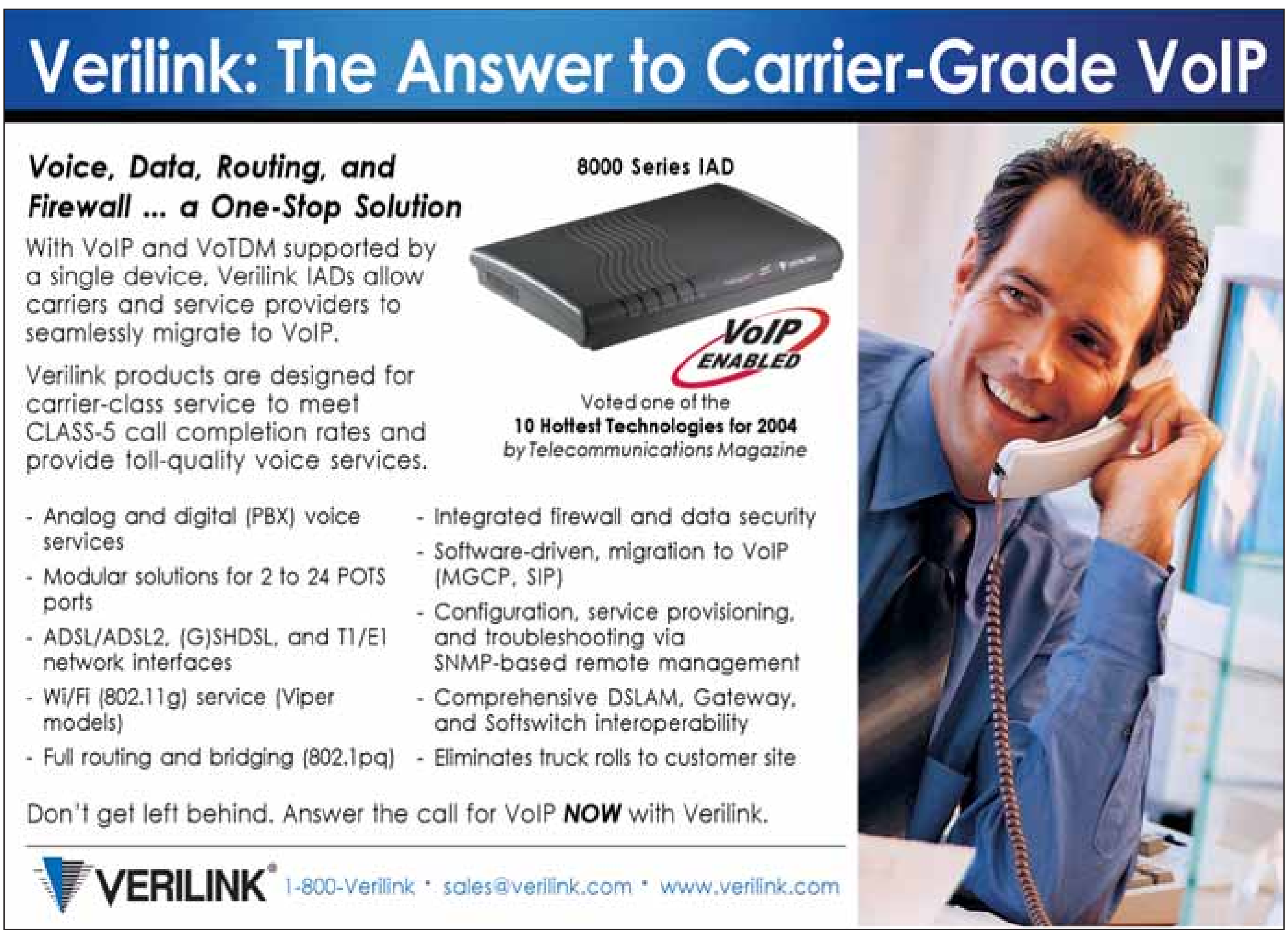
"The New York tel(x) facility is an ideal cornerstone upon which to build the core of InfiniRoute's VoIP Peering eXchange (VPX)," said Robert H. Turner, CEO for tel(x). "Using tel (x)'s diverse connectivity options, InfiniRoute can offer global carriers immediate access to high-quality VoIP-based interconnections to sites around the world."

InfiniRoute has numerous Tier 1 international carriers depending on the company's services for their global interconnections. InfiniRoute Networks' MVP service is the first hosted carrier-neutral VoIP peering service solution that fully manages the technical, operational, and commercial complexity of worldwide IP-based interconnections for wireline, wireless, and emerging carriers.

As one of the most carrier-dense interconnect sites in North America, tel(x) New York becomes the first hub of InfiniRoute's VoIP Peering eXchange (VPX), an interconnection infrastructure of InfiniRoute-enabled carriers using the MVP service solution to quickly and easily exchange VoIP traffic with global partners without the burden of capital investment or the complexity of interworking between competing VoIP standards and protocols.

By combining IP route optimization and voice-layer route control technology in a managed VoIP peering service, InfiniRoute offers its customers easy access to the benefits of VoIP technology while controlling voice quality and cost to maximize service reliability and profitability. In addition, partitioned Web portals provide full customer control over policy and routes.

http://www.infiniroute.com
http://www.telx.com



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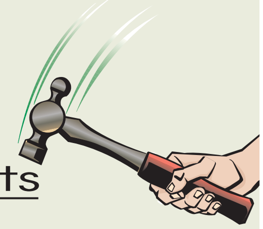

Quick Hits

IPtimize Deploys Kagoor For Managed VoIP Services

Kagoor Networks announced that IPtimize, Inc., a managed VoIP services provider serving small- and medium-size enterprises, deployed Kagoor's VoiceFlow session border controllers to deliver managed VoIP service to enterprises. By deploying Kagoor's VoIP over Broadband (VoBB) solution, IPtimize can leverage a client's existing broadband Internet connection to securely deliver local and long-distance voice services, along with easy-to-use, business-enabling applications.

As deployed by IPtimize, Kagoor's VoBB solution improves the efficiencies of service operation management by completely eliminating Internet Protocol address issues, and providing call monitoring, and statistics gathering. The VoiceFlow platform gives IPtimize the flexibility to direct customer traffic to different service platforms, allowing it to meet a variety of call-routing requirements. It also provides IPtimize with hosted Network Address Translation (NAT) and network protection solutions, which include advanced security features, such as topology hiding and protection from denial of service attacks.

http://www.iptimize.com
http://www.kagoor.com

SBC Selects Microsoft TV For IP Television Service

SBC Communications, Inc., announced an agreement with Microsoft Corp. to provide next-generation television services using the new Microsoft TV Internet Protocol Television (IPTV) Edition software platform. The SBC IPTV deal with Microsoft is valued in excess of $400 million over 10 years. SBC Labs has been testing an IP-based television service built on the Microsoft TV IPTV Edition platform since June 2004. SBC companies and Microsoft plan commercial availability of the IP-based television platform in late 2005.

In the first quarter of 2005, construction is expected to begin on the SBC Project Lightspeed, the company's initiative to deploy fiber closer to customer locations to provide new, feature-rich, IP-based services, including IP television, voice over IP (VoIP), and ultra-fast Internet access. Project Lightspeed is expected to reach 18 million households by year-end 2007.

http://www.sbc.com
http://www.microsoft.com/tv

Net Insight Introduces Device For Triple Play

Net Insight introduced the Nimbra 340 multi-service access and switching device. Housed in a small 2 RU outline, the new device can easily be located on customer premises and in co-location POPs. The Nimbra 340 represents a quantum leap forward in triple play transport by offering advanced video and data functionality that was previously only available in large central office units

Featuring bandwidth management with quality of service and multi-service capabilities, it would provide a flexible and cost-efficient access platform for service providers seeking to offer value-added, high quality triple play services such as multiple channels of Video on Demand and HDTV. Services such as Ethernet or ASI can have bandwidth allocated with guaranteed QoS in increments of 0.5 Mbps. An automated backplane providing up to 10 Gbps switching capacity makes the Nimbra 340 ideal for access switching of video and advanced data applications.

http://www.netinsight.net

SAIC Sells Telcordia To Providence Equity And Warburg Pincus

Science Applications International Corporation (SAIC) announced the signing of a definitive agreement to sell its subsidiary Telcordia Technologies, Inc., a provider of telecommunications software and services to Providence Equity Partners (Providence) and Warburg Pincus (Warburg) for $1.35 billion in cash. Providence and Warburg are equal equity investors in the transaction. The completion of the sale is subject to customary closing conditions, including regulatory approval.

http://www.telcordia.com
http://www.saic.com
http://www.warburgpincus.com

New Global Telecom Launches 6DegreesIP 3.1

6DegreesIP release 3.1 features support for G.729 voice protocol, which utilizes less bandwidth than the currently supported G.711 protocol; new TelPack calling features, including Emergency Zones, inventory reporting and call intercept; VoiceSelect Toll-Free service; a new generic URL for service providers; introduction of Outlook Conferencing Scheduler, which allows end-users to auto populate their 6DegreesIP Conferencing access information in Meeting Requests; and a new Advanced Provisioning training module.

http://www.ngt.com

ETI Announces Triad For FTTP Triple Play

ETI Software Solutions, Inc. (ETI), a developer of operational support software for the utility, municipality and telecommunication industries announced Triad, a robust software product that automates the provisioning of Fiber-to-the-Premise (FTTP) devices for the delivery of triple-play services (i.e., video, telephone, data, etc.) to residential and commercial customers.

http://www.etisoftware.com

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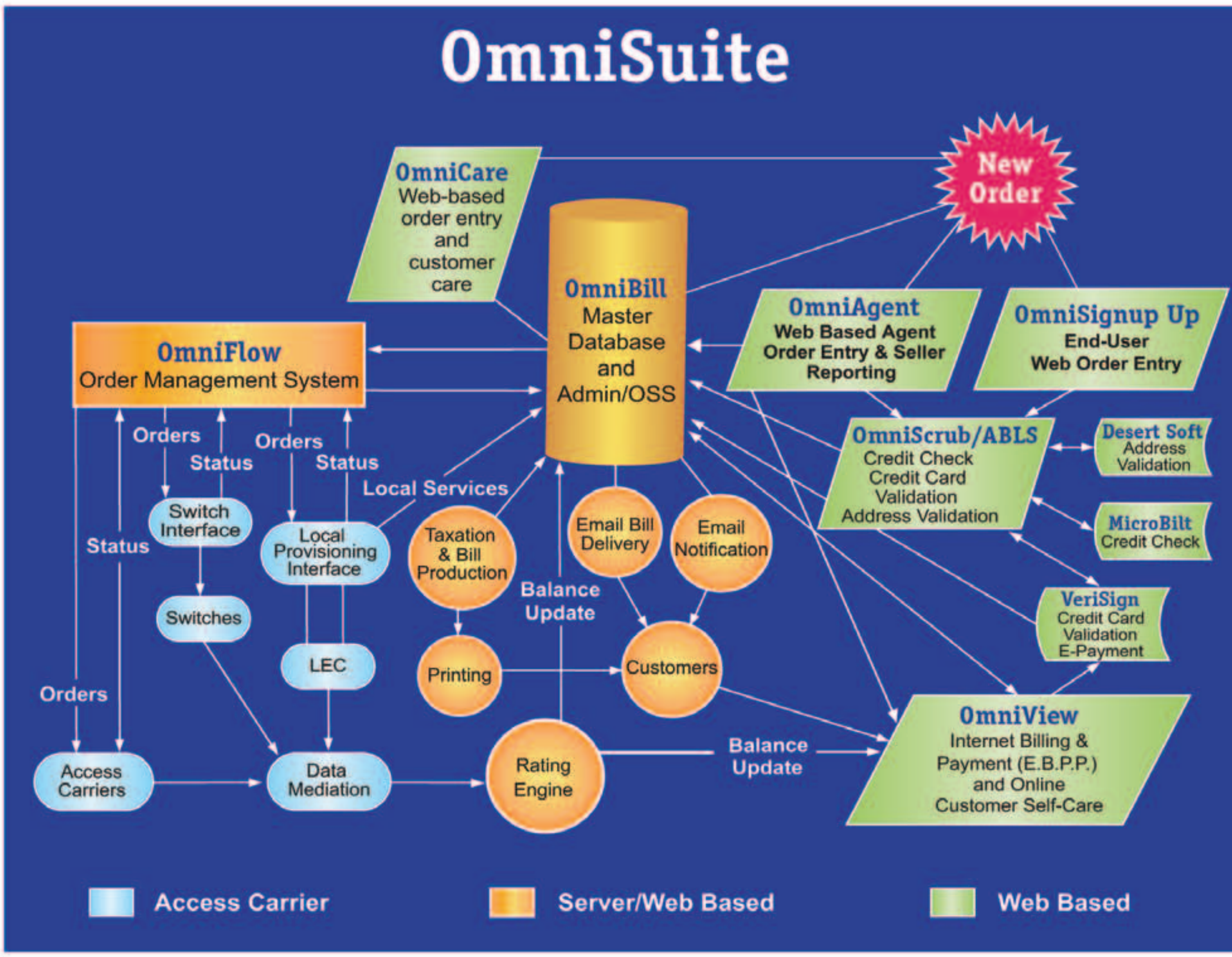
OmniSuite keeps your billing and OSS platforms working together



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DragonWave Introduces AIRPAIR Mesh

DragonWave, Inc., a supplier of wireless networks announced the release of the AirPair Mesh product configuration. AirPair Mesh enables 99.999 percent availability and increased network reach at a lower cost, providing customers a new level of service availability for fixed wireless services.

Further enhancing DragonWave's existing product portfolio, AirPair Mesh provides carrier grade service levels for Ethernet and TDM traffic. This is enabled by providing inherent equipment redundancy, as well as link redundancy. Additionally, AirPair Mesh reduces average path length, to further improve availability and antenna size requirements.

The AirPair Mesh increases the coverage of fixed wireless networks. The self-contained outdoor mesh infrastructure can be deployed at a service provider or customer site. This allows the service provider to expand well outside of their points of presence within a city. AirPair Mesh increases the reach of the POP, enabling service providers to get closer to the end customer, and alleviating many of the distance and line of sight issues that are common. DragonWave studies show that AirPair Mesh can increase the number of addressable customer sites by more than 10 times.

AirPair Mesh has already been deployed and is in-service in a number of carrier networks. Additionally, AirPair Mesh is being trialed by several North American carriers with the dual objective of addressing their access network constraints whilst supporting the migration to IP services.

DragonWave's AirPair product operates within both licensed and unlicensed radio frequencies of 18-to-26-GHz range, ensuring interference-free performance for customers' traffic. The DragonWave solution provides scalable, ultra-low latency, wire-speed native Ethernet connectivity up to 100 Mbps full duplex. AirPair has indoor and outdoor deployment options, and is a highly reliable platform enabling five 9's service availability.
http://www.dragonwaveinc.com

Taipei Project Selects Nortel Solution

Taipei City's "M-City" (Mobile City) project has selected a Wireless Mesh Network wireless mesh networks solution from Nortel to provide high-speed wireless local-area network (LAN) broadband access and new wireless services in Mass Rapid Transportation stations, selected commercial buildings and other key locations across the city.

The Taipei City Government evaluated numerous bidders for months before recently signing a contract with Qware, which plans to operate a wireless broadband network based on Nortel's Wireless Mesh Network solution. Qware has issued a Notice of Award to Nortel and the parties are currently negotiating a supply agreement. Qware expects to have 10,000 wireless access points in service by year-end 2005 to provide coverage for Taipei City, an area of 272 square kilometers where 90 percent of Taipei's 2.65 million people live.

"Our selection of Nortel was based on its unparalleled experience in deploying numerous wireless networks worldwide," said C.S. Lin, chairman of Qware and chief executive officer of President Group.

For this project, Nortel will provide its Wireless Mesh Network solution, including the Nortel Wireless Access Point 7220, Wireless Gateway 7250, Nortel Optivity Network Management System, and other related network management elements. Nortel's Wireless Mesh Network solution uses IEEE 802.11 standards, allowing users with Wi-Fi enabled laptop computers or handheld computing devices to access the network without new hardware or software.

Nortel's Wireless LAN portfolio is comprised of two powerful, complementary offerings: the WLAN 2200 series, which was introduced in March 2003, and the Wireless 7200 series, the Wireless Mesh Network solution.
http://www.qware.net

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Airespace Adopts Azimuth Test Solution

Azimuth Systems, Inc., provider of automated wireless network test systems announced that Airespace, provider of WLAN switch/controllers, selected the Azimuth W-Series test platform for its internal quality assurance, performance and competitive benchmarking test processes. Airespace products are built on a foundation of performance and quality that require world-class test processes.

"The W-Series is easy to use and is supported by a significant suite of standard test routines, covering 802.11 a, b and g protocols. The 802.11 a protocol is particularly critical in the enterprise segment in which we compete," said Bob O'Hara, co-founder and CTO of Airespace.

"Market data clearly indicates that Airespace is the number one solution in the WLAN switch/controller space, and the company's thorough approach to quality-oriented testing exemplifies how it has been able to achieve this top position," said Ray Cronin, president and CEO of Azimuth.

The Azimuth W-Series is a standardized test platform for system level testing of 802.11 wireless access points, clients and other devices. Designed from the ground up as an off-the-shelf, wireless LAN test platform, W-Series systems provide the ability to configure an entire WLAN network in a bench top chassis designed for complete Radio Frequency (RF) isolation and control. The flexibility and programmability of the W-Series allows for the thorough evaluation of wireless LAN equipment under varying mobility conditions and traffic patterns, as well as precise analysis of the results. The system offers an environment to perform software design validation and to test advanced wireless functionality and performance including the latest IEEE standards.

http://www.airespace.com
http://www.azimuthsystems.com

Symbol Introduces Wi-Fi Products

Symbol Technologies, Inc., a provider of mobile enterprise products, introduced its next generation WS 2000 enterprise wireless switch and new AP 300 802.11a/b/g wireless access port for branch offices and small and medium businesses. The launch of these products further continues Symbol provides customers with additional opportunities to untether their networks for true mobility.

To expand the wireless connectivity options supported by the WS 2000, Symbol also introduced the AP 300, an access port that supports 802.11a/b/g. Utilizing a tri-mode access port like the AP 300 along with the Symbol WS 2000, businesses will be able to further reduce the cost of deploying, implementing and managing a wireless LAN, while increasing features, functionality and security of the wireless LAN infrastructure.

Small and medium business (SMB) customers have many different requirements for their networks, and no mobility system is the same. Symbol's wireless portfolio, which supports tri-mode standards (802.11 a/b/g), offers customers the flexibility to continue to build and expand their enterprise wireless network. This is important for small and medium businesses, as they need the flexibility to grow with their business. The AP 300 provides such flexibilities as low out-of-the-box purchase price, features that provide lower total cost of ownership, and the opportunity to expand internationally with support for 802.11h, a worldwide standard.

The Symbol WS 2000 Release 1.5 and AP 300 are currently available. Existing users of the Symbol WS 2000 can upgrade their firmware to release to 1.5. The suggested list price for the Symbol WS 2000 is $999. The suggested list price for the Symbol AP 300 is $349.

http://www.symbol.com

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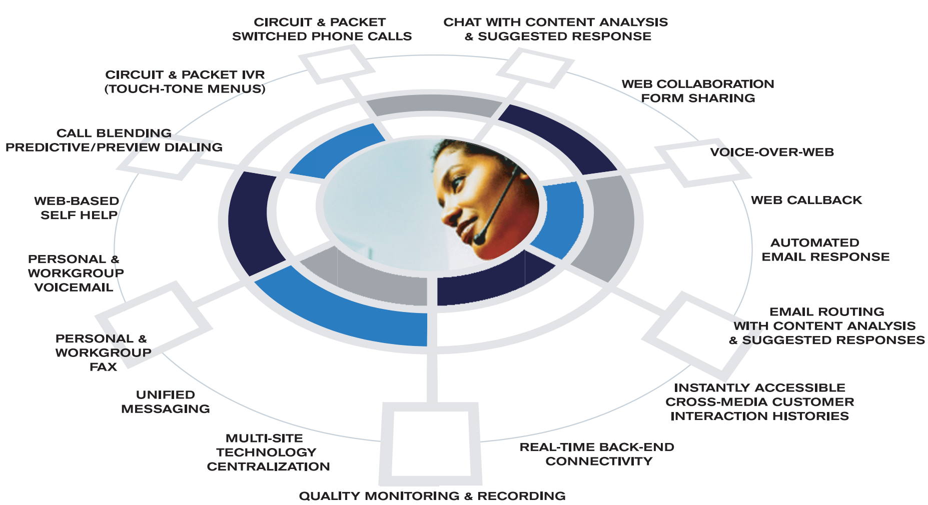

Most vendors require lots of time and money to program, integrate, implement and maintain multimedia contact centers or share technology across geography. They also can't implement changes without costs, risks and delays. And by the time "upgrades" are delivered, your needs may have changed. Telephony@Work technology leverages traditional needs-analysis questions yet allows your managers to define (or redefine) their answers via browser menus - in real-time and at no cost. This enables them to provision or modify any business process on-the-fly in order to increase efficiency on any communications channel - phone, fax or Internet - for any group, anywhere in the world. As you might expect, increased revenues are the natural result of being able to "fix" broken or strained business processes on demand.

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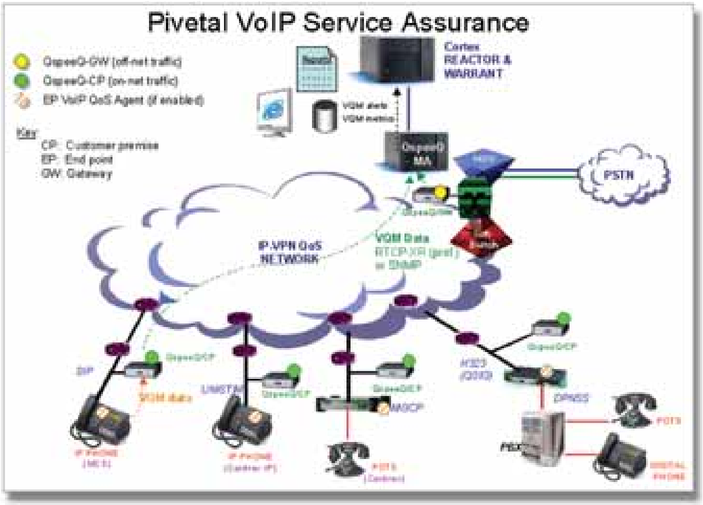
Pivetal Integrates Telchemy's VoIP Management Software

Telchemy, Inc., a VoIP performance management software provider, and Pivetal Limited, a provider of artificial intelligence-based network management systems, announced that Pivetal has embedded Telchemy's VQmon/SA(Stream Analysis) call quality analysis algorithms in their new QspeeQ family of VoIP Speech Quality Monitoring products.

By adding Telchemy's VQmon technology, Pivetal will provide VoIP-specific monitoring to its customers that assesses user-perceived quality in real time. This new VoIP specific capability further enables Pivetal's Cortex Service Assurance Suite to identify and fix network issues before they impact on service quality levels.

Specifically, Pivetal will feature VQmon/SA call quality monitoring and analysis technology in the following products:

- QspeeQ/CP, a customer premise probe monitors all voice call streams and deduces from these speech quality metrics for SLA supervision purposes. It also captures speech quality metrics reported by VQM enabled gateways and endpoints in the subnet it monitors and forwards this (along with its own VQM metrics) to a central QspeeQ/GW probe.



- QspeeQ/GW, a sophisticated stand alone probe also hosts an instance of QspeeQ whose function is to monitor and calculate the speech quality of all off-net calls at the point of egress from the VoIP 'cloud' adjacent to the media gateway. It produces an overall call speech quality record (CQR) by combining the VQM metrics of both 'ends' of each call. This raw CQR can then be used to generate alerts, aggregated into higher-level management information, "historised," etc.

"As service providers, large and small, begin rolling out real-time multimedia services such as VoIP, the need to monitor and enforce SLAs (Service Level Agreements) at the customer premise becomes critical," explained Louis Meyer, CEO and Founder of Pivetal.

http://www.telchemy.com
http://www.pivetal.com

RADCOM Unveils 3G Video Analyzer

RADCOM Ltd., a network test and service monitoring solutions provider, introduced its 3G Video Analyzer. The new solution monitors end-to-end cellular video service quality over 3G networks. It provides an objective evaluation of the quality of the video service delivered by cellular phones operating on various interfaces. The 3G Video Analyzer offers a comprehensive analysis of video service from the user plane to the core network and physical connection. It allows wireless device manufacturers and network operators to monitor and troubleshoot high-quality video sessions such as video teleconferencing, video streaming, video-on-demand and interactive video.

As cellular operators upgrade from 2.5G to 3G and next generation cellular networks, video services are regarded as a key advantage in increasing the Average Revenue Per Unit (ARPU). However, solving the complexity of running wideband real-time video services on cellular networks has, until now, not been easy. The 3G Video Analyzer isolates problems, such as long video setup duration, that cause poor quality video characterized by jerkiness, incorrect audio, blank picture transmission and poor lip synchronization. These problems and their adverse effects are often the result of interoperability issues, unstable bandwidth allocation and the number and diversity of signaling protocols and media codecs. The 3G Video Analyzer quickly identifies the source of the problem, minimizing troubleshooting time thereby improving the subscriber's quality of experience.

An add-on to the Cellular Performer, the 3G Video Analyzer examines video over 3G-324M circuit switch sessions. It functions on various connections such as UMTS and CDMA networks. It analyzes video over IP using different IP multimedia signaling protocols. In addition, it synchronizes multiple video playback of MPEG4 and H.263 sessions of all video participants in a single view. Furthermore, its perceptual impairment analysis capability provides objective video quality evaluation both intrusively and non-intrusively.

http://www.radcom.com

Real Protection

In an emergency, your VoIP subscribers need to know they can call 911.

VoIP is only as reliable as your subscriber's broadband connection. Until now, any disruption in service meant an immediate crash in VoIP – and severed access to 911.

Pannaway Technologies' intelligent Personal Branch Gateway (PBG™) lets telcos worldwide offer the advanced calling features of VoIP, without sacrificing E911, CALEA or Lifeline support. By dynamically re-routing emergency calls in order of priority, this multifunction premise device plays a critical role in disaster recovery.

Aside from carrier-class telephony services, the technologically engineered PBG also streamlines video and data service delivery for a true Triple Play of services.

Let your subscribers know they're protected. Guarantee simultaneous delivery of innovative IP-based voice, video and data services with Pannaway PBG.

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Web: pannaway.com

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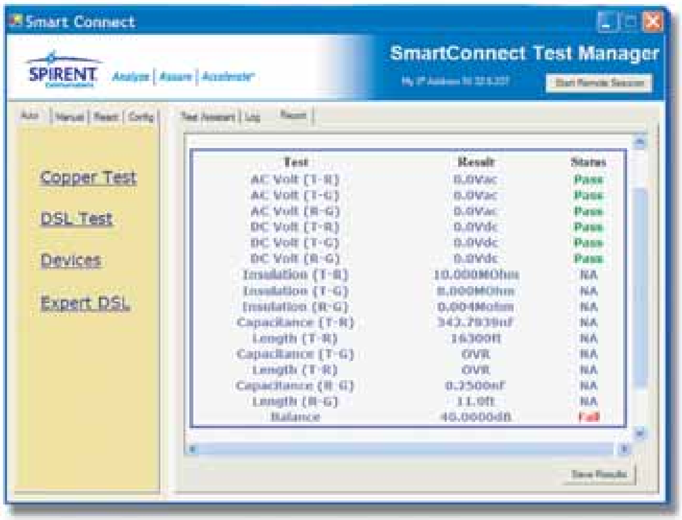
Spirent Intros SmartConnect Field Testing Tool

Spirent Communications announced availability of its new SmartConnect solution to help service providers streamline the field testing process, ensure proper use of methods and procedures, and virtually eliminate costly two-technician testing.

SmartConnect is a single software application that increases field efficiency by automating and documenting the field testing process and centralizing the storage of test results for reference and analysis by all testing personnel. The company says it provides a cost-effective field testing solution for all deployed services and integrates with back-office resources, such as test OSS, records databases and dispatch systems. It also provides troubleshooting expertise to field technicians, enables real-time visibility of field test scenarios, and standardizes service methods and procedures compliance to ensure consistency and completion of all necessary tests.



"SmartConnect is the latest addition to our portfolio of field test solutions," said Jim McRae, managing director of Spirent's Field Test unit. "With SmartConnect, service providers can make immediate improvements in the service management process. SmartConnect provides field techs with all of the resources needed to complete the work order the first time. This minimizes the long delays and costs associated with contacting back-office resources and various remote personnel, which means that field technicians spend less time per job."

SmartConnect is installed on a technician's laptop in minutes, driving improvements immediately. With Spirent's CopperMax OSP, a multi-function copper and DSL tester, seamless integration is optimized with a wireless approach. This solution links the laptop and field tester through a short-range wireless connection (such as Bluetooth), and the laptop and back-office network via a wide-area connection (such as cellular data).
http://www.spirentcom.com

Freescale Licenses Radvision Toolkits

RADVISION announced it licensed both its H.323 and SIP developer platforms to Freescale Semiconductor for the development of a wide variety of chip-based IP communications applications optimized for various Freescale processors.

Freescale, a global leader in the design and manufacture of embedded semiconductors for the consumer, industrial, networking, wireless, and automotive markets, plans to use the RADVISION toolkit to develop applications that sit on top of and leverage the specialized processing features of various Freescale chipsets including its ColdFire microprocessor core and i.MX21 multimedia applications processor.

Freescale chose the RADVISION SIP developer platform to develop a WiFi IP telephony developer solution based on the ColdFire microprocessor core. By leveraging the ColdFire core architecture and its hardwired DSP instructions for voice-over-IP, the combined platform represents an ideal choice for equipment developers who need an integrated hardware and software solution for turnkey development of a WiFi phone.

Based on the 802.11b standard, the Freescale system based on its ColdFire technology provides the equipment developer with a powerful low-power consumption processing and applications platform upon which they can develop cost effective, feature-rich WiFi IP telephony systems with unique functionality such as MIDI and MP3 ringtones.

In addition, RADVISION's H.323 and SIP developer platforms have been designed for the development of a voice, video and multimedia over IP reference design for the i.MX21 multimedia applications processor from Freescale's Wireless and Mobile Systems Group. The combined solution is optimized for WiFi and wired devices and pairs the i.MX21 processor's powerful multimedia processing with an advanced suite of features such as H.263 and MPEG4 video, JPEG imaging, mirroring, zooming, and rotation. The underlying IP signaling and call setup/call control portion of the applications were developed using the RADVISION H.323 and SIP developer platforms.
http://www.radvision.com

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SPIRIT Offers Multi-PASS Full Duplex Conferencing

SPIRIT, a supplier of embedded software products for telecom and VoIP, announced Multi-PASS — a new software product targeted for PC-based VoIP communication. SPIRIT Multi-PASS is an advanced conferencing solution providing both server-based and serverless peer-to-peer and multipoint conferencing. SPIRIT Multi-PASS enables any number of conferences and unlimited number of participants per conference without degradation of speech quality and user experience. So speech clarity is preserved even for multi-user conferences, as in a face-to-face conversation.

SPIRIT Multi-PASS is based on Prosodic Active Speaker Selection (PASS) algorithm. Analysis of prosodic speech characteristics combined with dynamic utterance modeling makes speakers activity estimation more accurate. The analysis is performed on each speaker's side and does not load server CPU. In accordance with the compact PASS info, conferencing engine directly passes voice data from active speakers to other participants. SPIRIT Multi-PASS doesn't require participants to have broad bandwidth. Due to smart traffic channeling, voice traffic and channel usage are optimized to reduce CPU load.

"SPIRIT PC-based VoIP solution is targeted to offer better-than-PSTN quality at low cost", says Andrew Sviridenko, founding CEO of SPIRIT. "Following this strategy, SPIRIT Multi-PASS is an additional feature to provide executive-level conferencing quality, optimized traffic and channel usage, and reduced costs of multi-user support. All these features significantly cut operator and enterprise expenses."

http://www.spiritdsp.com

MERA Announces SIP Registrar

MERA Systems, Inc., a supplier of scalable and feature-rich session controllers for intelligent management of VoIP networks, announced the availability of SIP registrar functionality in its MVTS Session Controller. The SIP registrar will guarantee fast and easy interconnects with SIP-based networks and equipment, thereby enabling carriers to fully realize business benefits from their VoIP networks.

The SIP registrar allows dynamic registration of SIP endpoints. It means that telcos can now collect calls from any type of SIP devices — even those devoid of a fixed IP address such as IP PBXs, IP phones, and low-capacity gateways. The ability to register such devices is especially critical for service providers that focus on the end user or business segment.

“The SIP registrar opens up the business segment for telcos,” says Konstantin Nikashov, CEO of MERA Systems. “Since most endpoints and apps are now SIP-based, carriers need a bridge between their switching facilities and the customer networks. Over the past year MERA has focused on strengthening the SIP capabilities of its Session Controller. Now, along with SIP-to-H.323 translation, transcoding and SIP stateful proxy, we offer carriers a means of dynamic IP address registration. In brief, the enhanced support for the SIP standard adds greater flexibility to inter-carrier peering and makes the benefits of the VoIP technology more evident to the enterprise customer.”

http://www.mera-voip.com

MIND CTI Expands Product Suite Family

MIND CTI Ltd., a provider of real-time mediation, rating, billing and customer care solutions for pre-paid and post-paid voice, data and video, announced the release of new modules that operate either in conjunction with the MIND billing and customer care solutions or with any third party solution. The new modules include enhanced provisioning for IP and traditional equipment, SIP application and network monitoring.

Features include:

Operation Monitoring System — A centralized monitoring tool that enables real-time monitoring of the various modules of the system, including mediation, provisioning and billing processes as well as providing real-time performance of each module and server.

Resource Management — Assigns and manages physical inventory items as well as logical items. The new module manages the complete lifecycle of the items, stores warehouse inventory with real-time information on each of the items and enables on-line allocation and reservation of the items by a customer care representative.

Convergent Provisioning — Enables automatic and online provisioning of network elements for service creation and activation, including SoftSwitch provisioning for VoIP networks, such as cable and ADSL. It supports HLR provisioning for mobile networks and content server provisioning for all IP based enhanced services. The solution allows easy audit trail and management of all provisioning activities and results.

SIP Application Server — Enables SIP based real time pre-paid services by connecting directly to a SoftSwitch or any other call routing network element. It provides seamless interoperability with leading SIP VoIP equipment vendors and is fully supported for advance call features such as call forwarding.

http://www.mindcti.com

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IBM Selects EADS To Power Sprint PCS Contact Center

EADS announced a new project working with IBM Global Services to unify all 40 Sprint PCS contact centers worldwide under a single platform. The first phase will initially include 12,000 agents in seven countries, including approx. 6,500 agents working from remote locations.

To power the project, IBM purchased EADS' Centergy Contact Center solution, which includes Centergy Remote and Centergy Reporting applications. Centergy can handle more than 1.5 million calls per hour in peak periods. With this capacity, IBM can meet Sprint PCS' requirements, including the need to handle 250 calls per second.

Centergy's architecture allows for multiple contact centers to operate as one, regardless of geography, consolidating data management and administration to maintain maximum service levels across the enterprise. Agents working in a Centergy environment can be located anywhere they have phone and Internet connections. This enables agents to receive the same features and functionality whether they are in a major contact center, operating on other vendor platforms or located remotely. This same approach allows Centergy functionally to be deployed in contact centers of as few as five agents.

This is the second such system deployed by Sprint that utilizes EADS' Centergy solution. Sprint Local Telephone Division has operated 6,000+ agents on a single Centergy system since 2002.
http://www.eadstelecom-na.com
http://www.ibm.com

FrontRange Launches IP Contact Center 3.7

FrontRange Solutions announced the availability of its IP Contact Center 3.7 software suite. Sharing an integrated architecture with other FrontRange products, IP Contact Center helps small to mid-size enterprises (SMEs) and distributed enterprises reduce telephony costs and increase agent productivity.

Developed to promote organizational efficiency, the product can be deployed on a modular basis either as part of the existing communications infrastructure or as a stand-alone solution.

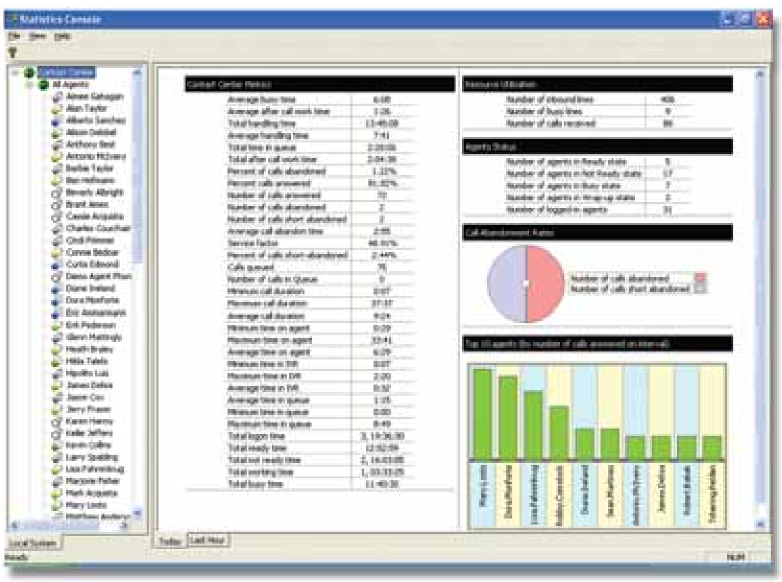
IP Contact Center 3.7 has a single point of administration, which reduces the typical complexity required to support a traditional telephony infrastructure. The product delivers real-time and historical reporting, queuing, automated call distribution (ACD), integrated voice response (IVR), and screen-pop capabilities, as well as integration with the award-winning FrontRange HEAT and GoldMine products.

Key Features include:

• Real-time and Historical Reporting: FrontRange IP Contact Center provides an in-depth set of historical reports and real-time views accessed through an easy-to-use dashboard.

• Agent Dashboard: The product uses a software-based SIP-phone, providing click-to-dial, dial-from-directory and desktop-conferencing features to elevate agent productivity, as well as presence management of queue-monitoring features.

• Advanced Skills Based Routing: IP Contact Center has priority, skills-based and group routing, as well as the ability to incorporate real-time business data from applications such as GoldMine and HEAT as a means of directing customer calls to the appropriate agent the first time.



• Virtual Agent: The product provides the customer with self-service options, speeding up customer queries and reducing administrative calls to the contact center. Agents are more productive, and customers receive a better service.

• Web-based Application Builder: The software includes an easy-to-use graphical interface that enables the administrator to quickly and easily build new call treatment and IVR schemes, as well as conduct powerful business application integration functions. This alleviates the need for expensive external resources.

• HEAT and GoldMine integration: Integration with FrontRange's Service Management and CRM solutions streamlines workflow by transferring live calls and screen pops.
http://www.frontrange.com

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Contactual, Netbytel In Speech Pact

Contactual DBA, a provider of OnDemand Contact Center solutions announced that it partnered with NetByTel, a provider of telephone self-service solutions. The partnership will accelerate availability of NetByTel's portfolio of hosted voice recognition solutions to current and future Contactual customers.

Over the past five years, Contactual has developed a feature-rich on-demand contact center product that leverages state-of-the-art technologies, like VoIP. Contactual has identified and delivered the key features that any call center needs, without lowering our industry-leading standards for reliability, scalability and ease-of-use.

Features include:

- Multimedia Interaction Router/ACD
- Multimedia switch
- Real-time reporting and monitoring
- Call recording and monitoring
- Interactive Voice Response (IVR)
- Computer Telephony Integration (CTI)
- Chat
- Co-browsing
- Knowledge Base
- Self Help
- Web Administration

http://www.contactual.com
http://www.netbytel.com

Retailer Design Within Reach Standardizes On Aspect Iphinity

Aspect Communications Corporation, a provider of enterprise customer contact solutions, announced that Design Within Reach Inc., a provider of distinctive modern design furnishings and accessories, has achieved a 20 percent decrease in contact center operating costs per order and contact since implementing the Aspect Iphinity Call Center.

The Aspect Iphinity Call Center solution was built for organizations like DWR with fewer than 150 contact center agents. DWR's agents handle 3,000 telephone calls and 1,000 e-mails weekly. Aspect Iphinity Call Center fits the company's criteria of an easily installed solution with high-end capabilities to support existing customer demand in the short term, as well as support longer-term goals, such as increased service levels, reduced hold times and the ability to track agent performance. The solution's reporting tools enable DWR's supervisors and managers to view contact center activity in real time, tracking service levels, abandonment rates, total orders, average order amounts and a variety of other statistics, so that adjustments can be made and agents can monitor and manage their own performance.

http://www.dwr.com
http://www.aspect.com

Unveil And Genesys To Deliver Speech Self-Service Systems

Unveil Technologies, Inc., a provider of conversational voice self-service applications for call centers, and Genesys Telecommunications Laboratories, Inc., an Alcatel company, announced that Unveil joined the Genesys InterActs Partner Program as a Strategic Member. As part of the agreement, Unveil can resell both the Genesys 7 contact center software and the Genesys Voice Platform in conjunction with its Conversation Manager solution.

Unveil's Conversation Manager speech self-service applications run on the Genesys Voice Platform, an open standards-based IVR solution that allows customers to leverage existing Web applications and infrastructure for voice self-service via any phone.

Unveil Conversation Manager is voice application software that enables enterprise call centers to build, deploy and maintain high performance speech applications, or "virtual agents," that increase automation rates, improve customer service and reduce the total cost of ownership. Unlike traditional speech self-service applications, Unveil virtual agents: Engage customers in unscripted, open conversations that enrich the customer experience and automate complex transactions; Integrate with live agents to extend self-service transactions and improve caller satisfaction (Conversation Assist); and Utilize real customer interactions to dynamically improve over time, reducing the total cost of ownership for speech (Adaptive Learning).

http://www.unveil.com
http://www.genesyslab.com

VegaStream Extends VoIP To Latin America With Smart Networks

VegaStream, a supplier of dedicated business VoIP gateways, announced an agreement with Smart Networks Solutions (SNS), a reseller and systems integrator to the Latin America market. Under the terms of the agreement, SNS will re-sell VegaStream's portfolio of VoIP gateways to its customers in the Latin American region. SNS chose VegaStream because of increased customer demand for reliable VoIP solutions based upon SIP.

Smart Networks Solutions is a Latin American system integrator of VoIP solutions providing business and network integration for financial institutions, the hospitality industry and other market sectors. Their work also includes provision of network equipment, hosting and collocation services for service providers and enterprise customers in the region. With a growing demand from their customers for VoIP services based upon SIP, SNS required a proven and reliable portfolio of SIP capable VoIP gateways.

VegaStream's portfolio of VoIP gateways provides seamless interoperability between the varied legacy of PSTN interfaces and new IP networks. Targeted at businesses and service providers, the products are based on international communications standards, including both SIP and H.323 and range from eight to 120 ports covering both analog and digital interfaces.
http://www.smartisvoip.com
http://www.vegastream.com

Peerio Enters The Japanese Market

Popular Telephony, Inc., the telecommunications middleware company behind the Peerio serverless telephony invention, announced a licensing agreement with E-with-you, an integrator of new technologies in Japan to deploy Peerio within the WiFi PAS telephony device, designed by E-withyou.

The Peerio-enabled WiFi PAS telephony device will be marketed first to the enterprise market in Japan. Companies deploying the Peerio powered E-withyou devices will eliminate up to 80 percent of the actual system costs and up to 90 percent of total cost of ownership when compared to a traditional telephony system. As with all Peerio-enabled products, the device will include a pre-selected set of telephony features upon release.

"Peerio is attractive in many ways including its scalability personal to enterprise, choice of functionality private to business, interoperability (i.e., any network, any carrier), and its future possibilities (any media-voice, sound, image). It has the potential to change the landscape of media communications," said Shun Matsuda, E-withyou's CEO.
http://www.populartelephony.com
http://www.e-withyou.com

ECI Telecom, Chiaro Networks In Partnership

ECI Telecom Ltd., a provider of telecom solutions to carriers and service providers, and Chiaro Networks, a developer of carrier infrastructure-class IP/MPLS routing platforms, announced that they have entered into a strategic partnership. As Chiaro's partner, ECI will exclusively distribute Chiaro's Enstara IP/MPLS platform worldwide and is also taking part in Chiaro's funding.

"As our customers move toward next-generation Ethernet and IP/MPLS-based networks, ECI is expanding its data capabilities and further complementing its broadband and optical product portfolio by providing innovative, competitive migration solutions," said Doron Inbar, ECI's president and chief executive officer. "Our strategic partnership with Chiaro will further enhance our overall IP competence, enabling ECI to provide important additional elements in response to our customers' needs as they fulfill their own Ethernet/IP network visions. Our decision to partner with Chiaro followed specific indications from several ECI customers on the fit and need of Chiaro's Enstara platform in their networks."
http://www.chiaro.com
http://www.ecitele.com

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TZ170-25	25-Node	**$CALL**
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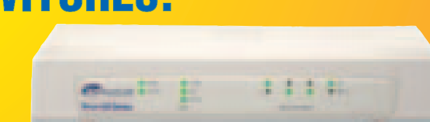

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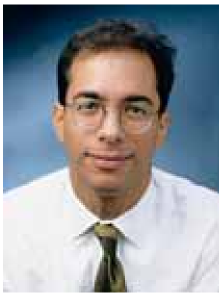

By Marc Robins

CASP Rebirth

The current mass mindset regarding the CASP (communications application service provider) (news - alert), or outsourced communications, space seems strikingly familiar to the positions most people now hold with respect to the dotcom space. The widespread negativity spawned by the crash and burn of 2001 that blinded many to the viability of such endeavors has dissipated, and there is a new sense of rebirth, optimism, and opportunity.

It's clear the Internet is here to stay, and people are not going to stop communicating and needing access to information and services. Indeed, the concept of outsourced communications services is even more valid today as it was four years ago, given the proliferation of "last mile" broadband access. Granted, market conditions have affected the way such businesses are set up and run, but market conditions are also setting the stage for some CASP offerings to flourish.

One only has to look as some of the players to see reason to cheer. From passionate regional outsourcers such as M5 (http://www.m5net.com) in New York to the growing adoption of service platforms such as those offered by Sylantro (http://www.sylantro.com), there is a new flurry of activity in the space.

M5 typifies the CASP of today: Young, aggressive and full of technological vim and vigor. The company serves the Metro New York area with Hosted IP PBX (define - news - alert) and data communications services, and was founded in 2000 by the same management team that ran Interport Communications, the ISP pioneer for New York businesses. Today, the company serves over 300 New York area businesses, and is embarking on an aggressive expansion plan to enable it to grow its customer base into the thousands.

Indeed, I believe there is a solid rationale to M5's optimism. Let's review some of the key reasons why outsourced communications services are (still) worth believing in.

Outsourcing Helps Control Costs. Cost cutting is a legacy from the recession that is here to stay: Managers are constantly under pressure to rein in costs, and to control expenditures on new technology, new marketing initiatives, and new hires. By outsourcing communications services, enterprises can eliminate a number of capital expenses, and many related costs associated with the acquisition of new communications technology.

Outsourcing Lets Businesses Focus on Their Core Competencies. The need to effectively compete and gain competitive advantages hasn't been affected by the bumpy economy. Outsourcing the care and feeding of communications systems can free up staff to focus on tasks that directly contribute to improvements in productivity and profitability. With fewer "distractions," companies will find their time-to-market with new products or services is reduced, and they become more agile competitors.

Outsourcing Provides Freedom From Obsolescence. All outright technology purchases have a limited lifespan. Phone and messaging systems generally last longer than PCs, but even in telephony the pace of change has been accelerating to the point where new generations of products — especially IP-based solutions — are appearing every six months. Outsourcing can insulate a business from the pain of dealing with obsolete technology.

Outsourcing Lets Businesses Take Advantage of New Technologies and Applications. Many legacy communication systems lock users into static feature sets, and are extremely difficult to upgrade. By outsourcing, businesses can add new features and capabilities to their communications arsenal — often times preserving their investment in existing systems. In addition, users are able to immediately enjoy the benefits of new, advanced services as new technologies and applications are implemented by their service providers.

Outsourcing Alleviates Staffing Woes. Outsourcing communications services can reduce or eliminate the need to hire and assign dedicated staff to manage and maintain in-house systems, perform upgrades, handle moves and changes, etc., especially with new services that provide Web-based management and provisioning capabilities.

Certainly, there are many other ways an outsourced communications strategy can aid a company's bottom line. What's clear is that in a time when many companies are hesitant to spend money, the value proposition for CASPs is ascendant. Business managers are eager to investigate the cost advantages an outsourcing strategy enables them, and are wide open to alternatives. IT

In a time when many companies are hesitant to spend money, the value proposition for CASPs is ascendant.

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

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By Tony Rybczynski & Paul Relf

Proactive Voice Quality Management: Can You Hear Me?

Your network is 100 percent up, but an executive calls in a trouble report saying that an IP telephony call with an important client was unintelligible and was dropped. What tools do you have to diagnose the problem beyond the description from your executive? Was the problem due to packet loss, buffer overflow or excessive delay, variations? Very likely, you're unable to do anything since any attempt to duplicate the problem will probably fail. In fact, according to the 2003 VoIP State of the Market Report by Steven Taylor of Distributed Network Associates, the second most common IP telephony deployment impediment was the lack of systems for managing and troubleshooting VoIP quality (foremost was lack of budget). A big part of the answer is proactive voice quality management, an approach that can be extended to multimedia as well.

Operationalizing IP Telephony Service Quality Management

Proactive voice quality management is a life-cycle approach, starting with a network assessment to identify systemic network impairments, followed by a pre-deployment phase to enable the network QoS and potentially run a small pilot project. Once the network is deployed, ongoing real-time monitoring of end user voice quality and overall system health is required. The last phase is voice and network quality reporting for SLA management and planning. Since the network is ever evolving, this process needs to be institutionalized as part of the operational environment.

The linchpin in linking network performance and end user voice quality is to be able to map how network impairments (e.g., delay, jitter, packet loss, and echo), media endpoint impairments, and analog effects impact the user perception of voice quality. On traditional telecom networks, a Mean Opinion Score from 1 to 5 (5 being 'very satisfied') is a widely adopted standard approach for assessing voice quality, which usually requires intrusive testing equipment to be measured accurately. The international standards bodies have developed what is known as the E.Model, which provides the basis to quantitatively measure network, media endpoint, and analog effects in real-time to accurately produce a call quality score.

Once the network is deployed, ongoing real-time monitoring of end user voice quality and overall system health is required.

This approach enables real-time monitoring for actual calls on a dynamic packet network. The E.Model estimates user satisfaction associated with a number of technical impairments and the result is an R-value from 0-100. Toll quality is usually considered as a MOS of 4 and R-value of 80. There is a well-defined relationship between R-value and MOS, resulting in an accurate way to determine MOS in real-time for actual VoIP calls.

To date, the networking toolkit for IP telephony management has been limited to a protocol called Real Time Control Protocol (RTCP) (define - news - alert), a control relative of the Real Time Protocol (RTP), which is used to monitor the effectiveness of delivering a multimedia stream of data across the network measuring round-trip delay, packet loss, and jitter. Unfortunately, these metrics alone are too coarse and are insufficient to calculate the R-value or how the end-user actually perceived the quality of a given call. Moreover, these statistics are often averaged out and do not capture the transitory nature of network impairments. What is needed is to unobtrusively monitor the right set of metrics, which determine end-user Quality of Experience and do it in real-time on an end-to-end basis.

The New Language for Voice Monitoring

The IETF has issued a new VoIP management protocol, RFC3611RTP Control Protocol Extended Reports (RTCP XR). It defines a set of key call-quality-related metrics that contain information for assessing IP telephony call quality and diagnosing problems. RTCP XR can be implemented as software in IP phones, soft clients and gateways. Metrics such as echo, packet loss rate, jitter buffer discard rate and the distribution of lost and discarded packets can be captured by a service management system. The loss/discard distribution describes the call in terms of bursts (periods during which the loss/discard rate is high enough to cause noticeable quality degradation) and gaps (periods during which lost or discarded packets occur infrequently and hence quality is generally good). It also reports end system delay, which represents the

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delay that the IP telephony endpoint adds (because of encoding, decoding, and the jitter buffer), as well as the signal and noise level associated with the received signal, making it easier to identify signal- and noise-level problems.

Making Your Life Easier

Packet loss and jitter affect voice call quality. Delay and echo can make conversations difficult and make the effects of echo obvious. The jitter buffer removes the jitter in the receiving IP phone or gateway, but this process adds delay and causes packets that arrive late to be discarded. If a signal is too loud, too quiet or too noisy, call quality suffers. The VoIP performance metrics defined in RFC 3611 RTCP XR take the big picture into consideration when calculating a voice quality score, resulting in a better match to the real end user Quality of Experience for a given call.

Proactive voice quality management helps you determine quickly and easily how well IP telephony will work on a network prior to deployment.

Proactive voice quality management helps you determine quickly and easily how well IP telephony will work on a network prior to deployment. It helps you configure the end to end system for optimal performance. It measures voice quality in real-time from and end-user perspective using standards based technology and explains why you are experiencing reduced call quality. Reports allow you to monitor service levels, call quality, overall performance, usage trends and capacity planning. Most importantly, it will allow you to know about and fix problems before they become service impacting. **IT**

Tony Rybczynski is Director of Strategic Enterprise Technologies at Nortel. He has over 30 years experience in the application of packet network technology. Paul Relf, VoIP QoS Management PLM, is responsible for the definition and delivery of an Industry leading approach to manage IP Communications service quality for Nortel solutions. For more information, please visit http://www.nortel.com.

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To succeed, service providers will have to provide voice, video and data to their customers, and perhaps even mobility services and data services like anti-virus. Like any great opportunity, you can't effectively deliver these products to your customers, however, until you've tackled a number of issues, including your underlying service delivery platform, access technologies, bandwidth issues, implementation challenges, management and billing and finally, user interfaces and CPE.

VoIP E-911

The positive press friendly to VoIP that we witnessed for the past year will vanish the moment someone is injured or worse because there is a problem with VoIP and e911 connectivity. I consider this a stumbling block that needs addressing on our way to achieving VoIP 2.0.

VoIP Security

There can be no greater prize for a hacker than to be able to crack your network and listen in on your customer's conversations. Furthermore, all the same issues any data network faces such as hacking, spoofing, viruses and spam are threats to stable and consistent VoIP service delivery. Deploying VoIP is one thing; another is effectively securing it in such a way that it is at least as reliable as the PSTN. In many cases, adequate security in the network is the deciding factor for an organization considering VoIP deployment. You need to know what types of attacks VoIP networks are most susceptible to and what points in your network are most vulnerable.

VoIP Peering

I am convinced that in a few years, virtually all VoIP service providers will peer with each other in order to save money. This largely misunderstood shift in telecom has potential to change the way VoIP works. Peering is the concept of interconnecting networks allowing IP and subsequently, VoIP traffic to be carried between service providers and companies without the need to pay a middle-man, or in this case, an additional 'long-distance' service provider. By using session border controllers placed neatly between the service providers, you can provide a 'translation service' between your caller and the recipient who uses a different provider.

UNE-P To VoIP

Yesterday's CLECs exist because of a FCC ruling called unbundled network access-platform, or UNE-P for short. This rule specifies the rates that incumbent carriers can charge CLECs to lease their lines. Recently, the FCC has rethought this concept allowing ILECs to raise these rates and subsequently reduce competition severely. UNE-P's demise is VoIP's gain, but the shift is not without it's challenges for CLECs. The biggest question for CLECs now is whether you should seek to build your own VoIP infrastructure or purchase/lease service from a new breed of wholesale carriers.

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Rich Tehrani

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By John Cimko

VoIP And CALEA

The FCC, responding to a petition filed by federal law enforcement agencies, has proposed to apply the Communications Assistance to Law Enforcement Act (CALEA) to certain types of Voice over Internet Protocol (VoIP) (define - news - alert) service. The FCC's proposal raises two central questions. Has the FCC presented sufficient legal arguments supporting its tentative conclusion that CALEA covers VoIP? Should Congress get back involved to resolve key policy issues?

CALEA requires telecommunications carriers to design their services and facilities to enable law enforcement agencies armed with court orders to intercept calls and obtain call identifying information.

Before imposing CALEA requirements on VoIP, the FCC must make the legal case that Congress intended CALEA to reach Internet services such as VoIP. But Congress exempted Internet-based and other information services from CALEA. The FCC, however, has tentatively concluded that this exemption does not apply to VoIP because the statute permits coverage if a service constitutes a replacement for a "substantial portion" of "local telephone exchange service."

The FCC believes this statutory language means that, if a VoIP service replaces portions of a subscriber's plain old telephone service, then the VoIP service is subject to CALEA. The agency concedes the statutory language is ambiguous, but argues that its interpretation is reasonable because its "functional" test is consistent with CALEA's goal of preserving law enforcement surveillance capabilities in the face of changing technologies.

The FCC's approach raises two problems. First, there is some legislative history suggesting the "substantial replacement" test applies geographically, not based on the functional nature of services provided to individual customers. In other words, a VoIP service would not be subject to CALEA unless it had substantially replaced local exchange service throughout a geographic area, such as a state. This geographic test, some parties argue, serves the congressional objective of benefiting competition, consumers, and technological innovation while still preserving surveillance capabilities.

Second, and more fundamentally, some parties contend that the FCC is wrong in assuming that the "substantial replacement" test can trump the statute's information service exemption. These parties suggest that, once a service fits under the exemption, Congress did not intend to permit the FCC to bring the service back under CALEA's coverage.

If the FCC persists in its interpretation of the statute, the order it eventually adopts will likely face court challenges. FCC Commissioner Michael Copps already has complained that the FCC's tentative conclusions "stretch the statutory fabric to the point of tear." A court battle could be avoided, however, if Congress decides to revisit CALEA to address the VoIP issues. Reauthorization of parts of the USA Patriot Act this year could serve as a vehicle for Congress to take up these CALEA issues.

One reason Congress should intervene is that, if industry players and observers are correct, VoIP may be the "killer app" that will reshape telecommunications in a way rivaling the invention of the telephone. A recent study by Atlantic-ACM, for example, predicts that the retail VoIP market will grow at a compound annual rate of 64 percent through 2009 and will reach $20.4 billion in annual revenues by 2009. If the Internet eclipses the circuit-switched network as the voice communications backbone, then congressional action at the threshold of this sweeping change could be the best means of ensuring that VoIP fulfills this promise in a way that realizes the potential of the Internet, while also retaining viable means for court-sanctioned surveillance.

Another reason for congressional action is that there are a host of policy issues that Congress, rather than the FCC, may be in a better position to sort out. Here are some highlights:

How should surveillance needs and privacy interests be balanced? There are growing concerns in some quarters that personal privacy may take a back seat to the government's interest in maximizing its surveillance capabilities. Congress could determine whether additional statutory ground rules are needed to guard against problems such as inadvertent interception of data packets that are not covered by court orders.

Are there any options for dealing with offshore VoIP providers? It may not do much good to subject domestically-based VoIP providers to CALEA requirements if offshore providers can serve the U.S. market but escape these requirements.

Have law enforcement authorities shown an urgent need to extend CALEA to VoIP to aid the war against terrorists? In 2003, a total of 1,442 court-authorized wiretaps were granted, and 1,104 of these (76.6 percent) were for crimes related to narcotics. Wiretaps for terror-related activities did not make the tracking list. Congress could evaluate the claims of urgency, and then design CALEA requirements that meet law enforcement needs without compromising privacy interests, competition, or technological innovation.

Finally, how much leeway should VoIP providers have in designing software and facilities to comply with CALEA? Some industry players worry that law enforcement authorities will insist on a "one size fits all" approach because this makes law enforcement's wiretapping job less burdensome. But this could increase costs and stifle innovation. Congress could decide whether current CALEA protections regarding system design should be strengthened to meet industry concerns. IT

John Cimko served for fifteen years at the FCC, and currently practices law at Greenberg Traurig LLP in Washington, D.C. The views expressed are solely those of the author and should not be attributed to his firm or its clients. For additional information, visit the firm's Web site at http://www.gtlaw.com.

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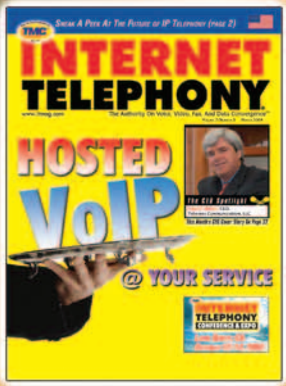

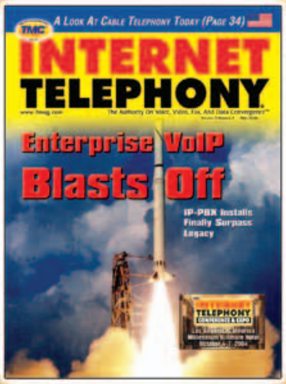

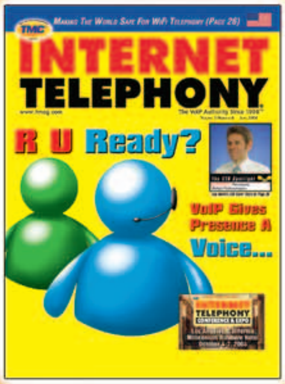

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Will Voice Over WLAN Go Mainstream In 2005?

Seven Key Deployment Issues To Consider

By Lawrence Gary

We humans are a mobile species and, thanks to the interoperability brought about by the 802.11 standards for wireless LANs (WLANs) (define - news - alert), we've wasted little time in shedding the wires that have bound us to our data networks in the home, at work, and on-the-go. So-called WiFi (define - news - alert) networks based on those 802.11 standards are popping up everywhere. In the U.S. alone, more than 12 million homes have them; almost half of all enterprises have installed them in some limited fashion; and nearly 20,000 public "hotspots" are now to be found in airports, cafes, and fast-food outlets.

As this WiFi phenomenon has grown, another disruptive technology has emerged alongside it: Voice-over-IP. VoIP has turned the telecom industry on its head by hollowing out time-worn business and pricing models. Now accounting for just one percent of phone users worldwide, VoIP is expected to capture more than a quarter of all phone traffic by 2008, with much of that traffic effectively free between users sharing the same network. Interestingly — and not so coincidentally — 20 percent of new wireless installations are anticipated to be driven by VoIP adoption by year-end 2005.

It seems inevitable then that these two technologies would merge to create a new application: Voice-over-WLAN. VoWLAN can enable businesses to respond to customers or clients faster, resolve issues more quickly, accelerate processes and decision-making, and reduce errors caused by miscommunications (i.e., "the telephone game"). It can even increase job satisfaction by reducing phone tag and enabling employees to respond better and faster to customer or client issues.

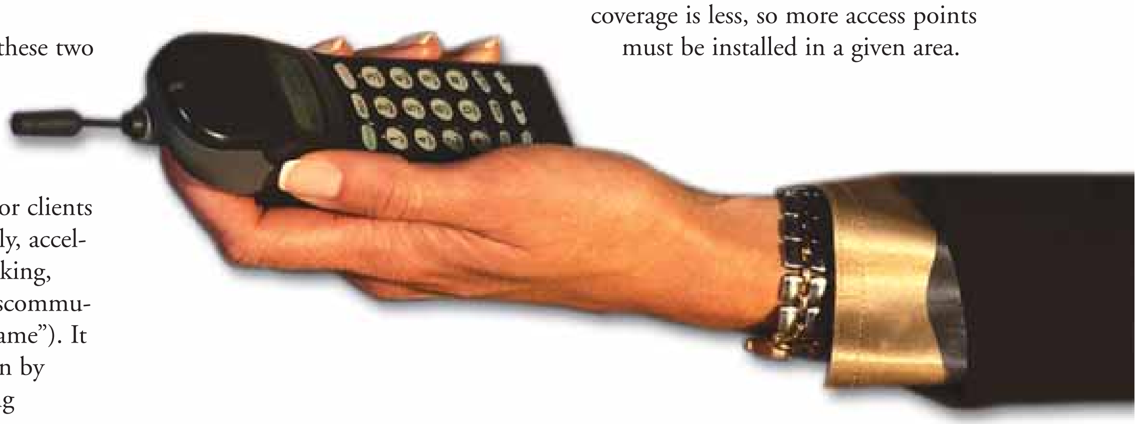
Prepping For VoWLAN Deployment

Before considering the key issues involving a VoWLAN deployment, it's important to understand some WLAN basics. WLANs operate in the unlicensed frequency bands the ISM (Industrial, Scientific and Medical) band at 2.4 GHz and the U-NII (Unlicensed National Information Infrastructure) band at 5 GHz.

Three variants of the 802.11 standard — b, g, and a — are used in WLANs, with 802.11b products the first to emerge several years ago. It supports three independent 11 Mbps channels at 2.4 GHz and uses direct sequence (DS) and frequency hopping (FH) for modulation. Although DS allows higher data rates than FH, it also uses more power in mobile client devices.

Also operating in the 2.4 GHz band and backwards compatible with 802.11b is the 802.11g standard, which can deliver up to 54 Mbps and is fast gaining momentum as single-chip implementations of it are being used in laptop PCs. It uses Orthogonal Frequency Division Multiplexing (OFDM) for modulation.

The 802.11a standard operates in the less crowded 5 GHz band, uses OFDM and can achieve data rates up to 54 Mbps. Its downside is that it's not backwards compatible with "b" and its radio coverage is less, so more access points must be installed in a given area.



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Voice Over WiFi: Four Challenges, Countless Opportunities

By Robert Sparks

The emergence of voice over WiFi (VoWiFi) broadband wireless stands as a logical component of a broader goal now shared by most buyers and sellers of communications: to liberate services from the limits of a single delivery system and make any given IP application or service available to paying customers wherever they are, via the best available delivery network at hand, and featuring emerging VoIP telephone calls.

WiFi is particularly attractive to IP applications users for its enterprise LAN-like speed and quality — virtues that make WiFi particularly accommodating to VoIP. As mobile manufacturers deliver handsets capable of accessing both WiFi and cellular networks, VoIP subscribers could soon begin to enjoy premium quality connections wherever they are available.

However, a handful of substantial hurdles must be overcome to make IP-based VoWiFi practical, available, secure, affordable, and easy to use. Most of these hurdles can be largely surmounted over the next 12 months, given the marketplace urgency to do so. As they are surmounted, VoWiFi will present opportunities to transform the ways people use networks and devices.

In the meantime, the industry must overcome challenges in four particular areas: Identity Sharing, Ubiquitous WiFi Network Access, Seamless Roaming Across Unlike Networks, and Device Capability Limits.

Identity-Sharing Mechanisms

In VoWiFi, as with any other networks, users calling each other must be enabled to securely authenticate their identities to both service providers and each other. Yet WiFi's highly diversified ownership and current dearth of inter-operator business relationships preclude any simple application of authentication methods or technologies from other types of access networks.

In traditional phone networks, subscribers delegate to the service provider authority to accurately identify any caller or 'callee' through the service provider's own contracts to share that information with other service providers. This traditional caller implicitly trusts that "Caller ID" information is accurate, primarily because the service provider owns the physical drop into the subscriber's premises and so is assumed to be the only entity using that connection.

However, because IP services are decoupled from physical networks, trust based on a physical network connection no longer applies. Encrypting presence, instant messaging (IM) and voice are the challenges. Which party is to take responsibility for making the assurance? What are the means to establish trust?

Web certification mechanisms provide one model through which third-party certificate authorities provide secure-site certificates based on trusted route. However, while this model works well with a limited number of fixed location servers, it is not designed to tackle a far greater number of individuals who are not tied to a single location. A VoIP phone will effectively need to attach a certificate to a person rather than a location.

Promising answers lie with the IETF's current work on SIP Identity and SIP Certificates. SIP Identity allows a server or end point to make an assertion that is cryptographically strong — a solution. At the same time, aug-

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The most important goal for deploying an enterprise VoWLAN is to deliver the voice quality, reliability, and functionality similar to what users expect from their wired business phones. Here are seven key issues to consider in providing a VoWLAN application that meets their expectations:

WLAN Coverage

Just as wired LANs must be upgraded for VoIP to account for QoS, reliability, and security concerns, chances are that an existing WLAN will also need upgrading. For starts, data-only WLANs are not designed for mobile use of data devices. VoWLAN applications, on the other hand, need to be designed for mobile use and need blanket coverage of the entire premise, not just cubicles and offices but lobby areas, stairwells, conference and break rooms, even bathrooms.

Also, mobile VoWLAN phone users will typically walk while they talk, moving among various access points (APs). And unlike data devices that are held or used at arms length or greater distances from the body, users tend to hold phones to their ears, so additional radio signal attenuation is introduced.

For all these reasons, WLAN APs must be positioned with sufficient overlapping coverage throughout the operating environment to eliminate dead spots between them. Then, adjacent APs need to use different radio channels to avoid interference between them. The 802.11b standard provides three non-interfering channels (1, 6 and 11 for North America), so mapping AP coverage must account for APs within range of each other to be set to non-interfering channels.

In addition, all APs should be set to the same power output and data rates. Variances in power could cause interference between a higher and lower powered AP. Differences in data rates could result in wireless phones may not associate with the closest AP if a more distant one supports a higher data rate.

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A comprehensive site survey can help optimize all these factors.

WLAN Capacity

While data traffic tends to be sporadic and bursty, voice QoS can't tolerate delays, so it's critical to plan WLAN data throughput capacity carefully.

The number of simultaneous calls a single AP can support varies depending on the codec and data rates used. For example, an 802.11b AP using the common G.711 codec can generally support up to 12 simultaneous calls at 11 Mbps but only seven calls at 2 Mbps. The G.729 codec can support roughly 50 percent more calls at these respective data rates.

Overall, WLAN needs to be engineered for overall voice usage (measured in units of Ehrlangs, with a single Erhlang equal to the traffic generated by a single phone call lasting one hour) weighed against the probability of call blocking. If an AP can support a maximum of 12 simultaneous calls, then moderate calling intensity (0.15 Erhlangs) with one percent blocking probability would translate into about 40 users supported per AP.

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http://www.xten.com

Wi-Fi Alliance
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IETF
http://www.ietf.org

System Infrastructure

In addition to APs, a VoWLAN system requires Ethernet switches, a VoIP gateway, a WLAN controller, a PBX to the PSTN and messaging servers, CAT 5 10/100 Base-T wiring, and hard or soft phones for end-users.

At the physical layer, Ethernet switches are recommended over hubs to allow adequate bandwidth and limit traffic collisions. IP addresses can be assigned to handsets either statically or dynamically, but if the latter, a DHCP service is needed.

Subnets are logical boundaries between network segments that can terminate an active call for all practical purposes should the handset's user roam beyond the originating subnet. This issue is addressed by a new device called a WLAN controller that enables fast, secure handovers across subnets and provide needed scalability for large physical environments.

End-user devices need to be selected depending on the environment and business requirements. "Hard" phones tend to share the form factors of cell phones. But some are hardened for industrial purposes. And some have push-to-talk capabilities. At a minimum, they should have a clear display, multichannel capabilities and long battery life. "Soft" phones are software clients for laptops or PDAs that provide VoWLAN connectivity.

Quality Of Service

Whether deploying a VoWLAN application over a voice-data WLAN or an all-voice WLAN, treating all packets equally will result in jitter and delay in voice quality, so a critical provision for voice packet prioritization must be made.

Early 802.11 standards lacked any QoS mechanisms, so vendors hatched their own proprietary schemes. Soon the 802.11e protocol will be released which should allow for greater interoperability (and lower prices) among vendor equipment.

All in all, managing QoS is all about optimizing the balance between band-

width, signal strength, and device power conservation while maintaining seamless, full-featured enterprise communications with minimal jitter, delay, or packet loss.

Security

WLANs have long had a reputation for being inherently insecure, but despite security shortcomings in early standards and implementations, vast improvements in the encryption of air interfaces and authentication mechanisms have made WLANs secure for both voice and data across the enterprise.

While WPA and WEP have provided interim solutions, 801.11i has formally standardized WiFi security. In addition, wireless users can be placed into a separate security domain similar to users connecting remotely via VPN mechanisms.

Availability And Reliability

Speaking of end-user expectations, VoWLAN applications must provide the same high availability and reliability of wireline communications. All network components must have a fail-over strategy in case of an outage, and the network itself must have a fallback in case of power loss. Handset power consumption and battery life also need to be accounted for.

Administration

A VoWLAN application needs proper ongoing administration to ensure the continuing optimization of all the previous considerations. A WLAN administrator's toolkit should include the means for hardware and device management as well as operational performance and troubleshooting. A combination of tools provided with system components and from third-parties are required.

With these deployment issues so stated, the allure of VoWLAN applications to a network administrator may have dimmed a bit. And let's face it: VoWLANs may not be for every enterprise — today. But they will be soon

menting the public key infrastructure (PKI) used by the Web, SIP Certificates would allow a subscriber to obtain a certificate from his service provider. In this scenario, the service provider knows that his subscriber is who he/she claims to be, and on that basis provides a certificate for the duration of an authenticated connection. Someone wishing to call this person securely can obtain this certificate from the service provider by subscribing to it using the SIP Events mechanism.

Ubiquitous WiFi Network Access

Currently, as end users move in and out of various WiFi network service areas, they are forced to manually log in to each new network. Such users engaged in a VoIP call would experience a dropped call, forcing them not only to dial again, but first to log in to the new network.

Cellular carriers have largely solved a similar cell-to-cell hand-off through a combination of home location register (HLR) technologies and provider-to-provider business arrangements, thereby enabling seamless national roaming across networks owned by different providers.

WiFi service providers will not be afforded a similarly elegant solution to provider-to-provider roaming, particularly because there are many more providers operating hotspots than there are cellular providers nationwide and worldwide. Pervasive and proactive business-to-business engagements will be more critical than technology in forging a common single-log-in solution.

Seamless Roaming Across Unlike Networks

Cellular carriers have proved that telephone call set-up signaling can be made to seamlessly hand off a call in progress from cell to cell for customers on the move. Such a seamless hand-off must be extended across unlike networks, if cellular subscribers are to be afforded the opportunity to enjoy higher quality WiFi connections wherever they are available.

Handsets must be able to detect WiFi services, and providers must make the signaling work. Building blocks do exist, but research on this remains incomplete.

Again the answer may lie as much with business arrangements as with technology. Because HLR technology does not apply to IP, carriers must work to define how a VoIP carrier may move a call across a cellular or WiFi boundary and what that will look like. Different segments of the call might be billed separately. Whatever mechanisms are applied behind the scenes, the key will be to make it appear to the subscriber that he/she is having one conversation.

Device Capabilities

Wireless industries are asking more and more of portable handset devices that are compute and power limited. Providing identity certificates and real-time detection and analysis of networks will challenge the current generation of devices. Despite promise at the high-end of these devices, the average person on the street will demand an affordable device with more power than is available to date.

Handset manufacturers will have to meet this challenge by wrestling with the tradeoffs between central processor unit (CPU) cycles and battery time. The technology must be evolved to provide the kind of portable computing these applications will demand by delivering the greatest amount of CPU cycles at the lowest consumption of the battery and at the least cost to users.

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On The Other Side

As the industry breaks through these barriers, it also must develop and standardize protocols to leverage a particularly promising opportunity: integration of presence to empower subscribers to know a person's availability, capabilities, and environment.

Action choices multiply through the knowledge that someone is on the network but in a meeting, or on a handheld device rather than a laptop with a multimedia coder/decoder. Such capability status might be indicated by a power bar similar to signal strength indicators common to today's cell phones. Through such knowledge, users can be afforded choices to accept or defer a call, to talk with a client over an optimal or sub-optimal connection, to click or not click on an ad for information on a local restaurant. A presence server might auto-conference board members, family members, or members of a fan club who have been waiting for a moment of common availability to discuss a decision.

With presence-enabled VoWiFi, real-time interactive communications will expand in dimension, challenging service providers to develop and package presence capabilities that appeal to — and raise the expectations of — business users, consumers, advertisers, and other stakeholders in their services. Raising and meeting these expectations lies within the industry's grasp. IT

Robert Sparks is the CTO for SIP softphone provider Xten Networks, Inc. For more information, please visit the company online at http://www.xten.com.

enough as more and more WLANs are deployed in lieu of wired LANs, as buildings are remodeled and new ones erected without end-point wiring, as workers become ever more mobile, and as WLAN system elements are further standardized and costs are driven down. Will 2005 be the year? Maybe not for full-blown ubiquity, but certainly it'll be much further on its way. IT

Lawrence Gary is product manager, desktop products at Siemens Communications, Inc. For more information, please visit the company online at http://communications.usa.siemens.com.

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2004 Internet Telephony Product Of The Year

It's the most wonderful time of the year! A time of lists. A time for reflection. A time for making resolutions. And of course, a time for **INTERNET TELEPHONY**® to look back on the past year and offer up our annual Product of the Year winners.

2004 was a very exciting year in the VoIP industry. I remember having a conversation at Internet Telephony Conference and EXPO back in February 2004, where we were trying to put our finger on something, trying to figure out if the warm breezes we were feeling were a hint of better things to come or just the warm weather of Miami getting the better of our imaginations. We settled on the fact that several big announcements in December 2003 might set the wheels in motion for a decent year to come. Well the wheels spun faster and faster and 2004 ended up being a banner year for our industry. Major announcements in the residential space from telecom giants AT&T (quote - news - alert), SBC (quote - news - alert), and Verizon (quote - news - alert) were just the tip of the iceberg. New products aimed at carriers and enterprises alike came to light and drove the VoIP market further in to the mainstream consciousness than ever before.

Many of the companies present on this year's winner's list were in hiding for the past few years, understandably so. The economic conditions were not ripe for maintaining a very public profile, so companies hunkered down in their development labs, readying products in the hope that industry conditions would turn favorable once again. And turn favorable they did. The products and services represented on this list are all the result of hard work and innovation and nose-to-the-grindstone dedication during the dark days of 2002 and 2003, and the companies receiving this award should feel very proud of their accomplishment. These vendors spent 2004 building momentum, and I have no doubt that 2005 will be a time for them to build on that energy and a time for them to keep the promises they made to themselves and their employees to succeed in VoIP.

I urge my readers to use this list as a starting point when looking for solutions to solve your particular needs. In parting, I implore you to do your homework. Research these companies thoroughly, follow up with them. Call them, meet with them, demo their products, and most importantly: Check out those customer references!

-Greg Galitzine

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Canadian City Adopts IP Telephony

Salmon aren't the only ones migrating to the riverside city of Coquitlam, which lies just 30 minutes upstream from Vancouver, British Columbia. The city, which takes its name from the Coast Salish native word for a little red sockeye, is spawning double-digit growth every five years in the residential, commercial/retail, and industrial arenas.

"Through partnership in the Industry Canada Smart Choices project and other initiatives, Coquitlam is capitalizing on emerging technology solutions to enhance all facets of customer service and operations," says Jon Kingsbury, mayor of Coquitlam. "We have invested up-front in modern infrastructure that paves the way for continued growth and development while retaining the immediacy and responsiveness you would expect from a small town."

If you're a Coquitlam resident wanting to check the status of a building permit, reserve a park shelter or sign up for swimming lessons, the most visible "emerging technology solution" you'll see is the city's new IP telephony (define - news - alert) communications system.

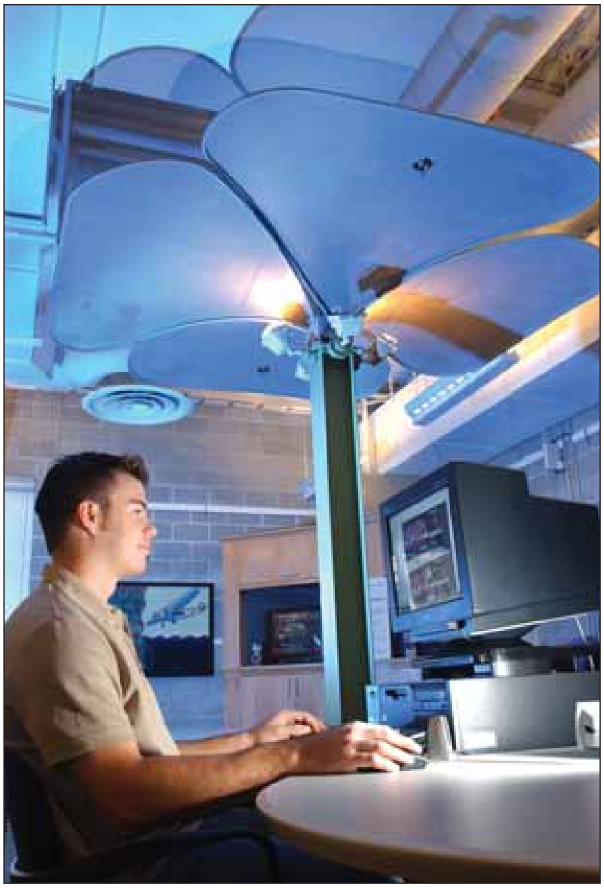
Strategies For An On-Demand World

Until May 2003, Coquitlam city departments relied on a mix of telephony services from various sources. Most users were served by a 10-year-old Centrex service from an external provider.

With this approach, city employees had a variety of different phone sets and services. It was costly and time-consuming to move, add, or change services, and some employees didn't have access to essential features, such as voice mail.

The city's five contact centers used automatic call distribution (ACD (define - news - alert)) systems, which would place calls in queue and send them to the first available agent, not necessarily the best person to handle their call. There was no intrinsic auto-attendant capability, so a patchwork of add-on systems provided basic interactive voice response (IVR (define - news - alert)) capability.

"Although we didn't have to buy and maintain customer premises equipment except our firewall, Centrex was fairly expensive for us in terms of annual cost," said Rick Adams, manager of information and communications technology. "We had to rely on the service provider to do moves and changes



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Using IP phones across the entire enterprise delivers instant mobility.

for us, so we had to factor in a long wait. The other big challenge was that we had such a variety of phones that training and movement of staff were very difficult."

In 2002, the city's information and communications technology team set out to find a solution that would reduce operating expenses while enhancing collaboration among city employees and responsiveness to city residents.

"We chose an IP infrastructure because we needed to be dynamic," said Michel Labelle, ICT supervisor of Network & Client Services. "We were looking for an architecture that would allow us to very quickly adapt to community needs and demands."

The communications system needed to span three main campuses with diverse operational needs. In order to do this with an IP network, it had to provide the necessary quality of service (QoS) to carry voice. It had to provide the reliability users had always expected from the public phone system and it had to make geographic distance between these buildings and campuses invisible to callers.

In 2000, the City of Coquitlam took its vision for enhancing community service through technology to the 'Smart Communities Project' — a competition offered by the Government of Canada to stimulate e-community initiatives — and won a matching grant to implement their ideas.

"Nortel became our technology partner of choice because they are a stable and reliable vendor with all of the cutting-edge technologies we need to make this vision a reality, and take us into the future," Adams explained. The details of the deployment can be found in the sidebar entitled "*Coquitlam Deploys End-To-End Nortel Solution*."

Advantages

"The solution is an outstanding business value for us," said Adams. "It makes staffing moves and changes as easy as plugging in a PC to the Ethernet network. It is so easy to administer and manage; we brought voice management in-house and saved $300,000 in outsourced voice service fees. Using IP phones across the entire enterprise delivers instant mobility, plus easy training for our 18-person IT support department."

"The primary advantage over everything I've ever implemented in my 16-year career is that an IP infrastructure is incredibly flexible," said Labelle. "We have a demonstrable ROI for VoIP (define - news - alert) with Symposium; we will save approximately $500,000 a year, all told — but that wasn't the major driver. With our vision of flexibility, we need the ability to relocate people and staff at a moment's notice."

For instance, under the old system, when renovation work had sparked a small fire in an emergency dispatch center, it took about three hours and 'quite a panic' to relocate dispatchers to an alternate location and re-route their calls. "Today, this would take us only

Coquitlam Deploys End-To-End Nortel Solution

The following products all play a role in the Coquitlam deployment:

• Nortel **Communication Server 1000** at each campus offers more than 450 IP PBX features. Communication Server 1000 offers no single point of failure, power fail bypass, and multiple tiers of optional redundancy.

• Nortel's **CallPilot** unified messaging on the Communication Server 1000 delivers centralized voice mail services, auto-attendant, voice menus for self-service call routing, voice-activated prompts, centralized directory services, and mobile voice mail access.

• A Nortel **Symposium Express Call Center** at the main operations center supports the city's five call centers and 50 agents with intelligent call handling, skill-based routing of calls to best available agents, and comprehensive performance and utilization reporting.

• A Nortel **Passport 8600 Ethernet Routing Switch** at the main operations center provides gigabit switching for the LAN fiber backbone. Designed for delay-sensitive, jitter-sensitive applications, the Passport 8600 satisfies the QoS demands of IP telephony.

• Nortel **i2004 Internet telephones** were issued to every desktop — 700 of them — to give users convenient, uniform access to Communication Server 1000 and CallPilot features. These standards-based IP telephones connect directly to the Ethernet LAN and are managed from the central operations center. No on-site visits are required to add or change features or users.

• Nortel **BayStack 460 Power over Ethernet** switches at all three campuses send power to the IP phones over the Ethernet links. With power delivered from a central switch, phones are guaranteed to be up during a power outage.

• Nortel **Contivity 1010 Secure IP Services Gateways** provide affordable, secure remote VPN access for small sites that previously were not even on the city's WAN. Now, these remote sites have secure connectivity to the Contivity 1700 Secure IP Services Gateway at the City Hall, with all the same features that users enjoy on the main campuses.

minutes," Labelle said. "Any site can quickly be re-architected to become the equivalent of any other site. By the time staff gets to the alternate site, their services are there waiting for them."

Not only can these relocations be managed remotely without on-site visits, the system constantly updates a central calling directory. "With a staff of 1,100, plus seasonal variances and the usual amount of turnover, we never kept up a central call list before," Labelle recalled. Now employees always have a current call list and can connect to anyone in the city network without help from an operator.

Here's a sampling of other ways the new network has reinvented communications in city divisions:

• The Leisure and Parks department has contact center agents distributed in nine buildings across the city. With the new system, incoming calls can be automatically routed to the most knowledgeable person anywhere, making it easy to provide the fastest response possible.

• The Operations department handles everything from permits and requests for filling potholes to more urgent inquiries such as non-working traffic signals or burst water mains. These calls can now be prioritized and sent to the most appropriate responder, even though Operations employees are located all over the city.

• Collections/Taxation operates a call center to support utility and tax collections. During the height of the busy period, callers can either remain in queue (and hear custom, informational announcements that save clerks' time) or mark their places in queue, so they don't have to wait as long.

More Than A Phone System

Because Coquitlam bet on convergence rather than extending investments in a 10-year-old technology, the infrastructure is in place to support exciting new initiatives. For example:

• Customer service enhancements. Computer-telephony integration (CTI) will deliver screen "pops" of the caller's history to agent desktops, so agents can resolve inquiries quickly and accurately. Residents can do business with the city on the Web, such as register for city programs or apply for licenses. Emergency notifications can be "pushed" or broadcast to residents by phone or e-mail.

• "City on dark fiber." With ample unlit capacity available, the city's new fiber WAN can be used by schools, enterprises, and service providers to offer a virtually unlimited menu of value-added services.

The city is also planning to use videoconferencing, unified messaging, wireless LAN access for staff, public wireless "hot spots" in community centers, and tablet technology for warehouse inventory management — all of which can be readily handled on the new Ethernet/IP infrastructure.

"The city of Coquitlam strives to be a leader and innovator in delivering municipal services, and the key to this mission is staying on the leading edge," said Mayor Kingsbury. "The advanced technology and breadth of Nortel's solutions have enabled us to engage with our customers at every level, helping to make the city a showcase model for e-community success in Canada." IT

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success
is to know something
nobody else knows."

—Aristotle Onassis

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It's VoIP 2.0 Time In Miami!

INTERNET TELEPHONY Conference & EXPO is proud to make its return to Miami this February. From February 22–25, anybody who's anybody in VoIP will be at the Hyatt Regency Hotel in Miami, FL to see what's new on an Exhibit Hall floor packed with vendors showing off their wares and to learn all about where this exciting technology is headed at a conference that is unrivaled in its depth and educational content.

The theme of this year's event is VoIP 2.0: The Evolution of IP Telephony. And that theme carries across the conference sessions targeted at an audience that will include Service Providers, Enterprises/Government agencies, and Developers. The conference will also feature in-depth content outlining the opportunity for resellers in our industry. Here's a look at just some of the more exciting subjects we'll be tackling in this year's conference.

Open Source Telephony Workshop

Open source telephony solutions are flourishing from the service provider to the enterprise. Behind the scenes, more and more service providers are using Digium's Asterisk to run their operations and support thousands of customers. Enterprise users are in the testing phase now and word is the early adopters are thrilled with Asterisk. INTERNET TELEPHONY Conference & EXPO Miami 2005 features a new workshop focused on helping developers take advantage of this platform to create new high-demand VoIP applications.

Regulation & Taxation/E-911 Summit

The FCC has vowed to do its best to keep from taxing VoIP, but politicians are screaming to regulate it, tax it, and secretly, even kill it. One problem is that the Universal Service Fund is drying up as expected. As of late 2004 the company that runs the fund, Universal Service Administration Corp. has asked regulators to increase the percentage of long distance revenue that service providers have to pay from 8.9% to 12.5%! Others say that amount needs to be doubled to 25%. Potential regulation combined with E-911 and CALEA concerns mean the road to VoIP 2.0 will have to be routed through Washington, D.C.

SIP Workshop

There remains little doubt regarding the significance of SIP — Session Initiation Protocol (define - news - alert) — and the profound effect it has had on the IP telephony industry. SIP is an IETF signaling protocol for Internet conferencing, telephony, presence, events notification, and instant messaging, and many developers are beginning to realize the tremendous potential of writing applications to take advantage of this standard. The SIP Workshop is a full-day conference designed to educate conferees on the subject of SIP. Special thanks go out to the SIP Forum and SIPCenter.com for sponsoring this event.

UNE-P To VoIP Summit

Yesterday's CLECs exist because of an FCC ruling called unbundled network access-platform, or UNE-P for short. This rule specifies the rates that incumbent carriers can charge CLECs to lease their lines. Recently, the FCC has rethought this concept allowing ILECS to raise these rates and subsequently reduce competition severely. UNE-P's demise is VoIP's gain, but the shift is not without it's challenges for CLECs. Our conference features an entire day of sessions designed to help you seamlessly transition to VoIP and begin making money immediately.

VoIP Peering Summit

This largely misunderstood shift in telecom has potential to change the way VoIP works. Peering is the concept of interconnecting networks allowing IP and subsequently, VoIP traffic to be carried between service providers and companies without the need to pay a middle-man, or in this case, an additional 'long-distance' service provider. By using session border controllers placed neatly between the service providers, you can provide a 'translation service' between your caller and the recipient who uses a different provider. The new VoIP Peering Summit at INTERNET TELEPHONY Conference & Expo explains both the

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network and business process implications of peering.

VoIP Security

Concerns over security in the VoIP network are exacerbated when you add mobility to the equation. Remote access over WiFi and WiMAX networks — a principle benefit of IP telephony — ups the ante for administrators charged with providing secure, seamless access to corporate data. The enterprise/government solutions track at INTERNET TELEPHONY Conference & EXPO prepares you to head off viruses, attackers, and hackers before they get to your network.

The Reseller Opportunity

There are two huge opportunities in VoIP for resellers: equipment and service. The equipment play is obvious; more and more VoIP products are being sold on a daily basis. Corporations are snapping them up at a rapid clip. Equally impressive is the opportunity to sell VoIP service. More service providers mean more opportunities to help these providers sell services to new customers. This is the best time to be in this business and resell VoIP. In fact, no other technology today is seeing as much growth and is generating as much spending as VoIP.

IP PBX Certification

This year, the organizers of the conference have supplemented an already great program with what is arguably one of the more innovative and significant additions to the learning experience: TMC University. TMC University's IP PBX Certification program is the only Independent certification program of its kind validating attendees' competency in IP PBX selection, deployment, implementation, and management.

Some of the benefits afforded by the IP PBX Certification program include:

- Independent accreditation for completing the curriculum;
- Independently certified evidence that you possess competencies in IP PBX selection, implementation, and management;
- Addition of an impressive certification from a respected source to your resume;
- Enhanced chances for a promotion; and
- Potentially lucrative consulting opportunities.

Keynote Speakers

TMC has yet again assembled a stellar lineup of keynote speakers who will share their vision with EXPO attendees.

- Stef H. van Aarle — VP Marketing & Strategy, Lucent
- Gary Barnett — President & Chief Executive Officer, Aspect
- Edward Cespedes — President, Voiceglo
- Phil Edholm — Chief Technology Officer & VP of Network Architecture, Nortel
- Kenn Epps — CEO, Baypackets
- Cathy Martine — Senior Vice President, AT&T Internet Telephony
- William Tauscher — President, CEO & Chairman, Artisoft
- Andy Voss — President & CEO, Sansay, Inc.

Register Now!

If you are interested in learning more about the IP telephony industry or if you are interested in becoming an indispensable part of your company's IP telephony decision-making team, then Internet Telephony Conference & EXPO — February 22–25, 2005 at the Hyatt Regency Miami — is the one event you can't afford to miss. Visit http://www.itexpo.com and register now to reserve your seat at the industry's leading conference. IT

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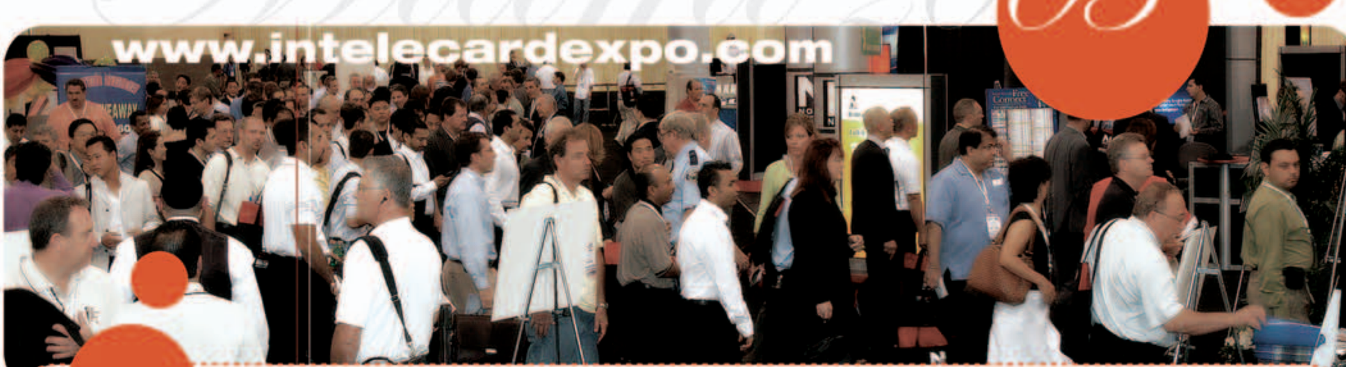

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INTERNET TELEPHONY® EDITORS' CHOICE AWARD 2005

Zoom X5v

Zoom Technologies, Inc.
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Tel.: 617-423-1072
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Price: $95–$99, available at CompUSA, PC Connection, and other retailers, as well as online retailers.

ZoomTel X5v Model 5565, which states on the box, "The All-in-One Solution for DSL Users. DSL + VoIP" is a unique ATA (Analog Telephony Adaptor) within the VoIP industry. Essentially it is an ADSL modem, four-port 10/100BaseT switch/router, integrated firewall, and VoIP-capable with the device pre-configured to use Zoom's Global Village VoIP-to-PSTN termination service. You can think of it as a "plug and play" ADSL modem with "plug and dial" VoIP capabilities.

Zoom also has another X5v model without the Global Village VoIP service bundled (Model 5585/5586) for third-party VoIP service providers that support the SIP protocol. In fact, Zoom told TMC Labs that one of their main goals is to preach the gospel of "openness" in the VoIP world, avoid proprietary methods, and promote open standards and protocols, such as SIP. We should point out that the X5v 5565 "Global Village" version, which they sent to TMC Labs to test, also works with other SIP-based VoIP providers.

The ADSL X5v integrates a full-rate ADSL modem, router, firewall, four-port 10/100 Ethernet switch, USB port, and VoIP phone port into a single product, which is perfect for home broadband users who don't want to buy multiple pieces of networking equipment. Zoom claims that the integration of the voice processing with the firewall in the same router ensures reliability of VoIP calls and reduces service provider support requirements.

Zoom's TelePort feature provides a phone port that lets you use a standard analog phone for conventional phone calls over the PSTN or low-cost Voice over IP calls or "free calls" to fellow Global Village users and other SIP users. The phone port of the 5565 is an intelligent relay that allows an analog phone to place and receive both VoIP calls and calls over the PSTN. PSTN Fail-Over allows the X5v to automatically route calls over the dial-up phone network when power is lost. When using emergency dialing services like 911, it will route over the PSTN thus negating the need for E911.

The 5565 and 5566 models come with Zoom's Global Village, a VoIP phone service that lets you dial PSTN phone numbers and the calls are routed over IP to Zoom, which then terminates the call. You can look up Global Village's calling rates and dialing plans on their Web site. In fact, one unique aspect of Zoom's product is that there is no monthly commitment — you don't have to pay a monthly fee for unlimited VoIP calling. Instead, the user fills out a form on the Web site and provides a credit card, opting for a pay-per-call approach. Global Village service also offers an option for the user to pay a low monthly fee ($3.95 per month) for a United States phone number to receive incoming calls from the public switched network. Wow! To pay only $3.95 for a U.S. number and then pay as you go? That's pretty impressive. Their rates aren't too bad either. Calls to the U.S., Hong Kong, U.K. France, and Spain are all currently $0.029/minute.

The X5v's has a built-in four-port Ethernet switch, which eliminates the need for a separate Ethernet hub for most home networks and small office environments. In addition, a USB port allows direct connection of a fifth computer to the ADSL X5v, though you can simply connect another switch/hub or wireless access point with a crossover cable. The USB port is a nice feature to

RATINGS (0–5)
Installation: 5
Documentation: 4.5
Features: 4.25
GUI: 4.5
Technical Support: 5
Overall: A-

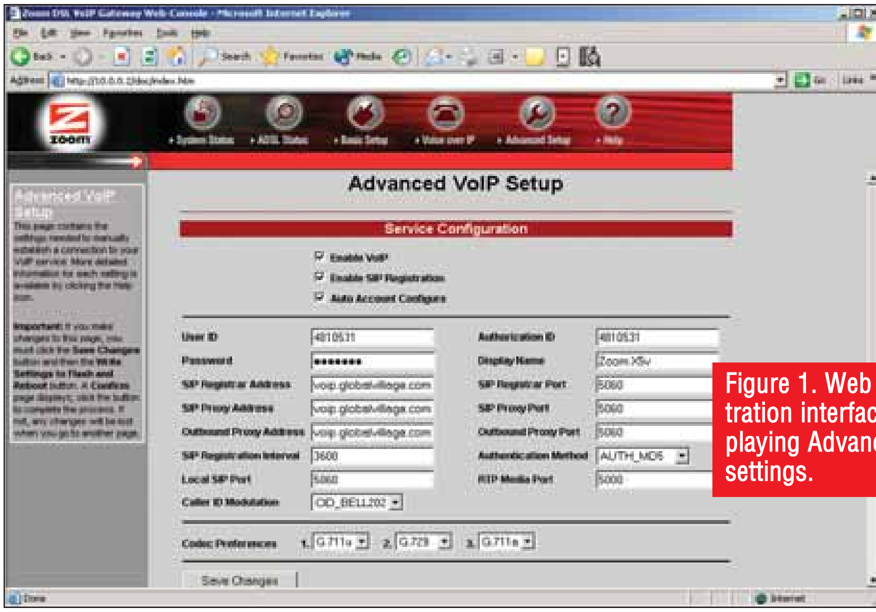

Figure 1. Web administration interface displaying Advanced VoIP settings.

have, but considering that USB cables are generally very short and the X5v may be far away, it is of limited benefit. By daisy-chaining more hubs or switches you can connect up to 253 network devices.

INSTALLATION

To install the X5v we needed an ADSL connection. Since TMC Labs has only T1 and dial-up connectivity in the labs and only cable broadband at home, we had to ask our fellow coworkers for a volunteer DSL line to play with. We found one such volunteer in the local area that uses SBC DSL and then proceeded with our testing during lunch.

Installing the X5v was a breeze, especially when using the Quick Start step-by-step guide. We disconnected the existing DSL router and plugged the phone wire into the X5v as well as connected a PC to one of the network ports. Next, we entered in 10.0.0.2 into our browser to access the X5v's Web interface. After authenticating, we entered in the DSL account information. We then tried to surf the Internet with no success.

Next, we ran the X5v's diagnostics from their Web interface and discovered that one of the diagnostics tests failed. After some troubleshooting, we decided to give their tech support a call. We have to say that Zoom's tech support was excellent — and we never mentioned we were with the press. She was able to pinpoint the problem in no time flat. It was in fact "user error" — our volunteer had mistyped the password in two repeated attempts. So if it weren't for a password typo we would have been up and running in less than five minutes — a true plug and play experience! We tested that it was working by surfing the Web and it performed flawlessly. We should mention that the Web interface was extremely easy to navigate and configure various settings, from the DSL connectivity, to the firewall settings, to the VoIP settings (Figure 1).

OPERATIONAL TESTING

The firewall was pretty powerful and includes stateful Packet Inspection, protection from Denial of Service attacks, NAT, and more. It also supports a DMZ host for special applications, such as gaming. The firewall's features seemed comparable to other SOHO firewall/routers and as such, we couldn't find any features lacking.

Since it took longer than expected to get the DSL connection working we were only able to make three test phone calls during this "lunchtime" test. The first test phone call was simply a traditional PSTN phone call by simply dialing a number. The call was connected using the DSL's PSTN connection without a hitch. Next, we made a VoIP phone call by dialing "#2035551212" to dial a cell phone number. Interestingly, the CallerID on the phone we dialed showed our seven digit Global Village number (4810531), so it took a second to get used to not seeing a 10-digit number with area code. The latency was extremely low, however the voice quality was only OK, but not great — it sounded a bit robotic and tinny as though the codec was performing too much compression. We tried a second call and the voice quality was the same. Perhaps SBC's network was a bit overloaded?

Since it may have been network congestion, we decided to make some test calls on another date. We had someone call us from another X5v unit and the voice quality was superb. Our 10-minute conversation sounded very natural with no delay or hiccups. It was equal or better than other VoIP services that we have tested — including Vonage, Packet8, Net2Phone, and more. So, the SBC DSL connection must have been congested after all during our first test.

PROS and	CONS
Built-in 4-port Ethernet Hub Easy Installation Excellent Tech Support	No support for multiple VoIP providers No integrated wireless capabilities

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To Pound or Not to Pound, that is the Question…

In order to make a VoIP call we needed the "#" prefix. We were curious if we could change this behavior so that the "#" prefix defaults to the PSTN or if we can set VoIP to be the default. For instance, if we decide to use VoIP more often than the PSTN, we'd like the "#" prefix to be used for PSTN calls instead, so we can simply dial numbers and by default the calls are router over IP. We discovered there is a "VoIP-Only Mode" which you can turn on so you don't have to dial the # first, however if you do that you can't initiate traditional PSTN calls. The unit should simply toggle what dialing method is used when dialing the # prefix.

The X5v is fully SIP-compliant and has the ability to reach other popular SIP destinations simply by hitting "**" followed by the pre-defined SIP "area code." Thus, you can dial SIPPhone, IPTel.org, FWD, etc. In addition, if you forget your Global Village number, rather than log on to the Web interface, you can simply dial # and then 1, 2, 3 and an automated attendant will respond with your phone number.

FEATURES

The X5v features Voice activity detection (VAD) and Comfort noise generation (CNG) as well as echo cancellation. The ring generation supports 5 REN (Ringer Equivalence Number), which supports more than 5 typical telephones. There is also a unique ring for an incoming PSTN call versus a VoIP call.

Voice over IP Analog Telephone Adapter Specifications

Analog Voice Port
- Type: Loop-start FXS interface
- DTMF tone detection/generation

Off-hook Detection
- V.21/V.25 Modem/fax tone detection
- Ring occurs for incoming VoIP and POTS calls

Call Control VoIP
- SIP (RFC 3261) Protocol
- Digest Authentication using MD5
- Record-Route Headers

Voice Compression
- G.729A, G.711 A-Law, G.711 U-Law Codecs
- Line-echo Cancellation
- G.165, G.168 compliant
- Eight to 128 msec configurable echo length
- Nonlinear echo suppression

Voice Features
- Dynamic jitter buffer (adaptive)
- Packet Loss Concealment
- Bell 202 and V.23 Caller ID

Zoom Router Mode
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- RIP V2, RIP V1
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- Configuration of static routes and routing table
- Configuration of NAT/NAPT
- Selection of bridge or router mode
- Specification of PPP user ID and password
- Configuration of virtual circuits.

ROOM FOR IMPROVEMENT

We'd like to see support for multiple VoIP providers all at the same time. In addition, perhaps Zoom can access the provider's current rates (often in XML format), load that info into the X5v firmware on a daily basis and then perform LCR across the VoIP service providers. This is going to happen one day — you can bank on this TMC Labs prediction. We'd also like to see a model with integrated wireless (802.11b/a/g) to negate the need to buy a separate wireless AP (access point).

CONCLUSION

Zoom's X5v is unique from other analog telephony adaptors (ATAs) in that it has a built-in DSL modem and a built-in SIP firewall. Also, many competing ATAs "lock" the device or use proprietary methods. TMC Labs loved the fact that Zoom's X5v product line was open and completely standards-based, including SIP and using standardized codecs. According to a source, they are also in discussions with Global IP Sound to possibly use their excellent GIPS iLBC codec, which just recently received standardization approval by the IETF. TMC Labs can best describe the Zoom X5v as a "plug and play router and firewall with plug and play VoIP service that uses open standards." If you have DSL broadband service in your home and are looking to easily add a DSL modem, VoIP, and a router all in one package, then Zoom's X5v is the perfect match. IT

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The Question Of Return On Investment Of VoIP

What does a developer need to consider before creating IP telephony solutions while keeping the total cost of an enterprise solution in mind?

The first consideration is the customer that the developer is targeting their application for. Depending on the company culture and current communications infrastructure, different aspects of the total solution become more critical — majoring on either a feature or a cost basis. A developer needs to ask questions and tease out the answers from their prospective marketplace, as the responses to these questions will affect not only the cost and features to be implemented but also the best enabling technology option. This article works through a number of these qualification questions and proposes the best way forward – always keeping an eye on cost and ROI.

VoIP (define - news - alert) enables the transport of voice along with data and other media over a single IP based network. Taking this into account provides some indication as to the many applications that are now emerging because suddenly the solution provider's palette contains more than just voice or just data — it consists of multiple communication methods.

This article will focus on how to create a solution depending on the enterprise's situation, and there are many considerations. Initially the key aspect is to understand how much the customer is planning to change their existing architecture. Will the customer maintain legacy equipment and cabling while adding on a new VoIP service? Alternatively, is the customer looking to move completely to a converged IP infrastructure using a single network?

'IP Ready' Architechture

If we look into this first scenario, there are still some architectural aspects to settle:

1. Is the legacy equipment 'IP ready'? A number of platforms may not have IP connectivity today, but can have a circuit board added that will introduce support. Computer telephony (CT)-based platforms are the likeliest candidate for this situation as they are the most flexible solutions in the market.

2. Is the legacy equipment upgradeable? One possibility here is to add an external gateway box to introduce a VoIP capability to the premises.

When reviewing these architectural variations we have to remember what the customer is trying to achieve. If it is to consolidate traditional TDM network trunks and data networking links to a single IP WAN (define - news - alert) connection then the options suggested by either a) or b) will do the job and the ROI calculations can be easily made. In many respects this is the easy part of the equation, as we haven't started looking at productivity benefits of VoIP — just simple 'like for like' cost comparisons. The big technical check to be made at this point is ensuring that the proposed equipment supports the two VoIP protocols present in the market today: H.323 (define - news - alert) and SIP (define - news - alert). Also worth checking is the commitment of the supplier to evolve the protocol support over time as enhancements are made.

When looking to the WAN service providers, it is usual to have a service level agreement (SLA) to specify a quality of service (QoS) to be achieved. This last point leads to consideration of the specification of any VoIP hardware introduced to the architecture. It is important to consider latency performance and ensure it meets a recognised standard such as the ETSI TIPHON rating.

All IP Solution

If the legacy equipment is 'IP ready' — we touched on this in 1.) above — the good news is that well specified CT platforms can be VoIP enabled with the addition of a circuit board and appro-

By Mike Matthews

priate software. The better news still is that by enhancing the existing platform the original investment is retained. And the stunningly good news is that a single CT platform can now connect not just to the WAN in order to consolidate traffic and save money, but also at the desktop, users can start using their new VoIP services.

This alternative is starting to sound interesting, so what other points do designers have to consider?

In creating this 'well specified CT platform' careful thought around the programming model needs to be made and most designs today are modular with the need to support multiple threads. A great asset to designers is being able to select additional CT hardware that supports more and different features like VoIP, yet easily fits into an existing application architecture as the way it is programmed — via the API — and is consistent across all CT product types.

A Twist In The Tale

So far we have been talking around hardware-based solutions and this is logical from the starting point of most businesses and indeed communications solutions of today. A more recent innovation — suitable for applications like an auto attendant that fronts and distributes businesses' phone calls — is to use host based media processing (HMP).

The HMP (define - news - alert) technique uses no dedicated specialist CT hardware and simply takes advantage of the improved processing power that is increasingly available. The connection to a communications platform running HMP will usually be via the standard network interface card (NIC) making the hardware line up a familiar one for the IT staff.

Small enterprise businesses, say less than 100 users, in an IP only situation could well benefit from a HMP-based solution. So how does the developer factor this into their equation? The same sort of programming issues need to be considered when looking at software instead of hardware — is the programming interface consistent with any existing application? This last point starts to become really important when we take a step back to look at the bigger picture for many enterprise businesses.

The most common model is one where there is at least one site with large capacity requirements — usually the head office. There are then a series of satellite offices serving customers' needs around the geography being covered. Historically this has been a challenge to hardware-based solutions with few 'high end' solutions scaling down very cost effectively. With both HMP and CT hardware products that employ consistent APIs in the tool bag, the developer now has a great opportunity to develop a single robust application that can be deployed in small to large installations. Some distinct benefits of this approach start to come through; support of the solution becomes more straightforward from front-line IT staff back through to the designer, and training of the end users becomes easier as each work station can be implemented in a consistent manner.

While these points illustrate distinct benefits we can see from a qualitative position, it becomes a harder challenge to perform the quantitative analysis. This is often the point where expert help in the form of consultancy is sought.

The Letter Q Is A Good Reminder

Having touched on a couple of the q words recently, we should take a few sentences to remember one of the early perceived issues with VoIP — speech quality and QoS. When traditional calls come over the circuit switched telephone network they use a full 64kbits/s bandwidth and employ a standard coder identified as G.711. The vast majority of early IP equipment has been sending and receiving traffic using this traditional high quality voice scheme.

There are alternatives that usually become considered when bandwidth is a more scarce resource than usual, say in an expensive connection with high call demand. The main alternative that is widely deployed is a compression scheme called G.729 (define - news - alert), which lowers the bandwidth down to 8kbps, yet retains highly intelligible speech. It is important to check that the proposed components for the design encompass key codec support along with the latency management referenced earlier. The good news for the designer is that there is widespread support for this technology making interoperability of IP-based solutions a basic expectation these days. The only negative item to report is one of cost associated with some compression schemes. G.711 is free while the others do carry a licence fee to a small number of companies involved in the original intellectual property development.

So in terms of checking out the solution's ROI, we need to ensure that only what you need is specified and this cost is another driver behind why G.711 has remained the default scheme of choice.

Summary

Time to bring together a checklist that can help improve the return on investment when designing a VoIP solution for the enterprise:

• Check the customer's intentions — is this going to be an 'IP ready' architecture or an all IP solution?

• Consider both hardware and software components in the solution mix — this could help manage cost and complexity.

• Ensure the programming interface is consistently implemented across the vendor's product range (hardware and software) to minimize time to market and ongoing application maintenance.

• Make sure both major protocols are supported and will evolve: H.323 and SIP.

• Check the specifications for QoS — pay attention to the support of standards like TIPHON (define - news - alert).

Finally, don't be afraid to discuss your thoughts and concerns with vendors of VoIP technologies. A wealth of experience already exists and working closely with vendors makes good business sense. IT

Mike Matthews is head of product marketing at Aculab. Aculab enables developers and systems integrators to produce a variety of high-performance communications solutions. For more information, please visit the company online at http://www.aculab.com.

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In Search Of Enterprise VoIP ROI?

The Contact Center Is The Killer Application

If a faster ROI from enterprise VoIP is what you're looking for, then implementing VoIP in your contact center is the answer. The VoIP contact center produces a faster ROI by reducing operating costs, generating efficiency and improving customer service — making the contact center the killer application for enterprise VoIP.

To achieve a faster ROI in the contact center, a pure IP system must be deployed. However, companies in need of extending the investment of their installed TDM switches often consider IP-enabling their contact center with a hybrid IP system. The only real benefit of IP-enablement is a savings on basic transport costs between switches. The benefits of an all-IP system are much more extensive.

Consider a typical company with three contact centers, 100 agents per center. With an IP-enabled system, potential savings are limited to inter-site telephony charges (transport costs between switches) and system administration costs. The total cost savings during the first year of deployment could be $200,000.

If that same company implements an IP-based system, not only can they experience the same cost savings as with an IP-enabled system but they can reap even greater savings from five additional areas. The cost savings of a pure IP system within the first year of deployment could amount to $3.5 million in the following areas:

• Lower cost and time of deployment for CTI applications.

• Improved agent productivity from wider deployment of CTI applications such as screen pop and intelligent routing.

• Labor cost reductions result from networking multiple centers into a single, unified pool of agents.

• Lower cost and faster deployment of new centers.

• Lower maintenance contract fees and licensing fees.

The faster that the entire contact center moves to an all-IP system, the quicker the ROI, including the realization of all of the business benefits that offers. Migrating to such a solution may mean that the most recently purchased TDM switches should be temporarily IP-enabled while some of the older TDM switches should be retired. For maximum benefit, any well-thought out migration strategy will allow the company to migrate at its own pace by moving either site-by-site or functional group-by-functional group. IT

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The faster that the entire contact center moves to an all-IP system, the quicker the ROI.

Voice over Broadband:
The Real Revolution In Personal Communications

While Voice over IP (VoIP) (define - news - alert) technology has been deployed for several years in many carriers' backbone networks, only in the past year has it captured widespread public attention. This is because carrier VoIP was completely transparent to the end user and did not change the way people used the phone. In the next few years, with the growth of voice over broadband (VoBB) (define - news - alert) services, this is likely to change. VoBB is creating a paradigm shift in the way people conduct their personal communications and represents the first true change in end user behavior since the inception of VoIP. Service providers can leverage this fundamental change to offer new services that will increase revenues and market share.

This article explains how the VoBB model differs from the Class5 softswitches that preceded it and, in essence, breaks the century-old Class5 centralized switching model. It also examines the key considerations service providers must take into account when deploying VoBB services.

Background

VoIP was launched almost 10 years ago with the introduction of consumer Internet telephony products based on the integration of phone capabilities into computers, enabling "free" PC-to-PC voice communications over the Internet. The enormous public interest and high expectations for consumer VoIP software never materialized then, mainly due to quality and bandwidth issues.

About one year later, the first telephony gateway appeared, enabling PC-to-phone and phone-to-phone calling. Increased gateway capacities, together with routing and billing capabilities, allowed for the first commercial VoIP network deployments, chiefly for prepaid calling card and international wholesale services. In these deployments, only the trunking ports were VoIP (the endpoints were PSTN). In 2003, VoIP networks carried approximately 18 percent of the world's international traffic and are expected to account for 63 percent of this market by 2008 (Probe Research).

All of the above, while critical for the evolution and development of VoIP, *did nothing to alter people's communications behavior* — in fact in the majority of cases people didn't even realize their calls were traversing IP networks. They did enjoy much lower rates for international calls using their "black" phones, in part due to carriers' use of VoIP.

Emergence Of The Softswitch

One of the industry's more nebulous terms, "softswitch" (define - news - alert) is most commonly used to describe the concept of separating call control from media processing, rather than referring to a physical entity (although many vendors use the term this way). When the first "softswitches" emerged in the late 1990's, there was a clear distinction between Class4 (Trunking) and Class5 (Access) softswitches. However, soon afterward, even this distinction started to blur. Instead of referring to the control layer, "softswitch" morphed into an all-encompassing term for next generation networks (NGNs) that included both call control and transport (gateway) layers.

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By Ari Rabban

There was also confusion between "pure" VoIP networks on the Class4 level (e.g., Deutsche Telekom's carrier services division and ITXC) and the NGN (Class5) switches, some of which included both TDM and VoIP ports.

Furthermore, some NGN solutions, such as Santera and Taqua, did not include VoIP. Instead, they were based on a "compact" TDM switch receiving aggregated "NGN TDM" traffic. Even in those solutions that supported IP, the end user experience didn't change since he/she was still using a "black phone" and receiving black phone services from the service provider. As these solutions only provided a subset of basic Class5 features, they were not really a replacement for the traditional Class5 TDM switch, rather provided carriers with a cheaper alternative in terms of opex and capex to the big Class 5 iron (e.g., Lucent (quote - news - alert) 5ESS and Nortel (quote - news - alert) DMS).

This is evidenced by the fact that the CLECs that bought these solutions mainly deployed them alongside their Class5 switches in new or smaller POPs, in essence settling for less functionality than provided by the traditional TDM switch. Very few of them deployed the NGN switches as a forklift replacement providing full Class5 switching functionality.

As noted, the key benefit of these NGN switches was substantial savings in both capex and opex. For example, a recent analysis demonstrated that using an IP/softswitch architecture to replace a TDM switch of equal capacity would shrink the space required in the central office from 28 racks to two. Reductions in air conditioning and power costs alone will allow carriers to recoup the cost of a new softswitch in only 18 months. In comparison, the cost of most circuit switches needed to be spread over more than 10 years.

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In summary, the state of carrier VoIP by the end of 2002 could be characterized as follows: legacy TDM switch vendors begin to announce end-of-life, Class4 international long-distance is being replaced by VoIP trunking solutions, and Class5 NGN is producing a lot of "buzz" but no real forklift upgrades on the carrier side, or behavioral changes on the end user side, are happening.

Though this article addresses the service provider market, it would be remiss to ignore the enterprise market, which was well ahead of the carriers in terms of VoIP deployments. Enterprises deployed IP telephony (e.g., IP PBX) services, using IP phones on their desks and softphone features that they could activate from their PCs (voice mail/e-mail convergence, unified messaging, drag and drop conferencing, etc.).

Residential VoIP Goes Mainstream

The fundamental change in VoIP usage and public perception came in mid-2003 with the emergence of Vonage. This is the first time residential VoIP went mainstream in the Western world. (Softbank's Yahoo BB! in Japan was the first commercial, large scale, voice over broadband provider and now has over 4 million users). The Vonage (news - alert) model shows how small companies with limited facilities can offer residential services with PSTN quality at highly competitive costs.

The concept of residential VoIP is simple. Its basic requirements include a computer with a broadband connection (DSL, cable) and a compact end-user device (IAD, etc.) which plugs into a regular black phone.

This is all the user needs to enjoy "sticky" services, not available over the PSTN, and with the impact to really change the way people behave in terms

VoBB may still not be ready to serve as VoIP's long-awaited "killer app." However, combined with WIFI, and WIMAX, it very well could be.

of personal communications. Here are a few examples:

- Set up conference calls with a few mouse clicks using Web browser;
- "Find Me' services so you never miss that important call;
- Set up your own 'virtual' phone numbers in different areas;
- Video calling;
- Do Not Disturb to avoid those annoying dinnertime telemarketers.

Most residential users still use VoIP as a second line. However, as users get comfortable with the quality and reliability and regulatory requirements concerning emergency services and lawful interception are relaxed, we can expect users to disconnect their PSTN line and use VoIP instead.

VoBB may still not be ready to serve as VoIP's long-awaited "killer app." However, combined with WiFi (define - news - alert), and later with WiMAX (define - news - alert) and video capabilities (as they become available to the masses), it very well could be.

Regardless, what is important is that the NGN capabilities have finally emerged — and that people will change their behavior to take advantage of these capabilities because they let you do things not possible with a PSTN phone. Major carriers and broadband service providers are aware that a fundamental behavioral change is taking place. Moreover, the success of initial VoBB networks has led the carrier market to the conclusion that VoIP-based Class5 solutions are the preferred model.

Today price is the key factor; later when incumbents inevitably lower prices to be competitive with the VoBB upstarts, services will be the key differentiator. VoBB allows service providers to offer enhanced services out-of-region since the entire network is IP-based and the local loop is circumvented. Thus, service providers can realize the true vision of NGN, in which end-user devices are connected directly to IP networks and boundaries between access and long distance networks cease to exist (i.e., the entire voice route is implemented over an IP network).

This new model presents opportunities for new types of service providers (incumbents, CLECs, MSOs, virtual network operators, startups), wishing to offer services to residential and SOHO/SME customers based on VoBB.

Big players will always have the advantage in the huge residential market due to mass, resources, marketing capabilities, cash, etc. Good examples of such are the AT&T CallVantage (quote - news - alert) prime time commercials during the 2004 Olympics. Small players could, however, penetrate the residential market with the right marketing plan and execution, while the enterprise market, due to its higher margins, is also a lucrative target for smaller players.

What Equipment Do Service Providers Need?

The VoBB model calls for robust application servers that can support residential services with carrier-grade reliability and scalability, Web-based management, feature database, emergency 911 services, and lawful interception support. The equipment should be vendor neutral to work with any independent or integrated access device.

We are still a long way from the seemingly inevitable transition to a pure IP world (today less than three percent of calls stay within the VOBB network), which means that the vast majority of calls terminate on the PSTN. And, since there is no direct IP interconnect between disparate VOBB networks, e.g., Packet8 to SoftBank, there is a need for strong IP peering capabilities. This can be done using wholesale partners for global termination, while the service provider need only supply the application.

Carriers require a VoBB platform that is interoperable with various types of SIP endpoints (IADs, SIP phones, softphones, etc.) and can communicate with SIP-based trunking gateways with robust SS7 capabilities for interconnection with the PSTN. In addition, as more carriers deploy VoIP as their technology of choice for voice trunking and enterprise solutions, there is a growing need for direct interconnection between carriers' VoIP (H.323 or SIP) networks. Carriers require IP peering solutions for secure, cost-effective traffic exchange across VoIP networks without the need for TDM conversion. This allows them to expand network reach and minimize PSTN termination costs.

These solutions should also be able to manage the routing of traffic to and from affiliates, using the most advantageous route by providing the carrier with different routing options and flexible routing management. Such capabilities minimize routing costs and optimize QoS.

Conclusion

As VoBB networks reach critical mass, carriers and service providers must accommodate this business model within their overall VoIP/NGN strategy. In terms of networks, service providers must weigh several options: from building their own independent network to outsourcing the entire network and limiting operations to marketing and management, with multiple variations between these two extremes. They must also find the right service focus, business model and technology partners. While even the very big players are not yet sure of their direction, it is clear that we are witnessing the beginning of the fundamental change that was promised when commercial VoIP products were first introduced to the world less than a decade ago. IT

Ari Rabban is vice president for corporate development and marketing at VocalTec Communications. For more information, please visit the company's Web site at http://www.vocaltec.com.

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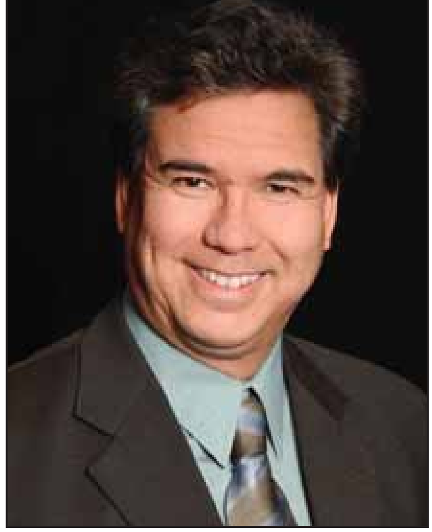

Alain Mouttham
CEO and cofounder
SIPquest

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

GG: What is SIPquest's mission?

AM: Our customers are the enterprise IP telephony (define - news - alert) equipment vendors, telecom IP telephony equipment vendors and IP telephony handset vendors who embed our SIP (define - news - alert)-based IP telephony software applications and clients into their hardware and software product offerings. SIPquest has the broadest offer on the market ranging from audio, video and data conferencing applications with presence, a SIP server and SIP-to-H.323 (define - news - alert) gateway, a desktop multimedia client, and a mobile multimedia client for PDAs and smart phones. Our ability to focus on developing these clients and applications, while our customers focus on their key strengths of bringing systems together for advance IP telephony, enables them to accelerate their deployment of new products and capabilities by six to 18 months.

It is our mission is to provide our customers, the IP telephony equipment vendors, with innovative SIP-based multimedia communications software that shortens their time to market with differentiated capabilities.

GG: What is the company's vision and how is the company positioned in the next generation telephony market?

AM: IP telephony is delivering convergence on several levels. Not only convergence between telecommunications and computing networks over a single network (the Internet), but also convergence between VoIP applications and other enterprise applications. This form of convergence is enabling better integration of multimedia communications in enterprise workflows. There is also convergence between WiFi (define - news - alert) and Cellular networks, enabling seamless mobility for VoIP and video calls. Then, there is convergence between multimedia communications and end-consumer services such as on-line gaming and home entertainment.

It is these various levels of convergence that are creating the conditions for disruptive technologies like SIP-based multimedia communications and IP telephony to reach the marketplace and to change the way we communicate. SIPquest's vision is to become a technology powerhouse in the IP telephony space. We have the capabilities to bring a steady stream of disruptive and sustaining innovations to the market, in part thanks to our exclusive partnership with Columbia University.

GG: What is SIPquest's relationship with Columbia University?

AM: The relationship with Columbia University has two components. First, we have a research agreement with Columbia and Professor Henning Schulzrinne, lead co-author of the SIP protocol. Columbia and SIPquest conduct joint research, with SIPquest funding several of Professor Schulzrinne's Ph.D. students. The second component is an agreement that gives SIPquest exclusive rights to commercialize the innovations and bring them to market on a worldwide basis. This unique relationship assures SIPquest a sustainable advantage well into the future, and our customers benefit by having access to the most advanced, research-leading products and technologies available.

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?

AM: I believe that the bigger challenge the industry is facing is not one of technology, but rather one related to new business models. For instance, an enterprise with VoIP (define - news - alert) gear needs a VoIP service provider who has a business agreement with a WiFi service provider, who has entered into a roaming agreement with a cellular service provider, and so on… to provide a seamless communications service. The key challenge is for all the players to agree on how to do roaming, handle mobility, bill each other, how to eliminate the need for the user to enter the same information five or six times, etc. It's really about putting a set of mutually beneficial multilateral business agreements between the service providers in place in order to provide seamless services to the end users. Finding the right business models between the VoIP players is really going to be the next hurdle.

GG: What makes the SIPquest product

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offering unique and how does it benefit your customers?

AM: SIPquest already has the most complete suite of SIP functionality on the market, ranging from video, audio, and data conferencing and instant messaging on the server side to multimedia clients running on desktops, PDAs, or smart phones, and we don't stop there. The relationship we have with Columbia ensures our customers a continued stream of leading-edge innovations.

For instance, all of our applications are presence-enabled and are also all mobility-ready. What this means is that the multimedia client can be running on a PDA, smart phone or dual-mode phone over a WLAN (define - news - alert) with seamless fast-handoff (<10ms) of a voice or video call to another access point, and soon to the cellular network. Our conferencing servers are also mobility-ready in that they can communicate with the multimedia client over a WLAN and support seamless handoff and roaming, while maintaining high-quality communications. By providing our applications with mobility and presence capabilities we make them future-proof. SIPquest is committed to servicing our customers with the same dedication that it develops its products.

GG: What does the future hold for the IP telephony industry?

AM: We are now in a new phase of VoIP deployment. Enterprise users have started to reap the benefits of earlier purchases of VoIP equipment not only from long-distance cost savings but from productivity increases due to presence-enabled communications at the desktop. Service providers are selling good quality VoIP services at reasonable prices and have started extending their SME and residential offerings to include some of the audio and video conferencing capabilities.

The industry is poised to evolve to the next phase toward seamless communications. Handsets will not only handle voice and messaging like our cell phones do today but they will handle video, data conferencing while taking advantage of presence information. All these capabilities enable multimedia communications between people to be totally spontaneous; to be ad hoc. We won't need to schedule our meetings ahead of time — we will actually communicate when we want, where we want, and how we want.

GG: What does the future hold for SIPquest?

AM: The future is very exciting! Our initial focus is enabling multimedia communications within an enterprise environment but we are extending these capabilities to address the residential market. Currently we enable an enterprise employee to be totally mobile within a building. We are also working on mobility in hot zones: large areas within city centers for example that are WiFi-enabled. Now we are also developing seamless mobility from a WiFi environment to a cellular network across 2G and 2.5G as well as 3G cellular networks. The opportunity is then to bring mobile multimedia communications deep into the home over a WLAN, where residential users will have a multimedia handset that allows them to make VoIP call, have video-chats, access presence information of their friends, download or stream songs and video clips on demand, etc. Basically — the full spectrum of multimedia-over-IP-over-WiFi at home, and to have access to the same capabilities on the very same handset outside the home.

The SIP revolution is all about giving people the ability to control their own communications in a seamless communication environment, and the bottom line is that SIPquest is ideally positioned to be a major player in SIP-based communications — from Enterprise to Service Providers, and from wireline to wireless. IT

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